

**PROCEEDINGS OF THE SPECIAL SESSION OF THE
SEVENTH MEXICAN INTERNATIONAL CONFERENCE ON
ARTIFICIAL INTELLIGENCE**

MICAI 2008

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SEVENTH MEXICAN INTERNATIONAL CONFERENCE ON
ARTIFICIAL INTELLIGENCE**

MICAI 2008

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Tecnológico de Monterrey, Campus Estado de México

Edited by

Alexander Gelbukh and Eduardo F. Morales



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Preface

The Mexican International Conference on Artificial Intelligence (MICAI), a yearly international conference series organized by the Mexican Society for Artificial Intelligence (SMIA), is a major international AI forum and the main event in the academic life of the country's growing AI community. In this year Mexico celebrates the 50th anniversary of development of computer science in the country: in 1958 the first computer was installed at the National Autonomous University of Mexico (UNAM). Nowadays, computer science is the country's fastest growing research area.

This volume contains the papers presented at the Special Session of the 7th Mexican International Conference on Artificial Intelligence, MICAI 2008, held on October 26–31, 2008, in Atizapán de Zaragoza, Mexico. For the Special Session, 266 submissions were considered. This volume contains revised versions of 65 papers selected according to the results of an international reviewing process. Thus the acceptance rate was 24.4%. The book is structured into 13 thematic fields representative of the main current areas of interest for AI community:

- Logic, Reasoning, and Knowledge-Based Systems
- Ontologies
- Natural Language Processing
- Machine Learning and Data Mining
- Pattern Recognition
- Neural Networks
- Genetic and Hybrid Intelligent Systems
- Computer Vision and Image Processing
- Robotics, Planning, and Scheduling
- Uncertainty, Probabilistic Reasoning, and Fuzzy Logic
- Multi-Agent Systems and Distributed AI
- Bioinformatics and Medical Applications
- Applications

The conference featured excellent keynote lectures by leading AI experts:

- Gerardo Jiménez Sánchez, Mexico,
- Stephanie Forrest, USA,
- Francisco Cervantes Pérez, México,
- Simon Haykin, Canada,
- Steven M. LaValle, USA,
- Georg Gottlob, UK / Austria.

The conference program included an oral technical session, keynote lectures, a special session, tutorials (some of them given by the keynote speakers in their respective areas of expertise), workshops, a Doctoral Consortium, and the First Mexican Symposium on Artificial Intelligence (COMIA), which were published in separate proceeding volumes.

The following papers received the Special Session Best Paper Award and the Special Session Best Student Paper Award, respectively:

- 1st place: *Use of intelligent emotional agents in the animation of autonomous virtual creatures*, by Héctor Rafael Orozco Aguirre, Félix Francisco Ramos Corchado, Luis Alfonso Razo Ruvalcaba, Jaime Alberto Zaragoza Rios, and Daniel Thalmann, Mexico / Switzerland;
- 2nd place: *A feature extraction method based on morphological operators for automatic classification of leukocytes*, by Pilar Gómez-Gil, Manuel Ramírez-Cortés, Jesús González-Bernal, Ángel García Pedrero, César I. Prieto-Castro, Daniel Valencia, Rubén Lobato, and José E. Alonso, Mexico;
- 3rd place: *Interpretation of mammographic using fuzzy logic for early diagnosis of breast cancer*, by Jorge R. Pérez-Gallardo, Beatriz Hernández-Vera, Alberto A. Aguilar-Lasserre, and Rubén Posada-Gómez, Mexico;
- Student: *New approaches for the school timetabling problem*, by Ana Cerdeira-Pena, Luisa Carpena, Antonio Fariña, and Diego Seco, Spain.

The special session best student paper was selected among papers of which first author was a full-time student.

We thank all the people involved in the organization of this conference. In the first place, the authors of the papers constituting this book: it is the excellence of their research work that gives value to the book. We thank the members of the Program Committee and additional reviewers for their great and very professional work on reviewing and selecting the papers for the conference. Our very special thanks go to Ángel Kuri and to Alejandro Peña Ayala for his help in the reviewing process.

We would like to express our sincere gratitude to the Tecnológico de Monterrey, Campus Estado de México for the warm hospitality to all MICA 2008 attendees. Special thanks go to the Vice President of Rectoría Zona Centro, Dr. Roberto Rueda Ochoa, and to the Director General of Campus Estado de México, Dr. Pedro Grasa Soler, both at Tecnológico de Monterrey, for their valuable participation and support for the organization of this Conference. Our gratitude also goes to the Division of Research and Graduate Studies and the Division of Engineering and Architecture at Campus Estado de México of the same Institute. We are grateful to the Conference Staff and Local Organizing Committee, led by Neil Hernández and Raúl Monroy.

Nothing would have been possible without the financial support of our sponsors. Special thanks go to the Dean of State of Mexico's Council for Science and Technology (COMECyT), Dr. Elías Micha, for financial support.

The entire submission and reviewing process was supported by the EasyChair system (www.easychair.org); we express our gratitude to its author Andrei Voronkov for his constant support and help. Last but not least, we are deeply grateful to the IEEE staff and especially to Randall Bilof for their great patience and help in editing this volume.

Alexander Gelbukh
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Logic, Reasoning, and Knowledge-Based Systems

Integrating Subjective Knowledge Bases through an Extended Belief Game Model

Ebrahim Bagheri and Ali A. Ghorbani

Faculty of Computer Science, University of New Brunswick
Fredericton, NB, Canada
{e.bagheri,ghorbani}@unb.ca

Abstract

Belief merging is concerned with the integration of several not necessarily consistent belief bases such that a coherent belief base is developed as a result. Various belief merging models often consist of two key functions, namely: negotiation, and weakening. A negotiation function finds the weakest belief bases among the available belief bases, and then the selected belief bases concede based on a weakening function. This process is iteratively repeated until a consistent belief base is developed. In this paper, we extend the existing belief merging models by introducing the extended belief game model. The extended belief game model operates over a subjective belief profile, which consists of belief bases with subjectively annotated formulae. The subjective information attached to each formula enables the proposed model to prioritize the formulae in the merging process. We provide several instantiations of the merging model by introducing suitable functions.

1 Introduction

Belief merging deals with the issue of acquiring a consistent belief base from several belief bases, i.e. given a set of information sources that may contain mutual inconsistencies, a coherent set of information be developed which is a fair representative of the content conveyed by all the information sources [9]. Several researchers have approached this issue as a multi-stage game. The game is a competition between the information sources where the weakest information sources are considered as the losers of the game and should make appropriate concessions. The basic idea of the merging game is to form some sort of coalition between like-minded information sources, and penalize those that are furthest away from the coalition.

The stages of the merging game are known as rounds of negotiation or competition. Konieczny's belief negoti-

ation model [9] and Booth's belief negotiation model [6] are among the most popular frameworks for belief merging, which are defined for a pure propositional logic setting. In these models, in each round of negotiation, some of the information sources are selected by the negotiation (choice) function. The selected sources should weaken their beliefs using a weakening (contraction) function. Focusing on a pure propositional logic formalism makes the definition of flexible negotiation and weakening functions more difficult in these frameworks.

Several models have been proposed which exploit priority relationships between the belief bases and their constituting formulae in the process of merge [3, 4, 5]. Priorities provide essential information with regards to the degree of formulae importance. They can be helpful in deciding which formulae need to be discarded in order to restore consistency. Possibilistic logic has been extensively used for this purpose where the formulae in the belief bases are annotated with a weight denoting their degree of necessity. One of the key factors that needs to be incorporated into the priority rankings is the degree of uncertainty of the information sources about their expressed opinions. Possibility theory addresses uncertainty through the necessity and possibility measures. Subjective logic [7], an overlay to Dempster-Shafer theory of evidence, is also able to explicitly address uncertainty. It defines uncertainty as one of the dimensions of belief in its three dimensional belief structure.

In this paper, we propose an extended belief game model, where the significance of each formula is addressed through the framework of Subjective logic using subjective opinions. Each belief source is represented by a Subjective belief base (SBB). Subjective belief bases have been shown to be a generalization of prioritized belief bases i.e. each prioritized belief base can be converted into a SBB [2]. The process of integrating various subjective belief sources is accomplished in two steps. In the first step, social contraction, some of the inconsistent belief bases selected by the choice function are both syntactically and semantically

enhanced using the enhancement functions to make them mutually coherent. For this step, some choice and enhancement functions are defined based on the subjective opinions of the belief bases. In the second step, called the combination step, the coherent belief bases acquired from the first step are combined using the Subjective consensus operator, which can have a reinforcement effect on the final subjective belief of the formulae.

2 Preliminaries

Throughout this paper, we let $\mathcal{L}$ be a propositional language over a finite alphabet $\mathcal{P}$ of propositional symbols. Ω represents the set of possible interpretations. An interpretation is a function from $\mathcal{P}$ to $\{\perp, \top\}$, where $\perp$, and $\top$ denote falsehood and truth, respectively. An interpretation ω is a model of formula ϕ , noted as $\omega \models \phi$ which makes the formula true. Furthermore, let ϕ be a formula, $mod(\phi)$ is employed to denote the set of models of ϕ , i.e. $mod(\phi) = \{\omega \in \Omega \mid \omega \models \phi\}$. Classical propositional logic deduction is represented using $\vdash$. Two formula such as ϕ and φ are equivalent, expressed $\phi \equiv \varphi$, if and only if $\phi \vdash \varphi$ and $\varphi \vdash \phi$. A formula ϕ satisfying $mod(\phi) \neq \perp$ is considered to be consistent.

A belief base φ is a consistent propositional formula (or, equivalently, a finite consistent set of propositional formulae $\{\phi_1, \phi_2, \dots, \phi_n\}$, considered conjunctively: $\phi_1 \wedge \phi_2 \wedge \dots \wedge \phi_n$). Let $\varphi_1, \dots, \varphi_n$ be n not necessarily different belief bases, we define a belief profile as a multi-set Ψ consisting of those n belief bases, i.e. $\Psi = (\varphi_1, \dots, \varphi_n)$. This definition permits two or more different sources to be identical. The conjunction of the belief bases of Ψ is represented as $\bigwedge \Psi = \varphi_1 \wedge \dots \wedge \varphi_n$. Belief profile inclusion is denoted as $\sqsubseteq$, and belief profile union is represented by $\sqcup$. The cardinality of a finite belief profile Ψ is noted as $\#(\Psi)$.

A belief profile Ψ is consistent if and only if $\bigwedge \Psi$ is consistent. Let $\mathcal{E}$ be the set of all finite non-empty belief profiles. Two belief profiles Ψ_1 and Ψ_2 are equivalent ($\Psi_1 \equiv \Psi_2$) if and only if there is a bijection (f) between Ψ_1 and Ψ_2 such that each belief base of Ψ_1 is logically equivalent to its image in Ψ_2 ($\forall \varphi \in \Psi_1, f(\varphi) \in \Psi_2, \varphi \equiv f(\varphi)$).

3 Basics of Evidence Theory

Evidence theory is one of the theoretical models, which is able to numerically quantify the degree of ignorance with regards to a certain phenomenon in an effective manner. It is a potentially useful tool for the evaluation of the reliability of information sources. Dempster-Shafer (DS) theory of evidence is one of the most widely used models that provides means for approximate and collective reasoning under uncertainty. It is basically an extension to probability the-

ory where probabilities are assigned to sets as opposed to singleton elements.

Subjective logic [7] adds to Dempster-Shafer theory of evidence. A belief expression in Subjective logic is defined as a 3-tuple $\chi_x^A = (b_x^A, d_x^A, u_x^A)$ also known as the opinion of expert A about hypothesis x (χ_x^A). It can be shown with this definition that belief (b_x^A), disbelief (d_x^A), and uncertainty (u_x^A) elements of an opinion should satisfy: $b_x^A + d_x^A + u_x^A = 1$.

The above condition restricts the possible values that can be expressed as an opinion by an expert only to the points placed in the interior surface of an equal-sided triangle. A Subjective Belief Base (SBB) is a set of formulae annotated with subjective opinions in the form of $B = \{(\phi_i^*, \chi_{\phi_i}^A) : i = \{1, \dots, n\}, \text{ where } \chi_{\phi_i}^A \text{ is a subjective opinion such that } \chi_{\phi_i}^A = (b_{\phi_i}^A, d_{\phi_i}^A, u_{\phi_i}^A)\}$. The classical propositional form of B is represented by B^* . We have previously shown that subjective prime implicants of an SBB can be developed such that they only consist of literals [2]. Therefore, without loss of generality, in this paper we assume that the formulae in B^* are literals and that the subjective belief bases are subjective prime implicants. A subjective belief profile $\mathcal{SP}$ consists of multiple SBBs. $\mathcal{SP} = (B_1, \dots, B_n)$ is consistent if and only if $B_1^* \cup \dots \cup B_n^* \cup \mu$ is consistent. $\mathcal{SE}$, $\mathcal{FB}$, and $\mathcal{K}$ are employed to denote the set of all finite non-empty subjective belief profiles, the set of all formulae in belief base B and the set of all subjective belief bases, respectively.

Definition 1 Let B be a SBB, and $\alpha \in (0, 1]$. The α -cut of B is $B_{\succeq \alpha} = \{\phi \in B^* \mid (\phi, \chi) \in B \text{ and } (b_\chi \geq \alpha \text{ or } d_\chi \geq \alpha)\}$.

Two subjective belief bases are equivalent, shown as $B \equiv_s B'$, if and only if $\forall \chi \in (0, 1], B_{\succeq \chi} \equiv B'_{\succeq \chi}$. Furthermore, two $\mathcal{SE}$ s are equivalent, denoted $\mathcal{SE} \equiv_s \mathcal{SE}'$, if and only if there is a bijection between them.

4 An Extended Belief Game Model

In this section, we propose an extended belief game model that incorporates features from both BGM [9], and BNM [6]. In the proposed model, the significance of the formulae is specified through subjective belief annotation values. Each source of belief in this model is represented using a SBB. It is assumed in our model that each SBB is internally consistent.

Definition 2 A choice function under the integrity constraint μ is a function $g_\mu : \mathcal{SE} \rightarrow \mathcal{SE}$ such that:

(ec1) $g_\mu(\mathcal{SP}) \sqsubseteq \mathcal{SP}$,

(ec2) If $\bigwedge \mathcal{SP} \wedge \mu \neq \top$, $\exists B_i, B_j$ such that $B_i^* \wedge B_j^* \wedge \mu \neq \top$, then $(B_i \text{ or } B_j) \in g_\mu(\mathcal{SP})$,

(ec3) If $\mathcal{SP} \equiv_s \mathcal{SP}'$ then $g_\mu(\mathcal{SP}) \equiv_s g_\mu(\mathcal{SP}')$.

Condition ec1 is a direct generalization from c1 in BGM. The second condition states that the choice function does not select those SBBs which are not inconsistent with any other SBBs. Therefore, it minimally includes any of the two SBBs whose conjunction produces an inconsistent belief base with regards to the integrity constraint μ . ec2 is required to ensure that the proposed extended belief game model properly terminates under any setting. It can also be seen in ec1 that the choice function is dependent on the syntactical format of the SBBs, since consistency checking is performed on the belief base without considering the subjective opinion annotation values. Finally, a negotiation function g is syntax-dependant if it satisfies ec3.

Definition 3 An enhancement function is a function $\nabla : \mathcal{K} \times \mathcal{SE} \times \mathcal{SE} \rightarrow \mathcal{K}$ where for a SBB B , and two subjective belief profiles $\mathcal{SP}, \mathcal{SP}'$, if $\mathcal{SP}' \sqsubseteq \mathcal{SP}$ and $B \in \mathcal{SP}'$ then $\nabla_{\mathcal{SP}, \mathcal{SP}'}(B)$ satisfies:

- (ee1) $\nabla_{\mathcal{SP}, \mathcal{SP}'}(B) - \nabla_{\mathcal{SP}, \mathcal{SP}'}^a(B) \subseteq B$,
- (ee2) If $\exists B_i \in \mathcal{SP}$ such that $B^* \wedge B_i^* \wedge \mu \not\equiv \top$ then $\nabla_{\mathcal{SP}, \mathcal{SP}'}^a(B)^* \wedge B_i^* \equiv \top$,
- (ee3) If $B = \nabla_{\mathcal{SP}, \mathcal{SP}'}(B)$, then $(\nexists B_i \in \mathcal{SP} \text{ such that } B^* \wedge B_i^* \wedge \mu \not\equiv \top \text{ and } \nabla_{\mathcal{SP}, \mathcal{SP}'}^a(B) = \emptyset)$.

where $\nabla_{\mathcal{SP}, \mathcal{SP}'}^a(B)$ denotes the justifications on B .

The enhancement function can be extended to subjective belief profiles by allowing $\nabla_{\mathcal{SP}, \mathcal{SP}'}(\mathcal{SP}) = \{\nabla_{\mathcal{SP}, \mathcal{SP}'}(B) : B \in \mathcal{SP}\}$. As opposed to a typical contraction function, an enhancement function can promote the contents of the information sources. Analogous to the weakening function in BGM, this function weakens the belief bases in order to remove inconsistencies, but additionally allows the information sources to add extra consistent formulae to the belief base to justify or extend their standpoint. ee1 ensures that the result of the enhancement function has been weakened except for the additional consistent formulae that have been added by the information source for standpoint justification. ee2 states that the additional formulae should be consistent with the formulae in the peer belief base with which the inconsistencies had initially occurred. Furthermore, the result of the enhancement function is only equivalent to the initial subjective belief base if a peer inconsistent belief base does not exist and no further justifications have been added to the belief base (ee3).

Definition 4 An extended belief game model is a pair $\mathcal{N} = \langle g, \nabla \rangle$ where g is a choice function and ∇ is an enhancement function. The final solution to a subjective belief profile $\mathcal{SP}$ for an extended belief game model $\mathcal{N} = \langle g, \nabla \rangle$, under the integrity constraint μ , denoted $\mathcal{N}_\mu(\mathcal{SP})$, is the subjective belief profile $\mathcal{SP}_\mathcal{N}^\mu$ defined as:

(sp1) $\mathcal{SP}_0 = \mathcal{SP}$,

(sp2) $\mathcal{SP}_{i+1} = \nabla_{\mathcal{SP}_i, g(\mathcal{SP}_i)}(\mathcal{SP}_i)$,

(sp3) $\mathcal{SP}_\mathcal{N}^\mu$ is the first $\mathcal{SP}_i$ that is consistent with μ .

Let $\mathcal{SP} = \{B_1, \dots, B_n\}$ be a subjective belief profile. The integration of the subjective belief bases in the extended game model is achieved through several rounds of negotiation. In each round the belief bases which create inconsistencies are weakened and justified through the enhancement function in order to obtain a consistent subjective belief profile $(\mathcal{SP}_\mathcal{N}^\mu)$ (social contraction step). Once a consistent $\mathcal{SP}_\mathcal{N}^\mu$ has been created, the SBBs in $\mathcal{SP}_\mathcal{N}^\mu$ are combined using the subjective consensus operator which creates a fair trade-off between the information of different sources (combination step) as a result of which an ultimately consistent subjective belief base is obtained.

5 Choice and Enhancement Functions

5.1 Choice Function

The structure of the choice function is based on a fitness measure (analogous to belief entropy functions, lower values are more desirable) that defines the quality of the subjective belief bases (SBBs) i.e. those belief bases which are less eminent than the others are selected as suitable candidates in the choice function. We define the fitness measure based on *intrinsic* and *extrinsic* properties of the belief bases. Intrinsic properties reflect the attributes of a belief base being considered in vacuum, whereas extrinsic properties of a belief base reveal its characteristics with regards to the rest of the existent belief bases in $\mathcal{SP}$.

Definition 5 The fitness measure of a subjective belief base B under integrity constraints μ is a function $fm : \mathcal{K} \times \mathcal{SE} \rightarrow [0, \infty)$. Given two SBBs B and $B' \in \mathcal{SP}$, B is considered more competent than B' if and only if $fm_{\mathcal{SP}}^\mu(B) < fm_{\mathcal{SP}}^\mu(B')$.

Now, we introduce the building blocks of the fitness measure.

5.1.1 Intrinsic Properties of Belief Bases in $\mathcal{SP}$

The intrinsic properties of each belief base is closely related with the subjective annotation values of its constituting formulae. These properties represent the strength and validity of the stated formulae, and the fortitude of beliefs of the information source that is providing the belief base. Here, two major intrinsic features are considered, namely: *Ambiguity*, and *Indecisiveness*.

Before introducing these properties, an aggregation function needs to be introduced. The aggregation function

is required in order to accumulate the overall value of each property for a subjective belief base.

Definition 6 An aggregation function is a total function f that assigns a non-negative integer to every finite tuple of integers and satisfies non-decreasingness (if $x \leq y$ then $f(x_1, \dots, x_n) \leq f(x_1, \dots, y, \dots, x_n)$), minimality ($f(x_1, \dots, x_n) = 0$ iff $x_1 = \dots = x_n = 0$), and identity ($\forall x \in \mathbb{N}, f(x, \dots, x) = x$).

Now, we can define the intrinsic properties. First, the ambiguity property provides the basis to calculate the degree of confusion in the subjective belief base about the exact fraction of belief that should be assigned to each formulae.

Definition 7 Let $\phi = (\phi^*, \chi_\phi)$ be a subjective formula in a subjective belief base, and ϵ be a normalization factor. Ambiguity is a function $\zeta : \mathcal{F} \rightarrow [0, 1]$, defined as:

$$\zeta(\phi) = - \left(\frac{1}{e^{1-(b_\phi+d_\phi)}} - 1 \right) / \epsilon.$$

Ambiguity is similar to the belief entropy metric in the generalized entropy criterion. The normalization factor, ϵ , can also be derived from the definition of the Ambiguity function. The lower limit of the function is reached when $b_\phi + d_\phi = 1$. In such a case, regardless of the value of ϵ , the function produces zero. The function outputs its upper limit when $b_\phi + d_\phi = 0$. In this case, the final value needs to be normalized to one; therefore $\epsilon = 1 - \frac{1}{e}$. The normalization factors of the rest of the functions can also be simply calculated accordingly.

The indecisiveness property is a measure of the ability of the information source to firmly state a given formula. The further away the degree of belief and disbelief of a given formula are, the stronger and more decisive the formula is.

Definition 8 Let $\phi = (\phi^*, \chi_\phi)$ be a subjective formula in a subjective belief base, and ϵ' be a normalization factor. Indecisiveness is a function $\vartheta : \mathcal{F} \rightarrow [0, 1]$, defined as:

$$\vartheta(\phi) = \left(\frac{1}{e^{|b_\phi-d_\phi|}} - \frac{1}{e} \right) / \epsilon'.$$

The intrinsic properties of the formulae can be extended to $B = \{\phi_1, \dots, \phi_n\}$, by allowing $\zeta(B) = f(\zeta(\phi_1), \dots, \zeta(\phi_n))$, and $\vartheta(B) = f(\vartheta(\phi_1), \dots, \vartheta(\phi_n))$, where f is an aggregation function. These properties are considered intrinsic, since they can be computed within the context of a single belief base. Therefore, the value of the other belief bases in the subjective belief profile is not important in the behavior of the intrinsic properties.

5.1.2 Extrinsic Properties of Belief Bases in $\mathcal{SP}$

These properties show the behavior of the belief bases and their sources in the context of the belief profile. They are

somewhat also dependant on the past behavior of the information sources. These extrinsic properties are only meaningful in comparison and within the context of a specific $\mathcal{SP}$, and not in vacuum. We introduce two extrinsic properties, namely: *Conflict* and *Reliability*.

The conflict property defines the degree of discrepancy between the belief of a single belief base towards one formula in comparison with another belief base's opinion on the same formula. Conflict can be taken as the extent of disagreement and divergence of the two belief bases.

Definition 9 Let $\phi = (\phi^*, \chi_\phi)$, and $\phi' = (\phi'^*, \chi_{\phi'})$ be two subjective formula. Conflict is a function $\delta : \mathcal{F} \times \mathcal{F} \rightarrow [0, 1]$, defined as:

$$\delta(\phi, \phi') = (b_\phi \times d_{\phi'} + b_{\phi'} \times d_\phi).$$

Analogous to the intrinsic properties, conflict can be extended to two subjective belief bases $B = \{\phi_1, \dots, \phi_n\}$ and $B' = \{\phi'_1, \dots, \phi'_n\}$ by letting $\delta(B, B') = f(\delta(\phi_1, \phi'_1), \dots, \delta(\phi_n, \phi'_n))$, where f is an aggregation function and ϕ'_i is the image of ϕ_i . Now, based on this extension, we can define the overall degree of conflict of a single belief base i.e. $\delta(B_j) = f(\delta(B_j, B_1), \dots, \delta(B_j, B_{\#(\mathcal{SP})}))$. The conflict property of a belief base shows the degree of divergence of a given belief base from the overall opinion of the other belief bases in the same belief profile.

The second extrinsic property of a belief base is related to overall behavior of the information source throughout a period of observation called *Reliability*. Under different contexts researchers have employed terms such as trustworthiness, reputation, confidence, and others to refer to the same issue. We formalize the reliability of the source of a belief base as a subjective opinion that represents the degree of belief in the fact that the information source is going to reveal a correct and consistent belief base.

Definition 10 Let $\chi_B = (b_B, d_B, u_B)$ be a subjective opinion about the reliability of a given belief base B . Reliability is a function $\mathfrak{R} : \mathcal{K} \rightarrow [0, 1]$ such that:

$$\mathfrak{R}(B) = \frac{b_B + \tau}{d_B},$$

where τ is a small correction value. It is used to prevent divide by zero.

5.1.3 Fitness Measure-based Choice Function

The formalization of the intrinsic and extrinsic properties of belief bases provides the basis for the characterization of an ordering mechanism based on the fitness measure.

Definition 11 (Extends Definition 5) The fitness measure of a subjective belief base B under integrity constraints μ is a function $fm : \mathcal{K} \times \mathcal{SE} \rightarrow [0, \infty)$ such that:

$$fm_{\mathcal{SP}}^\mu(B) = \left| \left(\zeta(B), \vartheta(B), \delta(B), \mathfrak{R}^{-1}(B) \right) \right|.$$

Definition 12 Let $\mathcal{SP} = \{B_1, \dots, B_n\}$ be a subjective belief profile. A fitness measure-based ordering, $\prec_{fm}$, of two SBBs B_i , and $B_j \in \mathcal{SP}$ is defined as:

$$B_i \prec_{fm} B_j \text{ iff } (B_i^* \equiv \top \text{ and } B_j^* \not\equiv \top) \\ \text{or } (B_i^* \not\equiv \top \text{ and } B_j^* \not\equiv \top \text{ but } fm_{\mathcal{SP}}^\mu(B_i) < fm_{\mathcal{SP}}^\mu(B_j)).$$

Theorem 13 Let $B_1 \equiv_s B'_1$, $B_2 \equiv_s B'_2$, $\mathcal{R}_{B_1} = \mathcal{R}_{B'_1}$, and $\mathcal{R}_{B_2} = \mathcal{R}_{B'_2}$. If $B_1 \prec_{fm} B_2$ then $B'_1 \prec_{fm} B'_2$.

Theorem 13 shows that fitness measure-based ordering is syntax-independent. It can be inferred that B_j is more convergent towards the general opinion than B_i within the context of $\mathcal{SP}$ when $B_i \prec_{fm} B_j$. We can now finally define the choice function based on the fitness measure-based ordering.

Definition 14 Let $\mathcal{SP} = \{B_1, \dots, B_n\}$ be a subjective belief profile. A fitness measure-based choice function under the integrity constraint μ is defined as follows:

$$B_i \in g_{\mu}^{fm} \text{ iff } \forall B_j \in \mathcal{SP} \text{ such that } B_i^* \wedge B_j^* \wedge \mu \not\equiv \top, \text{ and } B_j \prec_{fm} B_i.$$

According to this choice function, from the set of mutually inconsistent belief bases, the ones that are divergent within the context of $\mathcal{SP}$ are selected. The fitness measure-based choice function g_{μ}^{fm} should be checked to see whether it satisfies *ec1* – *ec3*. Based on Definition 14, *ec1* is satisfied. *ec2* is also satisfied from the definition of the choice function and the fitness measure-based ordering. *ec3* is also satisfied according to the following corollary. The corollary is a straight derivation from Theorem 13. It can be seen that g_{μ}^{fm} satisfies *ec1* – *ec3* and is therefore a valid choice function.

Corollary 15 g_{μ}^{fm} satisfies condition *ec3* of the extended belief game model choice function, that is, it is syntax-independent.

5.2 Enhancement Functions

The subjective opinions attached to each belief base formulae provide a suitable basis for developing attractive enhancement functions. Here, we first introduce an enhancement function that deletes the weakest inconsistent formula, and also provides the possibility for the information source to offer justifications in the form of additional formulae. Second, an enhancement function is developed, which does not require any deletions. It only manipulates the subjective opinions of the belief bases.

Definition 16 Let $\mathcal{SP}$ be a subjective belief profile and $(\phi, \chi) \in \cup(\mathcal{SP})$ be a subjective belief formula forming a singleton belief base $C_\phi = \{(\phi, \chi)\}$. (ϕ, χ) is considered to be one of the weakest inconsistent formula of $\mathcal{SP}$ if and only if:

(wif1) (ϕ, χ) is in conflict in $\cup(\mathcal{SP})$,

(wif2) $\forall (\phi_i, \chi_i) \in \cup(\mathcal{SP})$ such that $\phi \wedge \phi_i \wedge \mu \not\equiv \top$ then $C_{\phi_i} \prec_{fm} C_\phi$.

Definition 17 Let $\mathcal{SP} = \{B_1, \dots, B_n\}$ be a subjective belief profile, $\mathcal{SP}'$ be a subset of $\mathcal{SP}$, $B \in \mathcal{SP}'$ and $C = \{(\phi, \chi) \in B | (\phi, \chi) \text{ be one of the weakest inconsistent formula in } \cup(\mathcal{SP})\}$. The weakest inconsistent-based (WI) enhancement function for B is defined as:

$$\nabla_{\mathcal{SP}, \mathcal{SP}'}^{WI} = B \setminus C, \text{ if } C \neq \emptyset.$$

For each of the selected belief bases through the choice function, the WI enhancement function deletes those formulae which are among the weakest inconsistent formulae in $\cup \mathcal{SP}$, else the belief base remains intact. We now check that ∇^{WI} is an enhancement function. It is clear that *ee1* is satisfied. *ee2* is also satisfied since the information source is bound to respect this condition if any justifications are provided. Based on the definition of the weakest inconsistent formula (Definition 16), *ee3* is satisfied because no weak subjective formula can be found for elimination under such a condition, and therefore, $B = \nabla^{WI}(B)$. Let us now define a more flexible enhancement function that increases negotiability between the information sources.

Definition 18 Let $\mathcal{R}_B = (b_{\mathcal{R}_B}, d_{\mathcal{R}_B}, u_{\mathcal{R}_B})$ be the general opinion about the reliability of information source B , $\chi_p^B = (b_{\chi_p}^B, d_{\chi_p}^B, u_{\chi_p}^B)$ be B 's opinion about some proposition p . Discounting of χ_p^B by $\mathcal{R}_B$, denoted $\chi_p^B \otimes \mathcal{R}_B$, is defined as follows:

$$b_{\chi_p}^{\otimes} = b_{\mathcal{R}_B} \times b_{\chi_p}^B, \quad d_p^{\otimes} = b_{\mathcal{R}_B} \times d_{\chi_p}^B, \quad u_p^{\otimes} = d_{\mathcal{R}_B} + u_{\mathcal{R}_B} + b_{\mathcal{R}_B} u_{\chi_p}^B.$$

Definition 19 Let $\mathcal{SP} = \{B_1, \dots, B_n\}$ be a subjective belief profile, χ represent all possible subjective opinions, $\mathcal{R}_j$ be the reliability of the information source j , and $\mathcal{SP}'$ be a subset of $\mathcal{SP}$. Let $B \in \mathcal{SP}'$, $(\phi_i, \chi_i) \in B$ and $C = \{(\phi', \chi'), \dots, (\phi^{(n)}, \chi^{(n)})\}$ be the set of all images of (ϕ_i, χ_i) in other belief bases in $\mathcal{SP}$. The conformance subjective belief for (ϕ_i, χ_i) is a function $\wp : \mathcal{F}_B \rightarrow \chi$ defined as:

$$\wp_{\phi_i}^B = \oplus_{(\phi_j, \chi_j) \in C} [\otimes(\mathcal{R}_j, \chi_j)].$$

The conformance subjective belief for $(\phi_i, \chi_i) \in B$, denoted $\wp_{\phi_i}^B$, represents the general consensus about a given formulae. It is produced by the application of the consensus operator on the set of images of ϕ_i in peer belief bases discounted by the reliability of the information sources.

Definition 20 Let $\mathcal{SP} = \{B_1, \dots, B_n\}$ be a subjective belief profile, $\mathcal{SP}'$ be a subset of $\mathcal{SP}$, $B \in \mathcal{SP}'$ and $C = \{(\phi, \chi) \in B | (\phi, \chi) \text{ be one of the weakest inconsistent formula in } \cup(\mathcal{SP})\}$. The conformance belief-based (CB) enhancement function for B is defined as:

$\nabla_{\mathcal{SP}, \mathcal{SP}'}^{CB} = \{(\phi_1, \chi'_1), \dots, (\phi_n, \chi'_n)\}$
 where χ'_i is a half-open interval between the original subjective belief χ_i and $\phi_{\phi_i}^B$.

In the CB-based enhancement function, each belief base in $\mathcal{SP}'$ is required to update the subjective belief values of the formulae in its belief base. The effect of the update on the subjective beliefs should remain in a half-open interval recommended by the enhancement function for each individual formula. This way the information sources have more freedom in updating and improving the contents of their belief base. If all belief bases conform to the recommended subjective opinion $\phi_{\phi_i}^B$, consensus is reached and inconsistencies are removed. Furthermore, according to *ee2*, the information sources are able to add justifications as long as new inconsistencies do not arise. Similar to ∇^{WI} , it can be shown that ∇^{CB} satisfies *ee1* – *ee3*, and is therefore, a valid enhancement function.

6 Related Work

The Belief Game Model [9] and the Belief Negotiation Model [6] are two major work in the field of belief merging. One of the major differences between BGM and BNM is that BGM focuses on the logical content of the belief bases in making the choice for contraction, while BNM considers each source as a candidate. Therefore, BNM may weaken one of the two identical belief bases, and leave the other intact; whereas in BGM, these two belief bases are always dealt with similarly. This feature of BGM adds more anonymity to the process, since individual characteristics of the belief bases are not important. More recently, a class of general belief merge operators called DA^2 have been proposed that encode many previous merging operators (both model-based and syntax-based) as its special cases [8]. The general DA^2 framework is defined using a distance function d and two aggregation function $\oplus$ and $\odot$. The major shortcoming of the mentioned models is that they function over pure propositional logic belief bases, which makes the definition of choice functions complicated.

There has been two approaches to possibilistic belief base merging. In the first approach, researchers believe that inconsistency is totally undesirable and should be removed after merging [4]. The second approach claims that inconsistencies are unavoidable and they can exist after merging [1]. Qi et al. [10] propose a prioritized belief negotiation model that merges information sources represented by possibilistic belief bases. They introduce several negotiation and weakening functions that manipulate the necessity degrees of the formulae and employ the conjunctive operator (that can have a reinforcement effect) over the obtained consistent belief bases for combination.

7 Concluding Remarks

In this paper, we have proposed an extended belief game model which extends Konieczny's belief game model [9]. The proposed model functions over a subjective belief profile which consists of different belief bases annotated with subjective opinions. Our model is different from BGM in several aspects. First, our model considers extrinsic and intrinsic properties of the belief bases in order to make a final decision regarding the belief bases that are selected by the choice function; therefore, similar to BNM our model does not perform uniformly over content-wise identical belief bases. Second, Konieczny argues that social pressure is a permissible influence on the information sources [9]. This may lead to cases where a belief base is singled out and forced to conform with the general opinion although it possesses the correct information. To overcome this issue, our extended model employs enhancement functions instead of weakening functions that provide more space for negotiation by permitting the addition of justifications in the enhancement function (Definition 3). The flexible definition of the enhancement function provides more *room to maneuver* for those belief bases that have to concede in order for them to justify their stand point. Third, we consider the reliability of the information sources expressing the content of a belief base as one of its extrinsic properties. The reliability value is a static subjective belief about the trustworthiness of the information source.

References

- [1] AMGOUD, L., AND KACI, S. An argumentation framework for merging conflicting knowledge bases. *Int. J. Approx. Reasoning* 45, 2 (2007), 321–340.
- [2] BAGHERI, E., AND GHORBANI, A. A. On the definition of essential and contingent properties of subjective belief bases. In *Mexican Conference on Artificial Intelligence* (2008), Springer, LNAI.
- [3] BENFERHAT, S., DUBOIS, D., AND PRADE, H. Some syntactic approaches to the handling of inconsistent knowledge bases: A comparative study part 1: The flat case. *Studia Logica* 58, 1 (January 1997), 17–45.
- [4] BENFERHAT, S., DUBOIS, D., PRADE, H., AND WILLIAMS, M.-A. A practical approach to fusing prioritized knowledge bases. In *EPIA '99: Proceedings of the 9th Portuguese Conference on Artificial Intelligence* (1999), pp. 223–236.
- [5] BENFERHAT, S., AND KACI, S. Fusion of possibilistic knowledge bases from a postulate point of view. In *Proceedings of the Fifteenth International Florida Artificial Intelligence Research Society Conference* (2002), AAAI Press, pp. 519–523.
- [6] BOOTH, R. Social contraction and belief negotiation. *Inf. Fus.* 7(1) (2006), 19–34.
- [7] JOSANG, A. A logic for uncertain probabilities. *Int. J. Uncertain. Fuzziness Knowl.-Based Syst.* 9, 3 (2001), 279–311.
- [8] KONIECZNY, S., LANG, J., AND MARQUIS, P. DA^2 merging operators. *Artificial Intelligence* 157, 1-2 (August 2004), 49–79.
- [9] KONIECZNY, S., AND PÉREZ, R. P. Merging information under constraints: A logical framework. *J. Log. Comput.* 12, 5 (2002), 773–808.
- [10] QI, G., LIU, W., AND BELL, D. A. Combining multiple knowledge bases by negotiation: A possibilistic approach. In *ECSQARU* (2005), pp. 501–513.

A Logic Formal Validation Model for the Explanations Generation in an Intelligent Assistant

Juan Frausto
Tec de Monterrey*, Campus Cuernavaca
Autopista del Sol km 104
Xochitepec, Morelos, 62790, México
juan.frausto@itesm.mx

Francisco Elizalde* and Alberto Reyes
Instituto de Investigaciones Eléctricas
Reforma 113, Col. Palmira
Cuernavaca, Morelos, 62490, México
{fef;areyes}@iie.org.mx

Abstract

Given that users provided with explanations have a better performance than ones without them, we have developed an explanation generation mechanism based on the selection of the most relevant variable. However, explanations generated automatically by an Intelligent Assistant System (IAS) and those generated manually by a human expert could have inconsistencies. In this work, we present a formal validation model that uses first order logic to formalize the explanations given by the human and the IAS output as well. The aim of this validation is to prove the IAS correctness and the explanations soundness. Experimental results demonstrate that most of the explanations generated automatically in a training plant operator domain are consistent and sound with those provided by the expert. We consider that this method is a useful tool to evaluate the precision of the explanation generation mechanism that could also be extended to other domains.

1. Introduction

Most of the time, the power plants operate under normal conditions, and therefore, the plant operators' work becomes relatively a routine. However, under emergency conditions, the operator is suddenly exposed to a great amount of information. He has to be able to discriminate between erroneous inputs, and to promptly identify the source of the problem in order to define the corrective actions to be taken. To assist the operator to face these situations, an intelligent assistant system (IAS) to train and assist them has recently been developed [2]. An important requirement for intelligent assistants is to have an explanation generation mechanism, so that the trainee has a better understanding of the recommended actions and can generalize them to similar situations [6].

Although there is a lot of work on explanation generation for some representations, such as rules and qualitative models, there is very little work on explanation using probabilistic representations, in particular graphical models. The work on explanations based on graphical models (GM) can be divided according to the classes of models considered, basically Bayesian networks (BN's) and decision networks. BN's [7] graphically represent the dependencies of a set of random variables, and are usually used for estimating the posterior probability of some variables given another. So the main goal of explanations is to try to understand this inference process, and how it propagates through the network. Another possibility for static explanation consists of explaining the whole network. Markov decision processes (MDP) can be seen as an extension of decision networks, that consider a series of decisions in time (dynamic decision network). Thus, the work in explanation for Bayesian and decision nets is relevant, although not directly applicable.

Formal Logic gives us a mechanism to represent arguments in an strict and a formal way. It studies the fundamentals related with validation and propositional inference methods. The computational logic is a discipline that implies the formal logic application to represent computational arguments, automatic deduction techniques, validity and completeness fundamentals of propositional systems, and, the application of this techniques to the computational area. In this work, computational logic is a way to the establishment of a formal logic clauses to specify and verify the solution in the automatic explanation generation mechanism of the IAS as an important issue.

The following section describes the IAS, their components and a brief of the application domain. After this section we describe the Explanation Generation mechanism. Next, the proposal of the formal logic model and the experiments. Finally, the conclusions and future work.

2. Intelligent Assistant System for Operator's Training

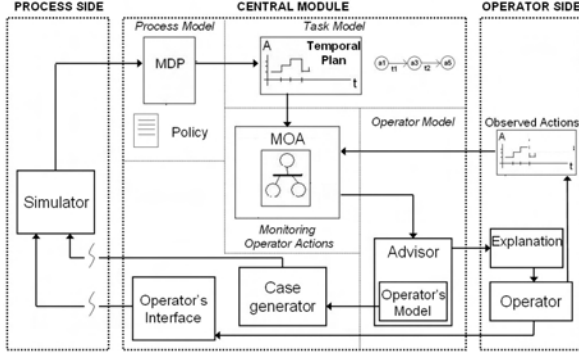


Figure 1. IAOT description.

The development of an Intelligent Assistant for Operator's Training (IAOT) is taken as a platform [2], see Fig. 1. The input to the IAOT is a recommended-plan generated by a decision-theoretic planner, which establishes the sequence of actions that will allow to reach the optimal operation of a steam generator [8]. Operator actions are monitored and discrepancies are detected regarding the operator's expected behavior. IAOT's architectural components are described as follows. Fig. 1 shows the following three main blocks: *process side*, *central module*, *operator side*.

The simulator that represents the plant is on the process side. The operator side includes the operator (human-machine interface) and the observed actions. The central module has 5 submodules: *process model*, *task model*, *operator model*, *Monitoring Operator Actions* and *interface*. The process model contains the MDP which generates the optimal policy.

The process starts with an initial state of the plant, usually under an abnormal condition; so the operator should return the plant to its optimum operating condition using some controls. Considering this situation, an optimal policy is generated based on an MDP, and an a temporal plan is obtained to reach the desired state. This plan has nodes that represent actions whose temporary relationship are represented through arcs. The model into the monitor of the operator's actions is applied in order to follow the sequence of actions from the operator to compared against those actions recommended by the adviser. If an error is detected when the action performed by the operator deviates from the optimal plan, either in the type of action or its timing. Depending on the of error and the operators model (novice, intermediate or advanced), an explanation is generated. These explanations gives evidence of the incorrect actions completed by the operator or the critical plant conditions, and

also IAOT suggest a recommendation with the correct actions. If the error is not critical the training session continues, otherwise it is terminated.

2.1. Domain

We considered a training scenario based on a simulator of a combined cycle power plant, centered in the dome (a water tank) and the related control valves.

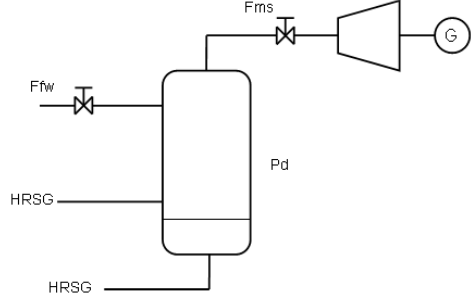


Figure 2. Schematic diagram for the steam section in a combined cycle power plant, indicating the four elements used in the explanation-framework.

Under certain conditions, the dome level becomes unstable and the operator has to return it to a safe state using the control valves. Fig. 2 depicts the main elements considered in the training scenario, including the main variables:

1. Dome pressure (Pd)
2. Main steam flow (Fms)
3. Feed water flow (Ffw)
4. Generation (G)

One of these is selected as a relevant variable for each state-action.

2.2. Explanation Generation

Built-in Explanations

As a first stage towards explanation generation, we started by defining a set of explanation units with the aid of a domain expert, to test their impact on operator training. This explanation units are stored in a data base, and the assistant selects the appropriate one to show to the user, according to the current state and optimal action given by the MDP.

The MDP representation establishes the optimal action given a plant state (optimal policy). Explanations are extracting from the MDP optimal policy by considering: a) the optimal action for the current state; b) the relevant variable (Var_{Rel}) for each action (determined by the expert); and c) the most adequate explanation that justifies the action. These explanations are defined by an expert and are encapsulated in explanation units ($Unid_{Exp}$), an example of an ($Unid_{Exp}$) is show in Fig. 3.

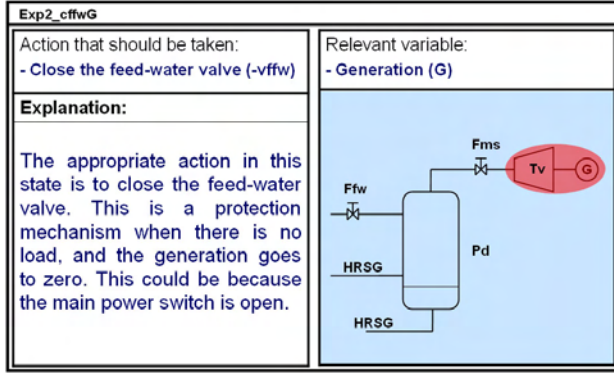


Figure 3. An Explanation Unit generated for a trainee as a consequence of an action taken during a real training session.

The built-in $Unid_{Exp}$ depicted in Fig. 3 was generated by the IAOT during a training session. An $Unid_{Exp}$ has two main components, the first one (left side) is an explanation and the second one (right side) is the selected relevant variable (Var_{Rel}).

In summary, the explanation mechanism developed is based on an optimal policy derived from an MDP, and the explanation units are defined by a domain expert.

2.3. Automatic explanations generation

Although manually generated explanations are useful, these are difficult to obtain, and to generalize to other plant subsystems or other domains. So we are interested in generating them automatically. For this, we consider at least 3 sources of knowledge: (i) a factored representation of the transition models in the MDP, (ii) an structured representation of the optimal policies and value functions, and (iii) domain knowledge stored in a knowledge base [4]. Additionally, we have defined a set of general explanation templates based on the explanation units provides by the experts. Thus, an automatic explanation mechanism, consists on extracting the relevant aspects from the knowledge sources to fill-in an explanation template. This process is illustrated in Fig. 4.

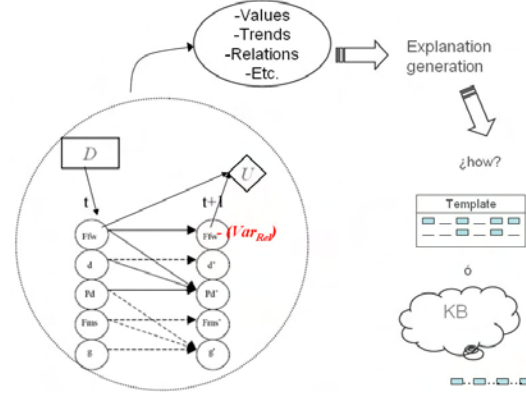


Figure 4. Automatic explanation generation: select a relevant variable from a MDP factored representation (left side) and complete the explanation with a knowledge base (right side).

From the expert's explanations, we observe that an important aspect on all explanation units is a "relevant variable". That is the state variable that is most critical under the current situation, and which is usually affected by the optimal action. So, as a first step towards explanation generation, we developed a method to identify the relevant variable.

2.4. Relevant variable selection

The algorithm [5] for determining the relevant variable considers two phases: (i) variable reduction, and (ii) variable ranking. For this we consider a structured representations of the transition function and the value functions for the MDP. The transition functions are represented as two-stage dynamic Bayesian networks (DBN) [1], one per action.

In the first phase, the variables that are not relevant to the optimal action for the current state are eliminated using the following algorithm.

Variable reduction algorithm

Given the DBN for the transition function under action "a", and the decision tree (DT) for the value function:

1. Eliminate all the variables that are non-connected (isolated) nodes in the DBN.
2. Eliminate all the variables that depend only on the same variable in the DBN (i.e., X_{t+1} depends only on X_t), and that do not change significantly (the conditional probability table is close to an identity matrix).
3. Eliminate all the variables that do not appear in the DT for the value function.

If after this stage, there remains more than one variable, we consider the second phase, variable ranking. To rank the variables, two aspects are considered: (a) the qualitative change of the variable in the transition model, and (b) the impact of the variable in the utility function. And, to evaluate the impact on the utility function, we use as a heuristic the distance of the variable to the root of the tree. That is, variables that are in upper levels of the utility tree are more important.

3. Validation Model

3.1. Description

As described before, we have an automatic mechanism based on probabilistic models to explanation generation, and this mechanism has to be validated in order to verify the coherence between human domain expert explanations and the automatic mechanism. Therefore, the validation is based on the development of a model to analyze and compare logical clauses between human domain expert samples and the probabilistic mechanism (see Fig. 5).

The validation model development consists in a three stages:

1. General model analysis and characterization.
2. Specific model analysis and clauses.
3. Verify clauses.

Clauses are generated to represent each model and to formalize validation between them. In a final stage, we obtain a measure as a percentage of verified clauses to determined the correctness of the automatic mechanism to generate explanations, based on a relevant variable, where matches between them represent that the use of explanations in an IAS is also valid.

3.2. General Model Analysis

As is described in Fig. 6, for all generated case (C_n) exists a Planning System (P_{spi}) and a Simulator (Sim_x) to generate sample tests.

In a test, each C_n of the human domain expert is equivalent to the C_n of the automatic mechanism based on probabilistic models, where $\Pi^*_{(exp)}$ is a policy that human expert analyzes to select an explanation and $\Pi^*_{(mod)}$ is the same policy but used by the automatic mechanism, therefore:

$$\Pi^*_{(exp)} \Leftrightarrow \Pi^*_{(mod)}$$

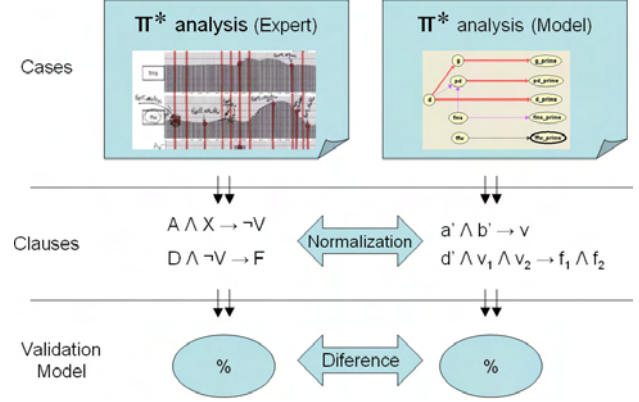


Figure 5. Main description of the validation model proposal.

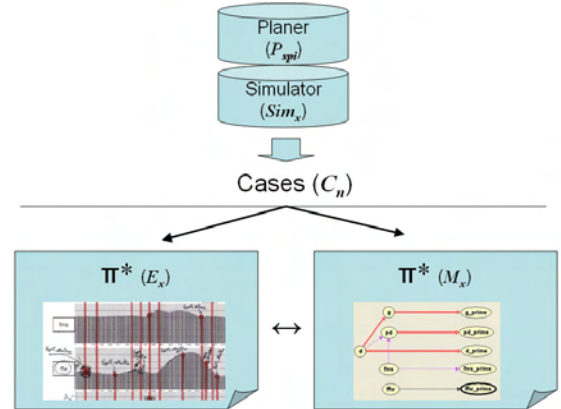


Figure 6. General model view for validation.

Therefore, exist a case C_n that is any of $c_1, \dots, c_3$:

$$C_n = \{c_1, c_2, c_3\}$$

And, if the cases are the same, the identifier for each case are:

$$C_{nexp} = \{c_{1exp}, c_{2exp}, c_{3exp}\} \text{ (for the human domain expert)}$$

$$C_{nmod} = \{c_{1mod}, c_{2mod}, c_{3mod}\} \text{ (for the automatic mechanism)}$$

3.3. Specific Model Analysis

In a more specific view, we need to describe that:

1. $\forall \{C_n = (c_1, c_2, c_3)\}$
2. $\exists \{X = (x_1, \dots, x_n)\}$ (a finite number of variables).
3. $\exists \{A = (a_1, \dots, a_n)\}$ (a finite number of actions).

A variable x_n in C_n has a set of absolute values:

$$x_n = (\text{Val}(\text{int})) \longrightarrow \text{not continuous.}$$

In to all human domain expert case $C_{nexp} = \{c_{1exp}, c_{2exp}, c_{3exp}\}$, a set of explanations is defined as (Exp):

$$\text{Exp} = (\text{Exp}_1, \dots, \text{Exp}_n)$$

In to all automatic probabilistic mechanism case $C_{nmod} = \{c_{1mod}, c_{2mod}, c_{3mod}\}$, a set of relevant variables ($Var - rel$) is associated (X_{rel}) and are selected from X :

$$X_{rel} = (x_{1rel}, \dots, x_{nrel}) \iff X = (x_1, \dots, x_n)$$

Considering that:

$$\{c_1, \dots, c_3\} \text{ has a set of } X = (x_1, \dots, x_n).$$

P.e. if variables are associated to values, then:

$$X_n = \{x_1, \dots, x_4\}$$

$$x_1 \longrightarrow G \text{ (Generation)}$$

$$x_2 \longrightarrow PD \text{ (Drum pressure)}$$

$$x_3 \longrightarrow fms \text{ (Flow main steam)}$$

$$x_4 \longrightarrow ffw \text{ (Flow feed water)}$$

3.4. The Validation Model

An state (S) is a set of associated values to the variables and has time intervals t_i :

$$(S = \{S_1, \dots, S_n\}) \iff (T = t_1, \dots, t_n)$$

In this way, a validation implies that S always has associated an explanation Exp :

$$S \iff Exp$$

Standardization of the X_{Rel} from the expert explanation

The clause equivalences among expert explanation and automatic mechanism are the same for states S and optimal actions ($a^* \iff \pi^*$), but clauses are not directly applicable to compare a relevant variable X_{Rel} with the expert explanation, due, the relevant variable has to be extracted from the first internal subsection of a verbal explanation:

An explanation contains n P sub-sections ($\text{Exp} = \{P_1, P_2, \dots, P_n\}$), where, $P_i = \{x_i \wedge x_i(S) \wedge S_t\}$, therefore:

$$(x_i(P_1) \Rightarrow X_{Rel}(Exp)) \iff X_{Rel}(Mod)$$

An state S is valid if it is selected to validate with the optimal policy from the human domain expert $\Pi^*_{(exp)}$. Therefore, an explanation is valid, if exists in a state S_i :

$$\forall S_i \iff S_i \in \Pi^*_{(exp)}$$

Then, to all model, the explanation is a function of the state S , the action a^* and the relevant variable x_{Rel} :

$$Exp \iff (S_i, a^*, X_{Rel})$$

If, the validation model is defined as a general case:

$$\{(x_1 \wedge x_2 \wedge x_3 \wedge x_4) \iff (G \wedge PD \wedge fms \wedge ffw)\} = S_i$$

If, it is defined as a specific case:

$$\{(x_1 \wedge x_2 \wedge x_3 \wedge x_4) \wedge a_i^* \wedge x_{rel}\} \iff Exp$$

Then, a clause is valid if:

$$\forall (S_i, a_i^*) \exists ((x_i(P_1) \in X_{Rel}(Exp)) \equiv X_{Rel}(Mod)) \implies \text{True}$$

Otherwise:

$$((x_i(P_1) \in X_{Rel}(Exp)) \neq X_{Rel}(Mod)) \implies \text{False}$$

3.5. Experiments

In a real case, a validating is presented over a results of a previous work [3] and the model analysis is contrasted in the las two columns of the table in the Figure 7 versus our results in [3] during the process of selecting a relevant variable, p.e.: the expert domain explanation (exp) is valid for S_i , if x_{rel} that corresponds to S_i and a_i^* , also is valid for the automatic mechanism.

Therefore:

$$\text{Exp} = \{(x_1 \wedge x_2 \wedge x_3 \wedge x_4) \wedge \Pi_i^*\} \longrightarrow \bigvee \iff X_{rel} = \{(x_1 \wedge x_2 \wedge x_3 \wedge x_4) \wedge x_{rel}\} \longrightarrow \bigvee$$

Test	selected S					Π^*	Experimental results				Relevant Var Selected by Expert	Validation		
	Var						Changes in Utility		Changes in actions			(VR(Exp)X(p1))--VR(mod)	Clauses	%
	fms	ffw	d	pd	g		$ \Delta U $		# of changes to π^*					
1	0	0	0	0	1	a1	G = 2132.44	4/5	1/1	2/7	1/1	G, Fms	Total: 80%	
2	0	0	0	1	3	a2	G = 2235.79	4/5	1/1	4/7	1/1	G, Fms	True	100
3	1	1	0	4	0	a2	fms = 2413.53	2/5	0/1	1/7	1/1	Fms, G	True	100
4	2	0	0	7	1	a4	Pd = 2762.60	4/5	1/1	3/7	0/1	Pd, Fms	True	100
5	3	0	0	0	1	a3	G = 2427.94	5/5	1/1	2/7	1/1	G, Pd	False	0
6	3	1	0	2	1	a2	G = 2109.73	2/5	1/1	2/7	1/1	G, Pd	False	0
7	4	0	1	0	1	a1	G = 2257.11	4/5	1/1	2/7	1/1	G, Pd	False	0
8	4	1	1	7	0	a4	Pd = 3271.43	1/5	0/1	2/7	1/1	Pd, G	True	100
9	5	1	0	1	1	a0	fms = 3720.05	1/5	0/1	7/7	1/1	Fms	False	0
10	5	1	1	1	1	a0	fms = 3642.53	1/5	0/1	7/7	1/1	Fms	False	0
11	0	0	0	4	0	a2	G = 2295.48	4/5	0/1	2/7	1/1	G	True	100
12	0	0	0	5	1	a0	G = 2155.54	4/5	0/1	3/7	1/1	G, Fms	True	100
13	0	0	0	6	1	a0	G = 2171.64	5/5	1/1	3/7	1/1	G, Fms	True	100
14	0	0	0	7	1	a1	G = 2132.44	4/5	1/1	5/7	1/1	G, Fms	True	100
15	0	1	0	5	0	a2	G = 2150.45	5/5	1/1	7/7	1/1	G, Pd	True	100
16	0	1	1	7	0	a4	G = 2036	1/5	1/1	2/7	1/1	G	True	100
17	1	0	0	5	1	a2	fms = 2116.43	5/5	0/1	5/7	0/1	Fms	True	100
18	1	0	1	4	0	a4	Pd = 3762.20	2/5	0/1	1/7	1/1	Pd, G	True	100
19	1	1	0	5	0	a2	G = 1719.22	0/5	0/1	1/7	0/1	G	True	100
20	1	1	1	7	0	a4	pd = 3666.26	1/5	0/1	1/7	1/1	Pd, G	True	100
21	2	0	0	1	1	a0	G = 5394.74	1/5	0/1	5/7	1/1	G	False	0
22	2	1	0	2	1	a2	Pd = 2685.48	2/5	1/1	2/7	1/1	Pd, Ffw	True	100
23	2	1	0	6	0	a2	G = 2783.64	0/5	1/1	2/7	0/1	G	True	100
24	2	1	1	1	1	a0	G = 5004.11	1/5	0/1	6/7	1/1	G, Pd	True	100
25	3	0	0	4	1	a3	G = 2427.94	4/5	1/1	2/7	1/1	G, Pd	True	100
26	3	0	1	0	0	a3	Pd = 2615.88	5/5	0/1	2/7	0/1	Pd, G	False	0
27	3	1	1	2	1	a2	Pd = 1521.18	2/5	1/1	2/7	1/1	Pd	True	100
28	4	0	0	0	1	a1	G = 2135.25	4/5	1/1	2/7	1/1	G, Pd	True	100
29	4	0	1	4	0	a4	Pd = 3304.97	2/5	0/1	2/7	1/1	Pd, G	True	100
30	5	0	0	5	1	a4	fms = 2116.43	5/5	1/1	2/7	1/1	Fms	True	100

Figure 7. Table to test the validation coincidence of the logic model versus to the previous results on the relevant variable selection stage.

This demonstrate that a clause is valid in certain state, not only for the human expert, also for the selected variable given the probabilistic model. It allows to compare how many of those states have a difference against the results appeared during the explanation generation process. In 80 % of cases of the test we obtain a *true* validation from homologation between the human expert and the IAS automatic mechanism.

4. Conclusions and future work

In this work a formal validation method was presented, it uses a logic formulas model to the human expert explanations and for the IAS's explanations in a mechanical theorems to test the precision of both explanation generation mechanisms. The IAS development was used as a platform to test the model in a MDP-based approach for explanation generation. From the MDP model of the process, the optimal actions are derived and used within an intelligent assistant for operator training. An initial evaluation of the effects of explanations in training was performed and the new optimization model for logical clauses validation was proposed based on the optimal policy and utility function evaluation. This model demonstrates that the explanations given by a human expert and by an automatic mechanism based on probabilistic models are valid for both models. Experimental results demonstrate de effectiveness of the validation by a formal model due a formal mechanism. As a future work, the refinement of the model is proposed to extend the generalization and for other domains. Also, the inclusion of more

variables in the model is an important issue to accomplish.

References

- [1] R. Dearden and R. Boutilier. Abstraction and approximate decision-theoretic planning. *Artificial Intelligence*, 89:219–283, 1997.
- [2] F. Elizalde, E. Sucar, and P. deBuen. An intelligent assistant for training of power plant operators. *ICALT*, pages 205–207, 2006.
- [3] F. Elizalde, E. Sucar, M. Luque, J. Díez, and A. Reyes. Policy explanation in factored Markov decision processes. In *Proceedings of the 4th European Workshop on Probabilistic Graphical Models*, Hirtshals, Denmark, 2008. PGM. In Press.
- [4] F. Elizalde, E. Sucar, J. Noguez, and A. Reyes. Integrating probabilistic and knowledge-based systems for explanation generation. In Roth-Berghofer, editor, *Proceedings of the 3th International ExaCt Workshop*, pages 25–36, Patras, Greece, 2008. ECAI.
- [5] F. Elizalde, E. Sucar, A. Reyes, and P. deBuen. An MDP approach for explanation generation. In *Workshop on Explanation-Aware Computing, ExaCt07*, Vancouver, CA., 2007. AAAI.
- [6] J. Herrmann, M. Kloth, and F. Feldkamp. The role of explanation in an intelligent assistant system. In *Artificial Intelligence in Engineering*, volume 12, pages 107–126. Elsevier Science Limited, 1998.
- [7] J. Pearl. *Probabilistic Reasoning in Intelligent Systems: Networks of Plausible Inference*. Morgan Kaufmann, San Mateo, CA, 1988.
- [8] A. Reyes, L. E. Sucar, E. Morales, and P. H. Ibarguengoytia. Solving hybrid Markov decision proceses. In *MICAI 2006: Advances in Artificial Intelligence*, Apizaco, Mexico, 2006. Springer-Verlag.

Ontologies

Facilitating Access Control at Groupwork by means of an Ontology of its Organizational Structure and Concept of Stage

Mario Anzures-García
Facultad de Ciencias de la Computación,
Benemérita Universidad Autónoma de
Puebla, 14 sur y avenida San Claudio.
Ciudad Universitaria, San Manuel, 72570
Puebla, México
manzures@siu.buap.mx

Luz A. Sánchez-Gálvez
Universidad de Granada, C/ Periodista
Saucedo Aranda, s/n, 18071
Granada, Spain.
luzsg@correo.ugr.es

Abstract

In the groupware applications one of the principal aims is to supply a common context, so the more information gets shared, the more easily common context can be achieved. However, the sharing of information in this type of applications leads to provide security mechanisms. Controlling the access to resources and activities is a key element in system security. In this paper, we focus in access control mechanism for security management. For this reason, firstly, we use a methodology to model the policy-based groupwork organizational structure management, which enables at system uses concepts to describe organization of the groupwork. The policy establishes the access control to the shared resources, since defines an organizational structure by means of the roles that users can play. Secondly, we consider that groupware applications contain several stages, for example, a conference management system has stages: submission, assignment, review, and acceptance of papers. At each stage role or roles that can participate are defined, in this way, we control access to shared workspace. Finally, we present an example to show as the methodology and concept of stages that we have used facilitate the access control in the groupware applications.

1. Introduction

It is common that in a long-term groupware application, the information sharing process takes place at various stages. A stage in a coordination model is defined as each of the collaboration moments [3]; for example, a conference management system has several collaboration moments: submission, assignment, review, and acceptance of papers. In addition, in this type of systems there are different groupwork organizational structures in accordance with the groupwork needs, which may differ in a new role for a user, changes in the rights/obligations of a role or even change the policy defined at a given stage. We consider that these aspects must be take in account in order to provide security mechanisms, since security is an important aspect in any system where the information is shared, as in the case of the groupware applications.

Information security deals with different aspects. From a security perspective it is important to ensure that the information is sound (integrity) and that it is authentic. It is necessary to be able to proof that an identity is who it claims to be (authentication). Furthermore access to information needs to be controlled (authorization). In most systems, security is achieved through mechanisms such as:

- *Authentication:* Mechanism to ensure the identity of an entity (users, sub-systems, etc.). It consists of two processes: identification (it obtains an identifier of the entity) and verification (it corroborates the unambiguous link between the identifier and the entity).
- *Access control:* Ensuring the protection of system resources from unauthorized access; process by

which accesses system resources, as well as the information in the workflow is regulated according to security policies and allowing access only to authorized entities.

- *Data encryption*: Process for the information treatment that prevents anyone except the recipient of the information can read it. Ensuring, therefore, confidentiality.
- *Digital signature*: Ensure accountability on a sequence of actions-finished.

While in the groupware applications, special attention has been given in authentication and mechanisms access control. In this paper, we focus on these two mechanisms to manage security. We think that to manage the user access in the groupware applications and shared resources is necessary to define conceptual models, which allow to model and to control the groupwork organizational structure.

Policies-based approaches to systems management are of particular importance because they allow the separation of the rules that govern the behaviour of a system from the functionality provided by that system [10]. In this way, it is possible to support dynamic systems and complex, by changing the policies, a system can be continuously adjusted to accommodate variations in externally imposed constraints and the environment conditions.

We use a policy-based approach to manage the group organizational structure, which is specified with an ontology that uses concepts to describe the application and the entities to be controlled. This ontology-based policy establishes who authorizes users' registration, how the interaction among users is carried out, and how users' participation is defined (e.g., by turns).

In addition, we consider that groupware applications contain several stages, for example, a conference management system has stages: submission, assignment, review, and acceptance of papers. Each stage controls the roles that can participate in it, which facilitates the authentication and interaction among users in the shared workspace.

This paper is organized as follows. Section 2 gives a brief introduction to the access control and the main models for groupware applications. Section 3 explains the ontological model of the group organizational structure we show as this operates by means of a real application case based on a well-known collaborative application: a conference management system. Finally, we present some conclusions and future work.

2. Models of Access Control for Groupware Applications

Security is an indispensable part of any information sharing system, because security emphasizes the information protection and not only user access at shared workspace. Among the several areas of security under consideration for groupware environments, authorization and access control are particularly important, because collaborative applications may offer open access to local desktops or networked resources.

Access control models are used to decide how the available resources in the system are managed. In order to implement effectively and appropriately these models into groupware applications, the following requirements have to be taken into account [9], in such a way that access control must:

- be able to protect any type of information and resources at different levels of granularity.
- facilitate transparent access for authorized users and strong exclusion of unauthorized users in a flexible manner that does not constrain groupwork.
- be expressive enough to allow high level specification of access rights, and thereby managing better the increased complexity that groupware introduces.
- be dynamic, it should be possible to specify and change policies at runtime, depending on the environment or groupwork dynamics.
- support the delegation, revocation and management of access policies in runtime.
- grant access by considering the current context user, mainly in groupwork.

There are several access control models for collaborative environments [9], such as Access Matrix Model [5], Role-Based Access Control (*RBAC*) [6], Task-Based Access Control (*TBAC*) [8], Team-Based Access Control (*TMAC*) [7], Spatial Access Control [1], and Context-Aware Access Control [2]. *RBAC* is very effective and the most important and popular for traditional and collaborative systems, but it has several weaknesses:

- The roles in *RBAC* lack flexibility and responsiveness to the environment.
- *RBAC* supports the notion of role activation within sessions, but it does not go far enough to encompass the overall context associated with any collaborative task.
- *RBAC* lacks the ability to specify a fine-grained control on individual users playing certain roles and on individual object instances.

- The specification of constraints has not been discussed in the RBAC model. Constraints are an important aspect of role-based access control and a powerful mechanism for laying out higher-level organizational policy.

We take the idea of RBAC that permissions are assigned to roles rather than users. In this way, the policy has not to be changed when users modify their role within the organization. In addition, our ontological model (see the next section) considers: more sophisticated and suited access policies, the overall context associated with any collaborative activity without being limited to a specific aspect, the dynamic nature of the groupwork, in such a way that permissions, roles and constraints are part of the model and they can be changed in a just-in-time fashion, and of course, all the aspects related to the collaborative application security.

3. An Ontology for Facilitating Access Control in Groupwork

We consider necessary to analyze and model the access control management from the first step of the development of groupware systems at both architectural and organizational level, in order to facilitate their dynamism and greatly decreases their final cost.

In this paper, we present a model that specifies the groupwork organizational structure, which is used to model the access control to the shared resources and workspace. We use an ontology to model this structure. An ontology, according to Gruber, is a formal and explicit specification of a shared conceptualization [4].

An ontology specifies the concepts, relations, axioms, and instances of the elements of the domain, which contains all the information necessary for access control. The concepts and relations enable the modelling of the group organizational structure. The axioms specify the restrictions into the domain, facilitating the access to the shared resources and the interaction among authorized users. We consider that our ontology supports flexibility in a collaborative environment in order to adjust to changes of the groupwork, facilitating the access control in each stage.

The collaboration stages in the groupwork ensure the protection of shared resources from unauthorized access, since each stage constrains to the users that can participate in accordance with the roles that they are playing. Any of the existent access control models for collaborative applications that we have studied do not

take into account this concept of stage, which facilitates the adaptation process and the access control to the shared workspace in a collaborative application.

3.1. Ontology Concepts

We think that various sessions can be run in a collaborative application, a session can have several stages, and at each stage the organizational structure is governed by a unique policy. We are going to describe the concepts, relations, axioms and instances of our ontology. In our work, the conceptual modelling (see Figure 1) considers the following concepts:

- *Group organizational structure*, which is governed at a given stage by a specific policy.
- *Policy*, which configures the above mentioned structure and defines a set of roles.
- *Role*, which must be played at least by one user so that the policy can operate in an appropriate way. Each role has a status and a set of rights/obligations.
- *Status*, which defines the role authority according to the position of this inside the organization.
- *Right/obligation*, which constrains the user actions in the groupwork.
- *User*, who is a person that plays one or more roles and these carry out collaborative tasks.
- *Task*, which is a set of activities carried out by one or more users to achieve a common goal.
- *Shared resource*, which represents the resources used to carry out a task.

3.2. Ontology Relations

The concepts are associated by means of the following relations (see Figure 1):

- *is_governedby* (Group_Organizational_Structure, Policy), which specifies that a group organizational structure is governed by a policy.
- *defines* (Policy, Role), which specifies that the policy defines the roles.
- *plays* (User, Role), which indicates that the user can play roles.
- *has* (Role, Status), which determines that the role has a status.
- *provides* (Role, Right/Obligation), which specifies that each role provides a set of rights/obligations.

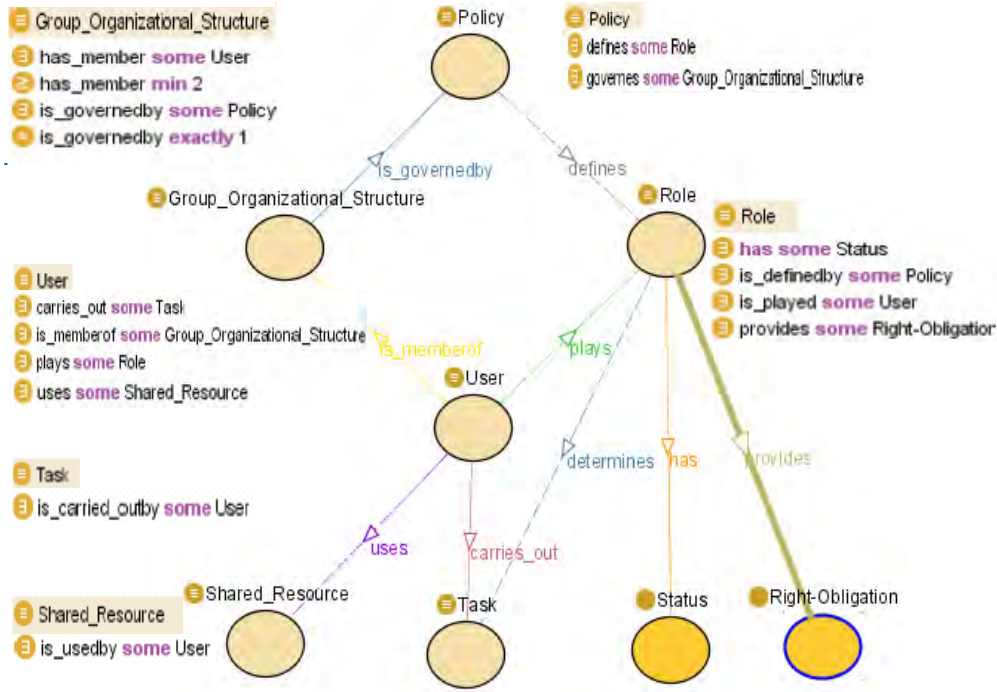


Figure 1. Ontological modelling of the groupwork organizational structure based on OWL [11].
Figure generated by Jambalaya [13] plug-in for Protégé [12].

- *determines* (Role, Task), which indicates that each role determines the tasks that a user playing it can carry out.
- *uses* (User, Shared_Resource), which defines shared resources used by a user.
- *carries_out* (User, Task), which specifies that the user carries out one or more collaborative tasks.

3.3. Ontology Axioms

The main axioms of our ontology are (see Figure 1):

- A group organizational structure is only governed by a policy in a certain stage.
- A group organizational structure has at least two users.
- Each policy defines at least one role.
- Each role has to be played by at least one user.
- Each task has to be carried out by at least one user.

3.4. Ontology Instances

The conference management system is an appropriate example of a collaborative application that shows the functionality access control using an ontology and collaboration stages. In this type of systems there are the following stages: submission, assignment, review, and acceptance of papers and commonly the roles that can participate are: Author, Program Chair (PC), and Program Reviewer (PR). A user can play one, two or even the three roles in the system.

Each stage establishes the roles that can participate and the collaboration type (that can be asynchronous or synchronous) that carry out. For example, let us the conference management system: The submission stage of papers is an asynchronous collaboration, where the user that plays the Author role submits its paper and its information, and the user that plays the PC role browses submissions. The assignment stage of papers also is an asynchronous collaboration, here the user

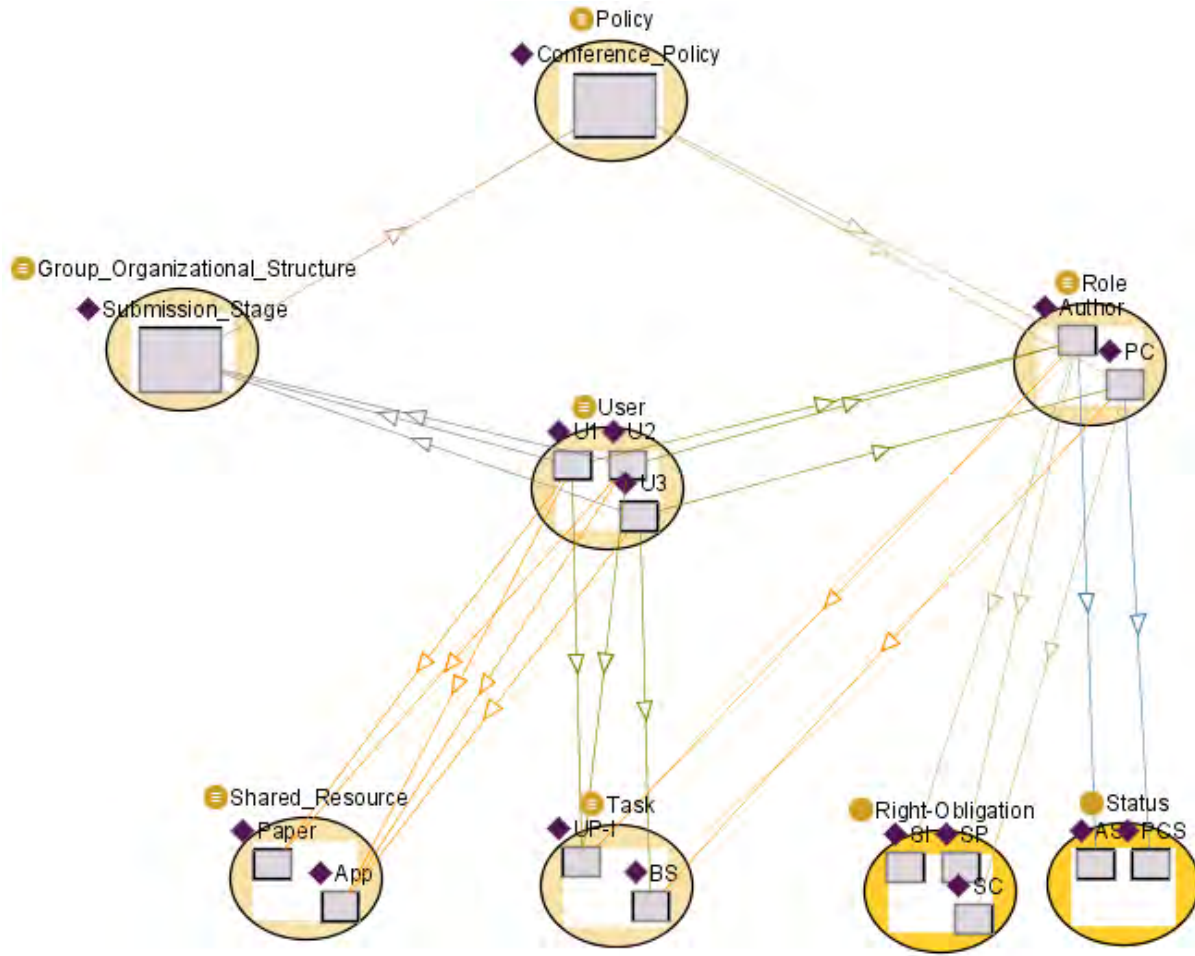


Figure 2. Ontology instances (based on OWL [11]). Figure generated by Jambalaya [13] plug-in for Protégé [12]

that plays the PC role assigns papers to the user that plays the PR role. The review stage of papers the user that plays the PR role reviews the assigned papers, and the user that plays the PC role browses that papers have been reviewed, this review stage can be both asynchronous as synchronous. Finally, the last stage (asynchronous), where the user that plays the PC role sends the acceptance notification to all the users that play the Author role and their papers have been accepted, these users re-upload their paper.

We focus in the submission stage of papers in order to define the ontology instances. The boxes in Figure 2 show the instances corresponding with this stage, which are:

- Group organizational structure: Submission Stage.
- Policy: Conference Policy.
- Role: The roles that can be played at this stage are:

- a. Author, whose *status* is AS and *rights/obligations* are submitting paper (SP) and information (SP, SI), and has the following tasks associated: upload paper and information (UP-I).
 - b. Program Chair (PC), whose *status* is PCS and *rights/obligations* are the system control (SC), and can carry out the following tasks: browsing submissions (BS).
- User: We define three users: U1, U2, and U3.
 - a. U1 and U2 play the Author role and they use the shared resources: paper and the application (App).
 - b. U3 plays the PCC role, which uses the application (App).

4. Conclusions and Future Work

In this paper, we have presented an ontological modelling and we have applied the concept of stage. The former is used for access control in collaborative applications and facilitates the adaptation dynamic of the groupwork. Ontology on one hand, it adapts the organizational structure when changes the role that a user can play and/or the tasks that a user can carry out in a session. On the other hand, it evolves the model when adds a new role or change the policy established. The latter facilitates the adaptation process and the access control to the shared workspace in a collaborative application.

The future work will be focused on to develop a set of services, which use our ontological modelling and the concept of stage to facilitate the access control and the adaptation with only modify the service correspondent.

5. References

- [1] A. Bullock, SPACE: Spatial access control for collaborative virtual environments, PhD. thesis, University of Nottingham, 1998
- [2] M. Covington, W. Long, S. Srinivasan, A. Dey, M. Ahamad, and G.D. Abowd, "Securing context-aware applications using environment roles", *ACM Symposium on Access Control Model and Technology*, 2001
- [3] C. Ellis, and J. Wainer, "A conceptual model of groupware", *Proceedings of the ACM conference on CSCW*, 1994, pp. 79-88
- [4] T.R. Gruber, Toward principles for the design of ontologies used for knowledge sharing, *International Journal of Human Computer Studies*, Vol 43(5/6), 1995, pp. 907-928
- [5] B. Lampson, Protection, *ACM Operatives Systems Rev.* Vol. 8(1), 1974, pp. 18-24
- [6] R.S. Sandhu, E.J. Coyne, H.L. Feinstein, and C.E. Youman, Role-based access control models, *IEEE Computer*, Vol. 29(2), 1996, pp. 38-47
- [7] R. Thomas, "Team-based access control (TMAC)", *Proceedings of ACM Workshop on Role-Based Access Control*, 1997, pp. 13-19
- [8] R. Thomas, and R. Sandhu, "Task-based authorization controls (TBAC): Models for active and enterprise-oriented authorization management", *Database Security XI: Status and Prospects*, 1997
- [9] W. Tolone, G. Ahn, T. Pai, and S. Hong, Access control in collaborative systems, *ACM Computing Surveys*, Vol. 37(1), 2004, pp. 29-41
- [10] G. Tonti, J.M. Bradshaw, R. Jeffers, R. Montanari, N. Suri, and A. Uszok, Semantic web languages for policy representation and reasoning: A comparison of KAoS, Rei, and Ponder. *Springer-Verlag, LNCS*, Vol. 2870, 2003, pp. 419-437
- [11] <http://www.w3.org/2004/OWL/>
- [12] <http://protege.stanford.edu/>
- [13] <http://www.thechiselgroup.org/jambalaya>

Ontology-based Knowledge Retrieval

Héctor Díez-Rodríguez and Guillermo Morales-Luna
Computer Science, Cinvestav-IPN
Mexico City, Mexico
{hdiez,gmorales}@cs.cinvestav.mx

José Oscar Olmedo-Aguirre
Electrical Engineering, Cinvestav-IPN
Mexico City, Mexico
jolmedo@cinvestav.mx

Abstract

We deal with knowledge retrieval within the context of Virtual Learning Environments (VLE). A good VLE should deliver relevant learning materials to the learner at the most appropriate time to facilitate knowledge acquisition by Problem-Based Learning (PBL). In PBL, students should retrieve information about a problem by working in small groups with the guidance of a learning facilitator providing materials required through the problem solving process. The use of ontologies to represent domain knowledge improves the information management in a VLE because it enables automatic reasoning and facilitates the processes of knowledge searching and retrieving needed to promote the interest in problem solving. We propose an ontology-based for searching, discovering and publishing relevant materials as Learning Objects to help students in the PBL approach.

Keywords. *Intelligent Tutoring System, Knowledge Management, Ontology, Problem-based Learning.*

1. Introduction

Relevant information is necessary to help students in *Problem-Based Learning* (PBL) [4]. With current computing and information storage, the volume of learning materials has increased dramatically. However, saturated with huge and diverse learning materials, users feel that retrieving relevant learning materials is still a challenging task.

A *Virtual Learning Environment* (VLE) [9] increases productivity in education since it provides access to learning materials at any time and any place, and it may provide information transmission for knowledge construction to a PBL. Historically, relevant learning materials are manually integrated to VLE by teachers, and this approach involves hard work. Although students using a VLE also have access to information retrieval tools in a network (e.g. Google, Lycos or CiteSeer), a proliferation of superfluous data obtained under these conditions in the Web does not guar-

antee any form of validation or trustworthiness. Overabundance of search results poses a *cognitive overload* [10].

Applications of ontologies to model related components of relevant learning materials contribute to effective knowledge search. Concept ontologies in any area aim for standardizing and improving knowledge search and discovery mechanisms. However, within VLE, there is a lack of formal ontology description of learning materials as a knowledge repository to help students to retrieve relevant domain information necessary to solve a specific problem. We present a constructivist VLE paradigm that integrates automated mechanisms of knowledge searching and publishing based on ontologies describing learning domains within a PBL approach. Our main contribution is content search in known repositories of relevant learning materials. The relevant solutions, useful to solve a specific problem, can be extracted by queries based on semantic relationships embedded in the domain ontology.

In section 2 some foundational concepts of PBL theory are described, in section 3 the integration of ontologies and Learning Objects in a VLE as a suitable knowledge representation for pedagogical design are analyzed, in section 4 a problem domain ontology and its use in a problem solving strategy is described and in section 5 related work is cited. Section 6 consists of concluding remarks.

2. A learning model from problem-solving

PBL has been used in education for over forty years as a method of teaching the practical application of knowledge in a real world setting [25], and it is defined as a way of constructing and teaching courses using problems as the stimulus and focus for student activity [6].

The traditional path that takes the process of conventional learning is reverted when working in a PBL model. Traditionally, the course material is exposed first and then its application is presented, including later on some comments about more general applications. In a PBL, the problem is presented and analyzed first, asking to the students to identify their learning necessities. Afterwards they search

for relevant information to solve the problem at hand, and if any information is found, they integrate it into the solution. There are certain common characteristics in PBL courses:

- use of real-life, complex, ill-structured problems, with any number of correct solutions;
- the problem is presented to students without direct information of how to solve it, however, resources and scaffolding are available for students to solve the problems by themselves;
- students work in small group, with a facilitator; and
- the problem is used as the focus from which the learning is structured.

In PBL, the students collaborate on complex problems, thereby distributing the cognitive load among the whole profiting of the distributed expertise. Collaboration is a social structure in which two or more people interact with each other, leading to social situations and interactions that have a positive effect on them [13]. Searching, discovering, exchanging and publishing information are important parts of PBL, because knowledge is constructed socially through the joint efforts directed towards common objectives. The appropriate application of PBL depends on the knowledge domain characteristics and representation, which conducts the solution process in PBL.

3. Ontology and Learning Objects for knowledge management in a VLE

Learning Objects conform one of the main research topics in the e-learning community on recent years [14], and they are widely adopted for knowledge representation in many VLE's. An ontology [17, 24] is a explicit representation of domain concepts that provide the basic structure around which knowledge bases can be built. The intention to represent concepts of any area in ontologies is to standardize and improve knowledge searching and discovery mechanisms. An ontology that actualizes its structure of concepts allows the user of VLE to play an active role in pedagogical development through semantically relevant knowledge searching.

3.1. Learning objects to represent knowledge in a VLE

Learning Objects (LO) constitute a new way of thinking about learning materials. An LO is a unit of digital resource that can be shared to support teaching and learning [28], it is an independent and self-standing unit of learning content. Because learning materials can take a variety of

forms, among plain text, program simulation, or any other form of multimedia content, there are LO's suitable to represent them. In VLE there are emerging standards for describing learning resources, among them Learning Objects Metadata (LOM) proposed by the IEEE [19] as an extension of Dublin Core, that gradually has become the standard reference for educational system managing LO. Usually LO metadata standards are intended to generalize taxonomies and vocabularies for learning objects repositories for the involved disciplines [15, 29]. The development of such taxonomies suggests that there is a tacit ontology behind the metadata standard. Though LO have been already used for knowledge sharing in previous VLE systems [12], [23], they have not been explicitly organized in ontological structures for solving problem purposes.

3.2. Ontologies as conceptual models

Ontology is a science that studies the explicit formal specification of the terms of a knowledge domain along with the existing relations among them [18, 20]. Many different definitions of the term are proposed. One of the most widely quoted and well-known definition of ontology is Grubers: Ontology is an explicit specification of a conceptualization [16]. Thus ontology is a logical system of concepts and their relations in which they are defined and interpreted in a declarative way.

The use of ontologies in the educational environment design is not new. Bloom defined a taxonomy of educative goals, in which the category *Contents* had a roll that specifies concepts that were taught in a course [5]. Bloom's taxonomy of education objectives is a framework which has been widely used in all disciplines. The original Bloom's framework includes six levels of learning: knowledge, comprehension, application, analysis, synthesis and evaluation. However, given the recent development in the field of knowledge management, the term knowledge is no longer appropriate in this context. Nevertheless, investigations that use ontology in a VLE have been focused in two fundamental issues [8]: a) Interoperability and classification of Learning Objects used in Learning Management Systems (LMS) [22]. The ontologies define a vocabulary that is shared by the applications. b) Generation of adaptable Learning Environment [27, 26]. The ontologies describe roles and contents to personalize a learning process. In a rather general way, we realize ontologies as directed multigraphs. The *nodes* are *classes* and the *edges* are ordered pairs of classes. Each edge is labelled by the set of *relations* among the corresponding ordered pair of classes. Typically, a *query* can be translated into an equivalent problem consisting in finding a special path connecting two classes. The search of such paths may be conducted with advising strategies based either on the discriminating

words in the query or in the query’s syntactical structure.

Few research reports provide an explicit generic structure of ontologies for knowledge sharing for educational purposes [26, 31], although the literature related to PBL and LOM is huge. While learning objects metadata describe the artifacts of LO shared by diverse domains, ontology represents a knowledge domain that shares the relationships of LO within a specific context. The use of ontology does not exclude the use of metadata. The intelligent discovery of LO requires information not supported by the current set of metadata in the LOM standard. For example, it is necessary for each LO to specify exactly how that LO is related to concepts in a particular domain and the kinds of learning outcomes that are possible in that domain, i.e. an ontology of concepts in a domain.

Another kind of ontologies required is for the physical structuring of LO. To allow LO being interpreted and rendered consistently in different learning environment, it is important that ontologies be developed for describing the structure of LO.

4. Use of ontology in guided-problem solving

In PBL, while students are identifying crucial parts of the problem, they are also conceiving possible solutions. These solutions can be characterized according to the description along with the restrictions of the problem domain to guide the student to a good solution.

In problem domains that are more susceptible than others of having a better formalization exist fundamental concepts that may be classified with the basic ontological relationships `SubclassOf` and `PartOf`.

The ontologies involved in the guided problem solving organize knowledge in two categories: **Concepts** and **Solutions**. The **Concepts** class describes the context of the problem domain, whereas the **Solutions** class describes existing algorithms or solutions that are related with this concept. In table 1 we show some examples of concepts and solutions from a Computational Algorithms course. Class **Concepts** organizes in subclasses concepts describing a problem domain and each subclass has the `name` and `solvewith` properties. Property `name` is used to identify class or subclass names. Property `solvewith` associates concepts with solutions. Class **Solutions** organizes solutions solving problems within the domain. Each subclass belonging to this category has several properties. Property `description` has a brief narrative description of the solution to the students. Properties `linkPW` and `linkOA` contain pointers to educative materials (as LO’s) describing solutions. Property `linkPW` points to the Permanent Learning Objects describing a solution normally elaborated by an expert. Property `linkOA` points to Temporal Learning Objects (TLO) which are elaborated by students and

Table 1. Concepts and Solutions examples

Domain Graph	
Concepts	Solutions
Positive/Negative weight	Bellman-Ford algorithm
directed graph	Dijkstra algorithm
shortest path	Kruskal algorithm
maximum flow	Boruvka algorithm
Domain Sorting	
Concepts	Solutions
Stable/Non stable	Radix/Quick sort
Quadratic/Semilogarithmic	Bubble/Merge sort
Time Complexity	
Large/Small Data set size	External/Internal sort

professors as explained in section 4.2. These links allow navigation to review a learning material stored in the Learning Objects Repository. When clicked in those links, relevant documents are displayed in the browser pane.

4.1. Searching for knowledge

Searching for the set of solutions to a given problem by a query consists on determining the set of Learning Objects that represents an appropriate set of solutions to the problem. The algorithm **SEARCH** shown in Figure 1 retrieves all the known solutions that can better solve the given problem. The algorithm **SEARCH** receives as inputs an **Ontology** and a **Query** that is an abstract narrative description of the problem, and returns as outputs the set of **Solution** that solves the **Query** according to the **Ontology** and the set of **LearningObjects** associated with the **Solution**. As **Ontology** has a hierarchical structure, search starts in the top of the structure descending by a breadth-first traversal from the most general to more specialized concept.

The algorithm begins by getting all the words extracted from the **Query** (line 2). The algorithm iterates for all word in set **Words** (lines 3 through 10) and for all **Concept** in the **Ontology** (lines 5 through 9) to find those **Concept** whose property `Name` is the root of a discriminating word. In case the `Name` identifies an abstract **Concept** in the **Ontology** (lines 6 through 8), a new entry in the **Solution** array is defined to associate the `Name` to the **Solution** obtained from property `SolveWith` of **Concept** (line 7). The set of all final **Solutions** are obtained by intersecting all partial solutions (lines 10 through 13) and the set of all **LearningObjects** are obtained by joining the sets of LO given by property `linkPW` of each final solution (lines 14 through 16). In this algorithm, function **Split(Query)** returns the set of all **Words** (with no duplicates) that appear in **Query**, func-

Figure 1. Algorithm SEARCH

```
ALGORITHM SEARCH
INPUT Ontology, Query
OUTPUT Solutions, LearningObjects
BEGIN
  1 Solutions, LearningObjects:=EmptySet
  2 Words := Split(Query)) \
    NonDiscriminantWords
  3 FORALL Word IN Words DO
  4   Name := Lexicon.GetStem(Word)
  5   FORALL Concept IN Ontology
  6     IF Concept.Name=Name THEN
  7       Solution[Concept.Name] :=
         Concept.SolveWith
  8     END IF
  9   END FORALL
10 END FORALL
11 FORALL s IN dom(Solution) DO
12   Solutions :=
     Intersection[Solutions,Solution[s]]
13 END FORALL
14 FORALL s IN Solutions DO
15   LearningObjects :=
     Union[LearningObjects , s.linkPW]
16 END FORALL
END
```

tion `GetStem(Word)` returns the root of `Word` by using a `Lexicon` such as `WordNet` [3]. The algorithm uses dynamic associative arrays (like those found in JavaScript) in which a new entry is defined by assignment (as in line 7). There are no duplicated entries for this array. Associative arrays have an intrinsic function `Dom()` that returns the set of all elements for which an entry for the array is defined. Predefined set `NonDiscriminatingWords` contains frequently used words, among articles, pronouns, and verbs, which do not contribute to determine the problem domain. The operations of `Union()`, `Intersection()` and `Difference()` for generic sets have their usual meaning. The algorithm also uses high-level iterator `FORALL` having the form `FORALL (element) IN (set) DO (action) END`, meaning that variable `(element)` is instantiated with each member of `(set)`, if non-empty, to perform the given `(action)` upon `(element)`. For the `Ontology`, the iterator traverses the hierarchy of nodes in a breath-first manner beginning by the top node, as explained before. Since a problem generally involves concepts whose solutions may completely differ from others, the algorithm returns no solution when `Solutions` is empty. No found solution means inconsistency in the `Query`.

4.2. Publishing of knowledge

Publishing consists on augmenting a centralized repository of LO, in a Sharable Content Object Reference Model (SCORM) standard [2], with the known solutions for the problem so far. The publication process is lead by an instruction facilitator. In practice, LOs can be either permanent or temporal, according to their duration in the repository. Permanent Learning Objects are elaborated by experts (generally the facilitators) to be used as reference in the subject matter and represents the most complete information available. Temporal Learning Objects are elaborated by students as incomplete, tentative, discardable solutions that arise during the problem solving process. These materials are implementations of a solution and complement an exposed description solution in Permanent Learning Objects.

4.3. Implementation

The domain ontology, the search algorithm and the LO repository are located in a dedicated server, which it is manipulated by means of a Java application that executes in a Web server. We have chosen Protégé [1] for ontology developing because it is a well formalized, well supported that outputs an ontology in the OWL language [21]. For knowledge management we implemented a `RibONTOMiddleware`. It is a middleware [24] that manipulates both the ontology and the LO repository, providing high-level services for discovery, searching and publishing of knowledge. In addition, we develop a constructivist VLE `EnEMoCi` that provides the functionality of a learning management system to conduct PBL tasks involved in teaching courses. It facilitates user's administration and knowledge retrieval as learning objects by `RibONTOMiddleware`.

4.4. Case study

Let us analyze two different problems, a Graph Theory problem and a Sorting problem, that are asked to the students of a Computational Algorithms course:

Problem # 1 *Travel Salesman Problem*: "A road map contains information about 20 cities and the roads that connect them have a length given in kilometers. There is always at least one route between any two cities of the map. The problem consists in finding an optimal route between any two cities that minimizes the distance covered by the route."

Problem # 2 *Bulk sorting*: "Sort 1 gigabyte of data using a computer with only 128 megabytes of RAM."

Following the PBL methodology, the students start their activities by identifying the learning objectives they have. For

Table 2. Search results from Google

Description	Avg (%)
Dijkstra's Algorithm Description	10 (*)
Directed weighted Graph Theory and Dijkstra's Algorithm Description	20 (*)
Data structure exercises	50
Floyd's Algorithm application paper	20

a) Graph Theory problem

Description	Avg (%)
Sorting Algorithms Description	20 (*)
Merge sort	10 (*)
Quick sort	10
Complexity Analysis of Algorithms	30
Design of Algorithms	20
The link could not be shown	10

b) Sorting problem

(*) Link has information useful for students. **Avg:** Average

those problems, the learning objective can be specified in abstract terms by the queries “Finding the shortest path in a directed graph” for the first problem and “Finding a suitable external sorting algorithm” for the second problem.

Documents related to the specified queries were obtained from a search engine like Google [24] that comprises databases containing million of documents organized by classical information retrieval methods. Table 2 summarizes the first ten results the search engine returned to answer each query. From the list of results, the students have to decide which information is most appropriate by examining each result. It was observed that only about the 30% from the retrieved information is useful, because it contains enough information (theoretical explanations and algorithms) related to the purpose of the query, so that the students can satisfy their learning objectives.

Nevertheless, if the learning objective that the students have identified can be situated in an ontological domain of Computational Algorithms, then more precise query results could be obtained using the **RIbONTOMiddleware**, a non-conventional search engine based on ontology containing abstract terms. The results obtained using **RIbONTOMiddleware** in same queries are summarized in table 3. Using this search engine based on a context ontology, the following conclusions can be derived: (1) 100 % of the retrieved information is useful for the students, and (2) the number of links was reduced significantly with respect to the results obtained from the Google search engine.

Table 3. Algorithm search results using RIbONTOMiddleware

Solutions	Solutions
Bellman-Ford	Merge Sort
Floyd-Warshall	Four Type Sort
Dijkstra	Polyphase Sort

5. Related work

Proposals that implement a constructivist approach and PBL [30] in the educational process have made emphasis in the experimentation phase for knowledge generation. Nevertheless, its effectiveness is limited because it has insufficient mechanisms for reusability and integration of generated knowledge and it lacks of motivation in searching for the known solutions and making widely available new knowledge as automated mechanism integrated to VLE.

There have been a number of recent efforts aimed at the development of ontologies for e-learning. The O-DEST system, proposed in [26], comprises ontology for e-learning process, such as course syllabus, teaching methods and learning activities. However the description only refers to pedagogical rolls and activities, and it does not approach the use of search mechanisms for knowledge discovery.

In [7] the authors give an example of an ontology development in accordance with the ACM Computer Classification, this ontology is represented in RDF and is used in the Edutella System. However, these solutions do not lead to the possibility of using a LO in different way. In [11] is presented COFALE, a system to support flexible learning. The system approaches problem-based learning by allowing an adaptable presentation of learning contents, pedagogical resources and generation of evaluations. Nevertheless, these systems do not include search and discovery knowledge mechanisms.

6. Conclusions

PBL is a constructivist learning process that requires knowledge searching and discovering, though historically knowledge discovery has not attracted too much attention in VLE design. We have outlined a mechanism for guiding learners to find a solution of the problem at hand by means of VLE ontologies. We show how a semantic model can improve searching and discovery of knowledge (represented as LO according to the SCORM standard). The ontology defines the vocabulary of the problem domain and a set of constraints on how the terms can be combined to model the domain. The search algorithm is included in

RIbONTOMiddleware, its use has demonstrated that a retrieval mechanism based on context ontologies reduce significantly the links amount that students should navigate with respect to the results obtained from traditional search. The retrieved information is more useful for students, and diminishes their cognitive overload. In order to guide them through problem solving, we developed EnEMOCi VLE which implements a methodology for knowledge searching and publishing. In the near future, we plan to include more domain ontologies and to improve the user interface.

References

- [1] Protégé. <http://protege.stanford.edu/>.
- [2] Scorm. <http://www.adlnet.gov/scorm/>.
- [3] Wordnet. <http://wordnet.princeton.edu/>.
- [4] H. Barrows. *Practice-Based Learning: Problem-Based Learning Applied to Medical Education*. Southern Illinois University School of Medicine, 1994.
- [5] B. S. Bloom and D. R. Krathwohl. *Taxonomy of Educational Objectives: The Classification of Educational Goals. Handbook I: Cognitive Domain*. Longmans, New York, 1956.
- [6] D. Boud and G. I. Feletti, editors. *The Challenge of Problem-Based Learning. 2nd Edition*. Kogan Page Limited, London, 1997.
- [7] J. Brase and W. Nejdl. Ontologies and metadata for e-learning. In S. Staab and R. Studer, editors, *Handbook on Ontologies*, International Handbooks on Information Systems, pages 555–574. Springer, 2004.
- [8] J. Breuker and B. Bredeweg. Ontological modelling for designing educational systems. In *AI-ED 99 Workshop on Ontologies for Intelligent Educational Systems*, 1999.
- [9] T. Browne. A longitudinal perspective regarding the use of VLEs by higher education institutions in the United Kingdom. *Interactive Learning Environments*, 14:177–192(16), August 2006.
- [10] P. Chandler and J. Sweller. Cognitive load theory and the format of instruction. *Cognition and Instruction*, 8(4):293–332, 1991.
- [11] V. M. Chieu. Cofale: An authoring system for supporting cognitive flexibility. In *ICALT '06: Proceedings of the Sixth IEEE International Conference on Advanced Learning Technologies*, pages 335–339, Washington, DC, USA, 2006. IEEE Computer Society.
- [12] E. B. Cohen and M. Nycz. Learning objects and e-learning: An informing science perspective. *Interdisciplinary Journal of Knowledge and Learning Objects*, 2:23–34, 2006.
- [13] P. Dillenbourg, M. Baker, A. Blaye, and C. O'Malley. The evolution of research on collaborative learning. In E. Spada and P. Reiman, editors, *Learning human and machine: towards an interdisciplinary learning science*, pages 189–211. Oxford: Elsevier, 1995.
- [14] E. Duval and W. Hodgins. A LOM research agenda. In *WWW (Alternate Paper Tracks)*, 2003.
- [15] N. Friesen. Interoperability and learning objects: An overview of e-learning standardization. *Interdisciplinary Journal of Knowledge and Learning Objects*, 1:23–31, 2005.
- [16] T. R. Gruber. A translation approach to portable ontology specifications. *Knowledge Acquisition*, 5(2):199–220, June 1993.
- [17] T. R. Gruber. Toward principles for the design of ontologies used for knowledge sharing? *Int. J. Hum.-Comput. Stud.*, 43(5-6):907–928, 1995.
- [18] N. Guarino and R. Poli. Formal ontology in conceptual analysis and knowledge representation. *Guarino, N. and Poli, R. (eds.) Formal Ontology in Conceptual Analysis and Knowledge Representation. Special issue of the International Journal of Human and Computer Studies*, vol. 43 n. 5/6, Academic Press., 1995.
- [19] IEEE Learning Technology Standards Committee. IEEE standard for learning object metadata (draft). IEEE standard 1484.12.1, 2002.
- [20] C. Knight, D. Gasevic, and G. Richards. An ontology-based framework for bridging learning design and learning content. *Educational Technology & Society*, 9(1):23–37, 2006.
- [21] D. McGuinness and F. van Harmelen. *OWL Web Ontology Language Overview*. W3C Recommendation, 2004.
- [22] P. Mohan and B. K. Daniel. A new distance education model for the University of the West Indies: A learning objects' approach. In *ICALT '04: Proceedings of the IEEE International Conference on Advanced Learning Technologies*, pages 938–942, Washington, DC, USA, 2004. IEEE Computer Society.
- [23] P. Mustaro and I. Silveira. Learning objects: Adaptive retrieval through learning style. *Interdisciplinary Journal of Knowledge and Learning Objects*, 2:35–46, 2006.
- [24] D. Serain. *Middleware*. Springer-Verlag, 1999.
- [25] P. B. A. Smits, J. H. A. M. Verbeek, and C. D. de Buissonje. Problem based learning in continuing medical education: a review of controlled evaluation studies. *British Medical Journal*, 324(7330):153–156, 2002.
- [26] C. Snae and M. Brueckner. Ontology-driven e-learning system based on roles and activities for thai learning environment. *Interdisciplinary Journal of Knowledge and Learning Objects*, 3:1–17, 2007.
- [27] K. Verbert, D. Gašević, J. Jovanović, and E. Duval. Ontology-based learning content repurposing. In *WWW '05: Special interest tracks and posters of the 14th international conference on World Wide Web*, pages 1140–1141, New York, NY, USA, 2005. ACM.
- [28] D. A. Wiley. Connecting learning objects to instructional design theory: A definition, a metaphor, and a taxonomy. In *The Instructional Use of Learning Objects: Online Version*, 2000.
- [29] K. Yordanova. Meta-data application in development, exchange and delivery of digital reusable learning content. *Interdisciplinary Journal of Knowledge and Learning Objects*, 3:229–337, 2007.
- [30] H. Zhuge and Y. Li. Active e-course for constructivist learning. In *WWW Alt. '04: Proceedings of the 13th international World Wide Web conference on Alternate track papers & posters*, pages 246–247, New York, NY, USA, 2004. ACM.
- [31] A. Zouaq, R. Nkambou, and C. Frasson. An integrated approach for automatic aggregation of learning knowledge objects. *Interdisciplinary Journal of Knowledge and Learning Objects*, 3:135–162, 2007.

Natural Language Processing

Syntactico-Semantic Analysis: a Hybrid Sentence Extraction Strategy for Automatic Summarization

Jorge García Flores
IRSN-DESTQ/DICST,
31, avenue de la Division Leclerc
92260 Fontenay aux Roses, France
jorge.garcia-flores@irsn.fr

Gaël de Chalendar
CEA LIST / Laboratoire d'Ingénierie
de la Connaissance Multimedia Multilingue
18, route du Panorama
92265 Fontenay aux Roses, France
gael.de-chalendar@cea.fr

Abstract

Automatic summarization systems often make use of sentence extraction methods to select significant content in texts. This paper presents two sentence extraction strategies. The first one is based on a semantic analysis using word senses built by means of a network of lexical co-occurrences. The second one uses a combination of semantic and syntactical analysis to extract a set of relevant sentences. Both strategies are evaluated against a large corpus of newspaper and scientific articles, and their results are compared to Mead, a reference summarization tool. Evaluations results suggest that, inside the extractive summarization framework, the proposed hybrid strategy improves the relevance of the resulting summaries.

1. Introduction

To take a text as an input, to extract important content from it, to condense this content and to produce a reduced text: this is the canonical definition of automatic summarization [11]. This definition raises a fundamental question for any kind of summarizer (human or computer): what can be considered as *important content*? Extractive approaches of automatic summarization suppose that meaning is concentrated in certain textual units: it would then be possible to produce a condensed version of the source text made from its most significant units.

Extraction has been used since the earlier years of automatic summarization [3], and extraction engines are critical components of both single and multi-document summarizers. The main textual unit for extractive summarizers is the sentence. As Mani points out, sentence extraction allows a better control over compression than paragraphs or phrases, and historically the sentence has been the central analysis

unit in linguistic studies. Therefore, at a sentence-level, the above question can be reformulated as: How to assess sentence relevance? How to measure the significance weight of a sentence?

questions. Basic statistical extractive approaches rely on the hypothesis that meaning is a numerical function of words frequencies (the *tf*idf* hypothesis). For Spärck Jones, this is a poor representation of meaning: "Clearly, as characterizations of source and summary text meaning, the representations used are weak and indirect, with sentences treated independently and their meaning as a numerical function of their component word frequencies." [14]. The underlying idea of this approach is that relevance can be assessed in an endotextual way (exclusively from intrinsic text elements [5]), which is a very tempting assumption from a computational perspective, but certainly simplistic. Meaning is also context: syntactic, semantic, discursive, pragmatic context. Comparative evaluations suggest that enriched statistical approaches (enriched by linguistic, contextual, exotextual analysis) produce more relevant extracts [8].

This paper presents a comparative evaluation of two enriched statistical approaches: Morcas.1 and Morcas.2. The long term goal of the Morcas project is to develop an opinion oriented summarization system, where automatic summaries reflect not only the most relevant topics from the source text, but also the author's position on these topics. Therefore, Morcas.1 [2] improved the *tf*idf* hypothesis with a word sense identification method. Instead of assessing relevance based on words frequencies, Morcas.1 makes use of semantically stable sets of terms considered as words *senses*. These term-sets are induced from a lexico-semantic cooccurrence network built from a large corpus of newspaper articles [4]. Salient senses are then used to weight each sentence and extract those with the highest *semantic density*. Our work presents Morcas.2, where Morcas.1 se-

semantic relevance assessment is completed by the analysis of nominal dependencies (syntactic dependencies indicating noun phrases). The underlying hypothesis is that noun phrases with a high number of intraphrastic syntactic relations are indicators of a relevant concept. The rationale of this hypothesis was to improve a method that relies on corpus-dependent criteria (words senses identification) with linguistic, corpus independent criteria (syntactic dependencies). The approach is hybrid in the sense of the hybridization of a statistical approach (words senses are built through a statistical analysis of a corpus) and a symbolic one (rules-based syntactic analysis).

The remainder of the paper is organized as follows: Section 2 presents an extractive approach based on words senses identification. Section 2.1 shows the results of an evaluation of this approach. Section 3 discusses the hypothesis that an hybrid strategy of words senses identification and syntactic analysis will improve the relevance of the extraction. Section 3.1 shows a comparative evaluation of Morcas.1 and Morcas.2 strategies. Finally, Section 4 summarizes conclusions and further work.

2. Morcas.1: Extractive summarization through semantic analysis

Morcas was originally developed for multilingual cross summarization [2]. After semantic selection, its first prototype used to perform sentence compression by identifying (and taking away) less relevant syntactic dependencies [6]. Afterwards, a generation module tried to produce new text by means of a language independent algorithm. The goal of this first Morcas prototype was to produce cross-lingual summarization. Because of the complexity of the algorithmic pipeline, we decided to “isolate” each of Morcas modules and to evaluate them separately in order to estimate the real impact of each element in the pipeline. That’s why, for the present work, we separated the sentence extraction engine: that’s what we call Morcas.1

Morcas.1 first step is the morphosyntactic and dependency analysis (see Figure 1). This analysis is needed because word senses are lemmatized in the cooccurrence network, so the source text elements must be lemmatized as well. Morphosyntactic analysis is performed by LIMA (Lic2M Multilingual Analyzer) [1], the multi-language linguistic analyzer developed in our team. LIMA identifies sentences, paragraphs and phrases from the source text, and produces a morphosyntactic graph. This “textual tree” generated by LIMA contains morphological categories, syntactic dependencies and, for French, named entities, dates and acronyms. This representation of text, where each sentence is annotated with such a rich amount of linguistic information, is a good ground to look for hybrid representations of meaning, where linguistic criteria enriches the *tf*idf* hy-

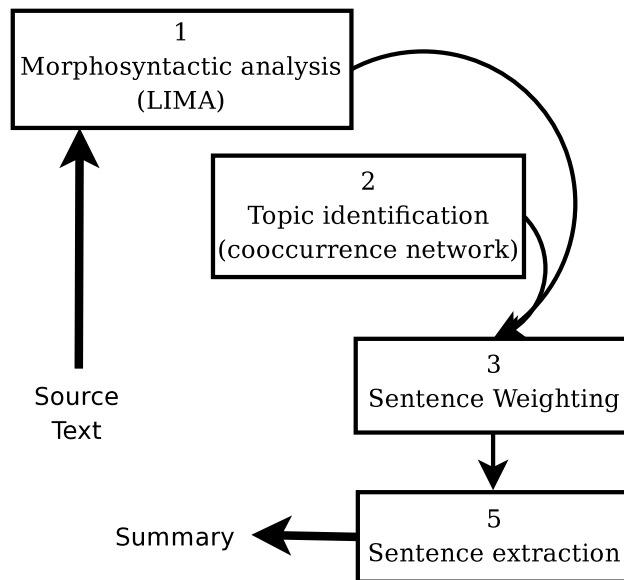


Figure 1. Morcas.1 pipeline

pothesis.

The second step is semantic selection. As previously said, Morcas.1 relevance assessment is based on the words senses frequencies: instead of counting single words, Morcas.1 counts occurrences of terms included in a semantically stable set defining a word sense. For Morcas.1, these words senses determine sentence relevance. In order to assess relevance, Morcas.1 makes use of a network of lexical cooccurrences. This network is the base of a word sense discovery algorithm developed by [4]. The goal of this algorithm was to automatically distinguish the various senses of a word by a cooccurrence analysis from a very large corpus. The underlying hypothesis is that textual cooccurrence is a sort of mutual definition between cooccurring words. The result is a lexico-semantic network that gathers cooccurrent terms according to their similarity or dissimilarity, as illustrated in Table 1. For French texts, Morcas.1 uses a network made from two years of *Le Monde* journal. The use of a lexico-semantic network enriches the *tf*idf* hypothesis in a way that the frequency criteria is subordinated to the topical pertinence of the counted terms. However, the terms to be counted depend on the nature of the corpus used to build the network. Our evaluation tries to determinate to what extent the relevance of the summary depend on the network’s corpora.

The following formulae describe in detail the Morcas.1 relevance assessment process, as it was first developed by [2]. Given a word lemma w , $frequency(w)$ is the number of times that w appears in the source text. $\{S_{w,1}, S_{w,2}, \dots, S_{w,n}\}$ represents possible senses of w provided by the cooccurrence network. For instance,

eau, mètre, lac, pluie, rivière, bassin, fleuve, site, poisson, affluent, montagne, crue, vallée (*water, meter, lake, rain, river*(2), *basin, setting, fish, affluent, mountain, swelling, valley*)

conducteur, trafic, routier, route, camion, chauffeur, voiture, chauffeur routier, poids lourd (*driver, traffic, lorry driver*(3), *road, lorry, car, truck*)

casque bleu, soldat, tir, convoi, milicien, blindé, milice, aéroport, blessé, incident, croate (*U.N. soldier, soldier, firing, convoy, militiaman, tank, militia, airport, wounded, incident, Croatian*)

Table 1. Senses found by Ferret’s senses discovery algorithm [4]

the word *mouse* has two senses in Ferret’s network: {*compatible, software, computer, machine, user, desktop,...*} and {*laboratory, researcher, cell, gene, generic,...*}. Instead of counting the frequency of the word *mouse*, Morcas.1 will count the frequency of each term (*u*, or the *definiens*) included in each senses of *w* (the *definiendum*: the *mouse* of our example):

$$frequency(S_{w,i}) = \sum_{u \in S_{w,i}} frequency(u)$$

The relevance (or the weight) of a certain word sense $S_{w,i}$ is computed by normalizing the frequency of each of its sense against the frequency of all the word’s senses found in the source text:

$$relevance(S_{w,i}) = \frac{\sum_{u \in S_{w,i}} frequency(u)}{\sum_{t \in \{source\_text\}} \sum_{x \in S_t} frequency(x)}$$

Therefore, the relevance of *w* will be the relevance of its weightiest (more frequent) sense.

$$relevance(w) = \arg\max_i relevance(S_{w,i})$$

...and the sentence relevance is the sum of the relevance of each word, normalized against the sentence size:

$$relevance(sentence) = \frac{\sum_{w \in sentence} relevance(w)}{|\{w \in sentence\}|}$$

It is interesting to observe that:

- Word *w* is the lemmatized form of a noun, an adjective or a verb.

- $S_{w,i}$ is certainly determined by the cooccurrence network input corpora, which must be as large and as diverse as possible [5].
- Further improvements of the network could include a more sophisticated relevance assessment for each word, which takes into account all of word’s senses, an not only the more frequent (argmax).

2.1. Morcas.1 evaluation

The goal of the evaluation was: i) to compare Morcas.1 extraction strategy against a mainstream summarizer; ii) to assess the impact of the document type on a corpus dependant strategy (the cooccurrences network was built from two years of *Le Monde* newspaper), and iii) to base further enrichment of Morcas.1 on the evaluation conclusions

The corpus used for Morcas.1 evaluation was made from a sample of IRSN (*Institut pour la Radioprotection et la Sûreté Nucleaire*) documentary base of nuclear energy related documents. The size of the corpus was around 300,000 words. The distribution according to the document genre was: 50% of scientific articles, 25% of newspaper articles, 15% of enterprise reports and 10% of PhD thesis. All of this documents are in French.

In evaluations campaigns, like DUC, text use to be homogeneous: same genre (newswire text), same size, almost the same structures. The IRSN document database presents the following variations:

- Size variations: from 400 pages PhD thesis to one page scientific newswire releases.
- Structure variations: from highly structured scientific papers to the loose paragraph structure of a meeting report.
- Linguistic register: from scientific objectivity to editorial subjectivity.
- Language variations: English, French and Russian (the scope of this evaluation is limited to French).

An automatic evaluation was performed using ROUGE [10]. Each document from the corpus had a human-written model summary. ROUGE calculates n-gram lexical overlap between the automatic and the human summary. The measures used were R1 (n-gram=1 word), R2 (bigrams), R3 (trigrams), R4 (quadgram) and RL (longest common word). The human summaries were written by IRSN domain experts.

In [10], Lin suggests that there is a possible correlation between human judgments and ROUGE measures, and ROUGE evaluations are relatively faster and cheaper than human ones. However, n-gram overlap might punish extractive approaches because it doesn’t takes into account

Evaluation on IRSN corpus				
		Full Text	Morcas1	Mead
ROUGE1	recall	45.21%	36.83%	35.20%
	precision	43.01%	37.86%	35.64%
	f-measure	44.04%	37.20%	35.18%
ROUGE2	recall	21.48%	7.49%	10.37%
	precision	20.44%	7.70%	10.35%
	f-measure	20.93%	7.56%	10.30%
ROUGE3	recall	14.11%	2.36%	5.29%
	precision	13.42%	2.47%	5.24%
	f-measure	13.75%	2.40%	5.24%
ROUGE4	recall	10.36%	0.83%	3.56%
	precision	9.86%	0.88%	3.50%
	f-measure	10.10%	0.85%	3.52%
ROUGEL	recall	43.84%	20.83%	25.84%
	precision	42.35%	21.40%	26.21%
	f-measure	43.03%	21.03%	25.82%

Table 2. Comparative evaluation of Morcas.1

linguistic reformulation and paraphrases performed by human summarizers (see [11] for human summaries and [13] for ROUGE limitations). Inspired by [9], we used the full

	MORCAS		MEAD	
document type	R1	R2	R1	R2
newspaper article	49.0%	10.0%	36.1%	8.3%
scientific article	30.9%	7.5%	34.3%	11.3%
enterprise report	40.8%	8.7%	37.4%	10.0%
PhD thesis	27.8%	3.7%	33.1%	10.5%

Table 3. Automatic evaluation results according to document type (better note 20/20)

text as a baseline, and for comparison purposes we adapted a version of the Mead summarizer [12] to French. Table 2 shows ROUGE scores for Morcas.1, Mead and the full text. In the automatic evaluation results we can observe that Morcas.1 gets a better score with short n-grams, but as the n-gram grows, Morcas.1 score falls, while Mead score remains high because in Mead’s ranking criteria, longer sentences are considered as more relevant, while in Morcas criteria the sentence weight is normalized against the sentence size (measured in lexico-semantic units: lemmas from the lexico-semantic network found in the sentence). In consequence, short (but semantically dense) sentences will get a better rank in Morcas.1

Table 3 shows ROUGE-1 and ROUGE-2 recall measures classified by document type. This table shows that Morcas.1 strategy gets a better overlap score with newspaper articles and enterprise reports than with scientific documents. From this fact we can conclude that the input corpus of

the cooccurrences network has a strong influence in Morcas.1 relevance assessment: both the network and newspaper articles share the same language register, which is not the case for the specialty language of scientific articles. There are two ways to balance this influence: to include corpus-independent criteria (like syntax or discourse based relevance, see next section) and by making a new cooccurrences network which captures the lexical field of nuclear energy domain.

The automatic evaluation performed by ROUGE measures the presence of important information in the summaries but not their linguistic quality and their understandability. It was thus important to do also a human evaluation of various linguistic characteristics. Due to the lack of resources to do a large scale human evaluation, we asked one people not involved in this work to note some characteristics of the summaries. In the results of the human judged evaluation (Table 4) we can observe the main shortcoming of extractive approaches: the lack of discursive cohesion [14]. Anaphora resolution abstraction, and generalization are needed in order to produce cohesive summaries. This particular defect is better observed at high compression rates, where both Morcas and Mead get poor notes for long documents summarization.

	MORCAS	newspaper articles	scientific articles	enterprise reports	PhD thesis
coherence		17.5	8.9	8.8	2.5
precision		16.3	7.8	6.3	2.5
redundancy		17.5	10.6	12.6	7.5
references		10.0	11.1	11.3	2.5
grammar		17.5	13.3	13.8	7.5
format		17.5	14.4	10.0	2.5
saliency		11.3	6.1	5.0	2.5
general		10.0	8.3	8.6	2.5
		14.7	10.1	9.5	3.8
	MEAD	newspaper articles	scientific articles	enterprise reports	PhD thesis
coherence		20.0	12.2	11.3	5.0
precision		18.8	12.2	10.0	2.5
redundancy		17.5	15.0	15.0	2.5
references		18.8	13.9	13.8	7.5
grammar		20.0	18.3	20.0	15.5
format		1.2	3.9	2.5	2.5
saliency		17.5	12.2	11.2	5.0
general		17.5	10.6	11.3	2.5
		16.4	12.3	11.9	5.3

Table 4. Human evaluation results according to document genre

3. Morcas.2: Enriching Morcas.1 with nominal dependencies

From Morcas.1 evaluation, we concluded that: i) it was necessary to balance Morcas.1 corpus dependant relevance assessment with corpus independent criteria, ii) a tuning factor was necessary to balance Morcas.1 preference for short sentences, iii) because of task motivated reasons (Morcas will support information research in the IRSN document database, rich in scientific documents), Morcas.2 will need a specific treatment of specialty language.

In consequence, we formulated the following hypothesis: between the most semantically significant sentences, those with the highest amount of nominal dependencies are the most relevant. Nominal dependencies are noun phrase constructors: syntactic relation established at an intra-phrastic level of a noun phrase. Table 1 shows the LIMA syntactic dependencies that we use as a tuning factor for Morcas.2 relevance assessment. As it is based on grammatical crite-

Dependency	Description	Example(fr)
COMPDUNOM	noun's complement	capitale de la République
SUBADJPOST	adjective after a noun	politique économique
ADJPRENSUB	adjective before a noun	principale artère
SUBSUBJUX	noun juxtaposition	six cent adeptes
ATB_S	subject's attribute	cyberdissident emprisonné

Table 5. Nominal dependencies indicating conceptual density

ria, our hypothesis is mainly corpus-independent. However, it is motivated by the fact that the syntactic structure of scientific language is characterized by a high use of nominal dependencies [7]. To tackle Morcas.1 preference of short sentences, we use the number of nominal dependencies as a tuning factor for the relevance assessment formula.

$relevance(sentence) =$

$$\frac{\sum_{w \in sentence} relevance(w)}{|w \in sentence|} * nominal\_deps(sentence)$$

Figure 2 shows Morcas.2 pipeline. After the morphosyntactic analysis, an XSLT engine counts nominal dependencies from the source text graph, in order to include this parameter during the sentence selection process (sentence weighting + sentence extraction). Given that syntactic dependencies are proportional to the sentence length, the new tuning

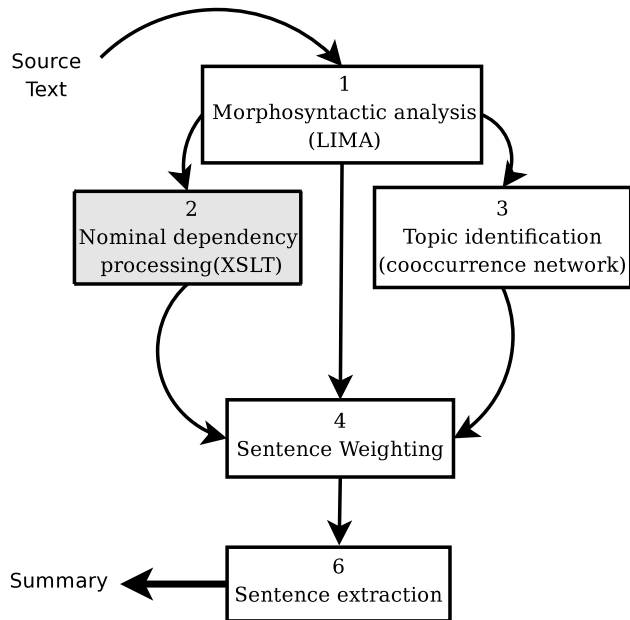


Figure 2. Morcas.1 pipeline

parameter of Morcas.2 will balance Morcas.1 short sentence preference.

3.1. Morcas.2 evaluation

For the evaluation of Morcas.2 we used the same corpus than for the evaluation of Morcas.1 (section 2). However, because of time limitations, the human judges data was not available, so we present only the results of the automatic evaluation using ROUGE (see Table 6). The first evaluation results show a significant improvement in Morcas.2 R1 and R2 scores, compared both to Morcas.1 and to Mead. R2 improvement is particularly significant, because it implies that nominal dependencies tuning had an impact on the length of the selected sentences.

At the time of writing this article we still don't have detailed information about each document type score, but higher R1 and R2 overlap suggests that scientific articles score might be higher. However, this results must be analyzed together with the question-set results from human judges.

4. Conclusion

In this paper we presented two sentence extraction strategies: Morcas.1, based on semantic analysis using word senses, and Morcas.2, where the semantic criteria is enriched by syntactic analysis. We evaluated both strategies against a corpus made of scientific articles, enterprise reports and newspaper articles in French. We compared the results of the evaluation with a French adapted version of a

Evaluation on IRSN corpus

		Morcas1	Morcas2	Mead
ROUGE1	recall	36.83%	40.50%	35.20%
	precision	37.86%	40.88%	35.64%
	f-measure	37.20%	40.65%	35.18%
ROUGE2	recall	7.49%	11.34%	10.37%
	precision	7.70%	11.48%	10.35%
	f-measure	7.56%	11.40%	10.30%
ROUGE3	recall	2.36%	5.01%	5.29%
	precision	2.47%	5.06%	5.24%
	f-measure	2.40%	5.03%	5.24%
ROUGE4	recall	0.83%	3.35%	3.56%
	precision	0.88%	3.37%	3.50%
	f-measure	0.85%	3.35%	3.52%
ROUGEL	recall	20.83%	23.67%	25.84%
	precision	21.40%	24.01%	26.21%
	f-measure	21.03%	23.82%	25.82%

Table 6. Comparative evaluation of Morcas.1, Morcas.2 and Mead

mainstream automatic summarizer: Mead [12].

From the evaluation of Morcas.1 we concluded that the relevance assessment strategy was corpus-dependant. More precisely, Morcas.1 gets better scores with newswire text because its relevance assessment algorithm depends on a lexico-semantic network built from two years of *Le Monde* journal. The evaluation showed us that Morcas.1 weighting strategy preferred short, semantically dense sentences, and that it performed better with general language documents (newspaper articles and enterprise reports) than with specialized language (scientific articles).

Morcas.2 modifies Morcas.1 relevance assessment strategy by taking into account a grammatical hypothesis: between semantically significant sentences, those with a high amount of nominal dependencies (syntactic dependencies established within the limits of a noun phrase) are more relevant.

A second evaluation showed us that the nominal dependency hypothesis improved Morcas.2 scores. These results suggest that, inside the extractive summarization framework, hybrid methods (both statistical and symbolic) are a good alternative to pure statistical relevance assessment strategies. They also suggest that language register (general versus specialty language) has an influence on the relevance of extracted sentences. However, task-specific evaluation is needed to confirm this result. The human evaluation clearly shows that an the extractive summarization approach is not enough for certain task (or certain type of texts, like long documents), so more research on abstraction, generalization and text generation is needed.

Further work will include: i) developing a lexico-semantic

cooccurrence network specific to the nuclear energy domain; ii) studying another type of syntactic relations (verbal dependencies) and their relation to relevance assessment; iii) a linguistic study of opinion stating expressions in order to produce opinion oriented summaries; iv) the introduction of other linguistically motivated methods to go beyond pure extraction and improve the summaries linguistic quality.

References

- [1] G. de Chalendar and R. Besancon. Lanalyseur syntaxique de lima dans la campagne d'évaluation easy. In *Actes de la 12eme conférence sur le Traitement Automatique des Langues, TALN 2005*, 2005.
- [2] G. de Chalendar, R. Besancon, O. Ferret, G. Grefenstette, and O. Mesnard. Thematic extraction, syntactic sentence simplification and bilingual generation towards crosslingual summarization. In *Proceedings of International Conference on Recent Advances in Natural Language Processing (RANLP)*, 2005.
- [3] H. P. Edmundson. New methods in automatic extracting. *J. ACM*, 16(2):264–285, 1969.
- [4] O. Ferret. Discovering word senses from a network of lexical cooccurrences. In *COLING '04: Proceedings of the 20th international conference on Computational Linguistics*, page 1326, Morristown, NJ, USA, 2004. Association for Computational Linguistics.
- [5] O. Ferret. Approches endogène et exogène pour améliorer la segmentation thématique de documents. *TAL*, 47(2/06):111–135, 2006.
- [6] G. Grefenstette. Producing intelligent tele-graphic text reduction to provide an audio scanning service for the blind. In *Intelligent Text Summarization, AAAI Spring Symposium Series*, pages 111–117, 1998.
- [7] M. Halliday and C. Matthiessen. *Construing Experience through Meaning: A Language-based approach to Cognition*. London Casell, 1999.
- [8] S. Harabagiu and F. Lacatusu. Topic themes for multi-document summarization. In *SIGIR '05: Proceedings of the 28th annual international ACM SIGIR conference on Research and development in information retrieval*, pages 202–209, New York, NY, USA, 2005. ACM.
- [9] S. P. Hobson, B. J. Dorr, C. Monz, and R. Schwartz. Task-based evaluation of text summarization using relevance prediction. *Inf. Process. Manage.*, 43(6):1482–1499, 2007.
- [10] C.-Y. Lin. Rouge: Recall-oriented understudy for gisting evaluation. <http://www.isi.edu/cyl/ROUGE/>, 2003.
- [11] I. Mani. *Automatic Summarization*. John Benjamins Publishing Company., 2001.
- [12] D. R. Radev, H. Jing, M. Styś, and D. Tam. Centroid-based summarization of multiple documents. *Inf. Process. Manage.*, 40(6):919–938, 2004.
- [13] J. Sjöbergh. Older versions of the rougeeval summarization evaluation system were easier to fool. *Inf. Process. Manage.*, 43(6):1500–1505, 2007.
- [14] K. Spärck Jones. Automatic summarising: The state of the art. *Information Processing & Management*, 43(6):1449–1481, Nov. 2007.

Single Document Summarization Based on Local Topic Identification and Word Frequency

Zhi Teng, Ye Liu, Fuji Ren and Seiji Tsuchiya
Faculty of Engineering
The University of Tokushima
2-1 Minamijosanjima Tokushima, Japan 770-8506
{teng, liuye, ren, tsuchiya}@is.tokushima-u.ac.jp

Fuji Ren
School of Information Engineering
Beijing University of Posts
and Telecommunications
Beijing, China 100876

Abstract

In this task, an approach for single document summaries based on local topic identification and word frequency is proposed. In recent years, there has been increased interest in automatic summarization. The physical features are often used and have been successfully applied to this field; it also has some disadvantages of non-redundancy, structure and coherence. Therefore, we introduced logical structure feature which has been successfully applied in multi-document summarization (MDS), and we designed a system to accomplish this task. Documents can be clustered into local topic after sentences similarity is calculated, which can be sorted by the scoring. Then sentences from all local topics are selected by computing the word frequency. Using this proposed method, the information redundancy of each local topic and among local topic is reduced. The information coverage ratio and structure of the summarization is improved.

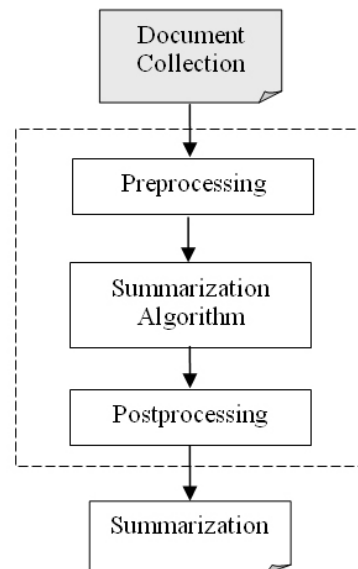


Figure 1. Summarization System Architecture

1. Introduction

Text summarization is a product of electronic document explosion, and can be seen as the condensation of the document collection. The use of text summarization allows a user to get a sense of the content of a full-text, or to know its information content, without reading all sentences within the full-text. Data reduction increases scale by allowing users to find relevant full-text sources more quickly, and assimilating only essential information from many texts with reduced effort [1]. Generic summarization system is divided into three modules: text preprocessing, summarization algorithm, and postprocessing. In Figure 1.

2. Related Work

We mainly investigate previous text summarization work related to local topic and frequency methodologies because our work largely involves these two approaches.

Word frequency, Summarization using larger units of text has also been researched. Most recently, the SumBasic algorithm uses term frequency as part of a context-sensitive approach to identifying important sentences while reducing information redundancy [2]. The use of frequency as a feature in locating important areas of a text has been proven useful in the literature [3]. This is most likely due to reiteration, where authors state important information in several

different ways, in order to reinforce main points [4]. Local topic identification approach, Qin [5] did a multi-document summarization experiment based on local topic identification and extraction. The similarity of sentences is measured by analysis of dependency and semanteme.

Local topic is found by sentence clustering. The centroid sentence is extracted from each local topic and is ordered to generate summarization. Zhang [6] proposed a method for representing and summarizing documents by integrating subtopic partition with graph representation. The method starts from the assumption that capturing sub-topic structure of document collection is essential for summarization.

In the word frequency research, everyone interest in how to find out good physical features arithmetic, did not attention structure and coherence. In the local topic identification research, everyone usually interest in how to improve the non-redundancy and coherence used structure feature. Thus, we did an experiment by consideration on physical features and logical structure feature (see Section 3 for details).

3. Summarization Methods

3.1. Local Topic Identification

Text summarization is a data reduction process. As summary is concise, accurate and explicit, it becomes more and more important. One document can be composed of some subdocuments. The described contents in each of the documents laid special emphasis on different aspects although these documents were all surround the same topic. It is obvious that document is composed of different side information and the different side information is the local topic.

In terms of the logical structure, a single-document set can also be considered as a set of local topics, a semantic paragraph comprises of similarity sentences, and each semantic paragraph is called a local topic. This approach can deal with the redundancy and improve the structure and coherence in the large documents.

In this task, after the similarity of sentences is measured by the Vector Space Model (VSM) method, the local topic is found by sentence clustering. The VSM has been one of the most successful models in various performance evaluation studies [9]. Most existing search engines and information retrieval systems are based on this model.

In this model, both training data and testing data are represented as vectors of index terms. Suppose there are a total of t index terms in an entire collection, then a given document D and query Q can be represented as follows:

$$\begin{aligned} D &= (W_{d1}, W_{d2}, \dots, W_{dt}) \\ Q &= (W_{q1}, W_{q2}, \dots, W_{qt}) \end{aligned}$$

where W_{di} , W_{qi} ($i = 1$ to t) are term weights assigned to different terms for the document D and query Q , respectively. The similarity between a query and a document can be calculated by the widely used cosine measure [10]:

$$Sim(Q, D) = \frac{\sum_{i=1}^t W_{qi} \times W_{di}}{\sqrt{\sum_{i=1}^t (W_{qi})^2 \times \sum_{i=1}^t (W_{di})^2}} \quad (1)$$

Documents then are ordered by decreasing values of this measure.

The art of designing ranking functions depends on the design of term weighting strategies to assign weights for terms in a document, W_{di} . Different term weighting strategies influence the similarity that is computed according to Formula 1.

Here we defined a threshold value named η . After the similarity measurement, we incorporate the sentences in the same local topics if the similarity value of sentence is larger than η . The optimal value of η , 0.59, was empirically determined using the similarity values of 2000 pairs of sentences.

3.2. Local Topic Score

After local topic identification processing, we have to compute and order each local topic, cycled pick the best sentence from the best local topic if the desired summary length has not been reached. The core system is SumBasic [2]. The SumBasic is a generic algorithm; it did not include other feature or information, therefore, we improved the SumBasic by sentence location feature and SumFocus method.

3.2.1 SumFocus

SumFocus made by Lucy Vanderwende [7], a new approach in the multi-document summarization system, captures the information conveyed by the topic description by computing the word probabilities of the topic description. In our experiment corpus every text includes news topic and content. The weight for each word is computed as a linear combination of the unigram probabilities derived from the topic description, with back off smoothing to assign words not appearing in the topic a very small probability, and the unigram probabilities from the document, in the following manner:

$$WordWeight = (1-\lambda) \times DocWeight + \lambda \times TopicWeight \quad (2)$$

The optimal value of λ , 0.9, was empirically determined using the DUC2005 corpus, manually optimizing on ROUGE-2 scores.

3.2.2 Sentence Location Feature

Sentence Location Feature is also important except words occurring frequently. According to the statistics, in Chinese news, the probability that the first sentence can be picked and make summarization is about 85%. The probability that the last sentence can be picked and make summarization is about 7% [8]. Thus we have to adjust the sentence weight that in the especial location. We use the following algorithm:

$$Weight(L_j) = (1 + \frac{(P - 0.75m)^2}{m^2}) \times ST_j \quad (3)$$

For each sentence L_j in the local topics, P is the serial number of sentence L_j in the document. m is the number of sentence in the document. ST_j is the similarity value between sentence L_j and title of the document.

3.3. Pick Sentences and Make Summarization

The improved SumBasic approach is described in the following manner:

Step1. Compute the probability distribution over the words W_i appearing in the document, $p(w_i)$ for every i used formula 2.

Step 2. For each sentence S_j in the local topics, compute the $Weight(S_j)$ and $Weight(L_j)$, calculate the scoring of each sentence and the scoring of each local topic. We used following algorithm:

$$Score(LocalTopic) = \sum (\alpha Weight(S_j) + \beta Weight(L_j)) \quad (4)$$

Here $Weight(S_j) = \sum_{w_i \in S_j} \frac{p(w_i)}{|\{w_i | w_i \in S_j\}|}$. We find out that we can get the best result when $\alpha = 0.9$ and $\beta = 0.1$ through some experimentations by used different α and β .

Step 3. Pick the best scoring sentence that contains the highest probability word from the best scoring local topic.

Step 4. Delete this local topic and all sentences in this local topic.

Step 5. For each word w_i in the sentence chosen at step 3, update their probability:

$$P_{new}(w_i) = P_{old}(w_i) \times P_{old}(w_i) \quad (5)$$

Step 6. If the desired summary length has not been reached, go back to Step 2.

Step 7. Use the sentences that picked up by Step 3 to generate summarization.

Step 2 ensures that the highest probability word is included in the summary; Step 3 gives the summarizer sensitivity to context and Step 5 gives a natural way to deal with the redundancy.

4. Evaluation

4.1. Corpus

A corpus of 45 news texts in Chinese was selected from a daily news database. The daily news database contains approximately 266 texts and all of them were downloaded from news website. Every news text includes a news topic and content. Of the 266 news texts, 45 texts were deliberately selected. The selection method is that:

Data 1: selected 15 texts which amount of sentence are not more than 10;

Data 2: selected 15 texts which amount of sentence between 10 and 20;

Data 3: selected 15 texts which amount of sentence are more than 20.

4.2. Vpercent and Hpercent

In order to evaluate our proposed approach, we used two evaluation criterion, *Vpercent* and *Hpercent* [8]. *Vpercent* is the coverage ratio of valid word (exclude stopwords). *Hpercent* is the coverage ratio of high frequency word (the valid word that appears more than twice). The measure is as follows:

$$Vpercent = \frac{SVWords}{TVWord} \times 100\% \quad (6)$$

$$Hpercent = \frac{SHWords}{THWord} \times 100\% \quad (7)$$

TVWord: the number of valid words in the original texts

THWord: the number of high frequent words in the original texts

SVWords: the number of valid words in the summarization

SHWords: the number of high frequent words in the summarization

4.3. NIST

The linguistic quality questions are targeted to assess how readable and fluent the summaries are, and they measure qualities of the summary that DO NOT involve comparison with a model summary or DUC topic. There are five NIST linguistic quality questions are relevant to automatic summaries: grammaticality, non-redundancy, referential clarity, focus and structure & coherence¹.

¹<http://www-nlpir.nist.gov/projects/duc/duc2007/quality-questions.txt>

Table 1. The Vpercent results based on Top-N and our system

	TOP-N	Our System
Data 1	32.98%	38.98%
Data 2	29.53%	39.84%
Data 3	25.87%	38.43%
Average	29.46%	39.08%

Table 2. The Hpercent results based on Top-N and our system

	TOP-N	Our System
Data 1	14.55%	28.17%
Data 2	15.72%	29.18%
Data 3	19.34%	28.12%
Average	16.54%	28.49%

NIST evaluated these questions by eliciting human judgments on a five point scale from "1" to "5", where "5" indicates that the summary is good with the respect to the quality under question, "1" indicates that the summary is bad with respect to the quality stated in the question, and "2" to "4" show the gradation in between.

5. Results

The experiment object is that we want to test the feasibility and validity of the proposed approach. To do this experiment we deliberately selected 45 news texts in Chinese from a daily news database. The summary length is not more than 20% words of original text.

To compare the result of our approach with existed methods, the Top-N method was used to build a reference summarization used the same disaster texts. The method of Top-N is that the first sentence in each paragraph in document is taken in turn until the number of sentences is satisfied. If the number of sentences is not satisfied, repeat the process on the second sentence of each paragraph.

5.1. Evaluation by Vpercent and Hpercent

In order to compute the *Vpercent* and *Hpercent*, we made a program to automatically evaluate the result. The Table 1 and Table 2 shows the coverage ratio of valid word and high frequency word based on Top-N and our approach.

5.2. Evaluation by NIST

In order to evaluate the NIST results, we invited 3 examinees who will assess the quality of the summary. Before the

Table 3. The NIST evaluation results

	Non-redundancy	
	Our System	TOP-N
Data 1	4.42	3.55
Data 2	4.20	3.18
Data 3	4.12	3.30
Average	4.26	3.34
	Structure and Coherence	
	Our System	TOP-N
Data 1	3.93	3.12
Data 2	3.53	2.60
Data 3	3.30	2.23
Average	3.59	2.65

examinees ask the question, they didn't know the anything of this system. Table 3 shows the results of NIST evaluation, the results of each test Data is the average value of 3 examinee's evaluation value.

5.3. Discussion

The *Vpercent* and *Hpercent* results show that the information coverage ratio of our approach is 39.08%, higher than compression ratios 20%, and 9.62% higher than Top-N method. The coverage ratio of high frequency word is 28.49%, higher than compression ratios 20%, and 11.95% higher than Top-N method. The NIST evaluation results show the Non-redundancy and Structure & Coherence higher than Top-N results. We can think that the local topics identification improved the Non-redundancy and Structure & Coherence; word frequency improved the coverage ratio of valid word and high frequency word. The results show that better performance is achieved when the two methods are combined in practice.

6. Conclusions

In this paper, we described our new approach for single document summaries based on local topics identification and word frequency. We assess the contribution of our approach to single document summaries. The *Vpercent*, *Hpercent* and NIST evaluation results show our approach undoubtedly contributes to non-redundancy and acquisition of better information coverage ratio.

We think we have to improve our system use sentence simplification and lexical expansion in the future.

The goal of sentence simplification is that produces summaries with as much content as possible that satisfies the user, given a set limit on length. We view sentence simplification as a means of creating more space within which to capture important content.

We believe the lexical expansions can be applied at the point where we choose the "best words" with which to determine local topic and sentence selection.

7. Acknowledgments

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References

- [1] Lawrence H. Reeve, Hyoil Han and Ari D. Brooks. 2007. The use of domain-specific concepts in biomedical text summarization. *Information Processing and Management*, 43 (2007) 1765-1776.
- [2] Nenkova, A. and Vanderwende, L. 2005. The impact of frequency on summarization. No. MSR-TR-2005-101. Redmond, Washington: Microsoft Research.
- [3] Hovy, E. and Lin, C. 1999. Automated text summarization in SUMMARIST. In I. Mani & M. T.
- [4] Sparck Jones, K. 1999. Automatic summarizing: Factors and directions. In I. Mani & M. T. Maybury (Eds.), *Advances in automatic text summarization* (pp. 2-12). Cambridge, MA: MIT Press.
- [5] Qin Bing, Liu Ting and Li Sheng. 2005. MDS based on sub-topic NCIRCS-2005, Beijing.
- [6] Jin Zhang, Hongbo Xu, Xiaolei Wang, Huawei Shen and Yiling Zeng. 2007. ICT CAS at DUC 2007. *DUC 2007*. April 26-27, 2007. New York USA.
- [7] Lucy Vanderwende, Hisami Suzuki, Chris Brockett and Ani Nenkova. 2007. Beyond SumBasic: Task-focused summarization with sentence simplification and lexical expansion. *Information Processing and Management*, 43 (2007) 1606-1618.
- [8] Qin Bing, Liu Ting Chen Shang-Lin and Li Sheng. 2006. Sentences Optimum Selection for Multi-document Summarization. *Journal of Computer Research and Development*, Vol.43, pp.1129-1134.
- [9] Harman, D. K. 1995. Overview of the fourth text retrieval conference (TREC-4). In D. K. Harman (Ed.), *Proceedings of the fourth text retrieval conference. NIST Special Publication*, 500-236 (pp. 1-24).
- [10] Salton, G. and Buckley, C. 1987. Term weighting approaches in automatic text retrieval. *Information Processing and Management*, 24(5), 513-523.

Two-Level Clausal Segmentation System Based on Semantic Information for Automatic Summarization

Kyonam Choo, Yoseob Woo, Hongki Min

Dept. of Info. and Telecomm. Engineering, University of Incheon, Korea Republic

{kyonam,yswooo,hkmin}@incheon.ac.kr

Abstract

This paper suggests a method that breaks a long sentence into simple sentences using not only syntactical information but also a sense of a word. Unlike uniformed sentences, common sentences are often ambiguous and frequently omit particles thus they should be divided into simple sentences at semantic level. This study shows the method that analyzes the dependency structure between predicate and complement using the information on sense from a subcategorization dictionary and thesaurus. Then it proposes a Two-Level Clausal Segmentation System (TLCSS) that includes other elements in the dependency structure gradually based on the syntactic information and grammatical knowledge.

1. Introduction

The significance of Clausal Segmentation is emphasized in many studies in the field of natural language processing. The method used for Korean Clausal Segmentation so far can be divided into a method that uses syntactical pattern, a method that uses syntactic analysis and collocation information, and a method that uses a subcategorization dictionary [1,2].

The method using syntactic patterns searches for the connection structure of the sentence. Then it defines the pattern including context information and segments the clause by the patterns. That is, this method divides a sentence after finding every possible area that can be split. However this method is unsatisfactory in processing the sentence that can be applied to more than two patterns. It also has a weakness in deconstructing adnominal clauses.

The method using syntactic analysis and collocation information is a statistical method. The case of a particle and collocation information is taken into account to remove the ambiguity of syntactic analysis. However the collocation information could be reliable when the information is extracted from large corpora rather than small corpora. In addition to this, this method segments the clause depending on the particle

extracted from the corpora of a specific area. Therefore when it is applied to corpora that have many delimiters and alternative particles the efficiency of the method deteriorates. Another problem is that it can not remove the syntactic ambiguity that could occur during Clausal Segmentation, because Korean sentences frequently omit the particles.

A method is required to overcome these problems of ambiguity and the limitation of Clausal Segmentation. This study suggests a Two-Level Clausal Segmentation System (TLCSS) that divides the clause using a large subcategorization dictionary with information on meanings, a thesaurus that has the concept of class, and features of the Korean language. The subcategorization dictionary and thesaurus make it possible to extract all possible candidates that have the complement, the head element of the predicate in the sentence. Among the candidates, the best candidate is chosen by the information of meanings from the subcategorization dictionary and the information of class from the thesaurus. In addition to this, a colloquial feature helps to extend the complement element. At the stage of post processing, this two-level method processes the remaining non-head elements.

2. Word Sense Information

This study utilizes the information of meanings from a subcategorization dictionary and thesaurus to disambiguate the process of Clausal Segmentation. The accuracy and the size of the subcategorization dictionary and thesaurus are significant in this study utilizing the proposed TLCSS method. However, it has space limitations when building information and for conducting the experimental procedure and testing [3]. This paper focuses on the structures and the information of the meanings of the subcategorization dictionary and thesaurus.

The subcategorization dictionary contains 25,000 patterns about 19,000 predicates extracted from the corpora. This dictionary defines 41 standard verb patterns and 17 standard adjective patterns. It has been built semi-automatically based on the four semantic cases. This dictionary is appropriate for this

experiment, since it is for general use. It does not have domain-dependant information, unlike other small subcategorization dictionaries used in other experiments. The structure of the subcategorization dictionary is shown in Table 1.

Table 1. An Example from the Subcategorization Dictionary

Head Predicate	Information of Essential Complement Elements
가다 (Go)	[Pattern] 1 가(is)[AGT] + 2 볼(to)[RNG] [Concept] 1. 인간(Human) 2. 장소(Place)
	[Pattern] 1 가(is)[AGT] + 2 에서(from)[SRC] + 3 로(to)[PTH] + 4 에 게(to)[GOL] [Concept] 1. 인간(Human), 인간이외의동물(Animal besides human), 탈것(Things to ride) 4. Place, Human, Direction, Building

The thesaurus is a modified form of the noun hierarchical dictionary to correspond to the subcategorization dictionary. For better understanding of the IS-A relation between nouns, the sense code is given in the thesaurus. The structure of the thesaurus with the sense code is shown in the Figure 1.

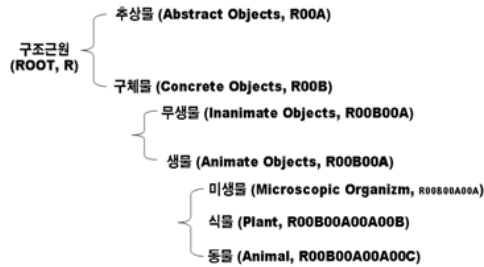


Figure 1. The Example of the Thesaurus Structure

By comparing the information on meanings in the subcategorization dictionary and the sense code of nouns in the thesaurus, the ambiguity is reduced when the structure of the sentence is analyzed depending on the complement of the sentence. The match between sense and the sense code is based on the IS-A relation and the meaning of the noun and predicate in a sentence could be well defined. This Selectional Restriction method [4] is used in TLCSS in this study.

3. The procedure of the Two-Level Clausal Segmentation

3.1 Algorithm of the TLCSS

This study suggests an algorithm (Figure 2) that segments a sentence into simple sentences through two steps in corpora that has morphological analysis.

Morphological analysis is the first and most important step in semantic analysis as well as in Clausal Segmentation [4]. Errors or ambiguity in morphological analysis have great impact on the accuracy of this experiment. In order to focus on the algorithm, the explanation of the morphological analysis is omitted. This study uses a corpora that manually analyzes the morpheme [3].

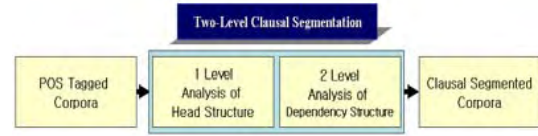


Figure 2. The Diagram of TLCSS

TLCSS is comprised of two procedures. The first is head element selection that ex-tracts the complement belonging to the predicates. The latter extends the dependency structure that includes the non-complement element. This procedure minimizes search space. It reduces the error and ambiguity of Clausal Segmentation since the Korean language does not have a cross-dependency. The predicate used in this Clausal Segmentation refers to verb, adjective, and affix that makes word predicate. The complement is defined as the essential part that the predicate has in a sentence. A substantive element excepting the complement is the non-head element.

3.2 Selection Algorithm of Head Structure

This paper suggests the method that extracts a predicate in a sentence then finds the complements that come with the predicate. The semantic information in a subcategorization dictionary and thesaurus are utilized to select the most appropriate candidate. When finding complements, the right element of the governing element as well as the left element to a certain extent are checked to include the adnominal phrase for Clausal Segmentation. The algorithm for head structure selection is shown in Figure 3. CountSentence refers to the length of a sentence, and CountHead is a variable that indicates the numbers of the head element in a sentence.

```

int CountSentence;
void SelectProcess(0, CountSentence){
    int CountHead = 0;
    int CountMatchSubcat = 0;
    int HeadVerbals[MaxWordsInSentence];
    SubcatItem SubcatforHeadVerbal[MaxSubItem];

    for(int i=0, j=0; i<CountSentence; i++){
        if(isHeadVerbal(i))
            HeadVerbal[CountHead++] = i;
        if(isDoubleQuote(i)){
            j = FindNextDoubleQuote(i);
        }
    }
}

```

```

SelectProcess(i, j);
i = j + 1;
}
}
for(int i=0; i<CountHead; i++){
SubcatforHeadVerbal[] = SearchSubcatDic(i);
for(int j=0; j<Length(SubcatforHeadVerbal[]); j++){
MatchedSubcatforHeadVerbal[CountMatchedSubcat++] =
MatchSubcatAndThesaurus(SubcatforHeadVerbal[j], i);
}
SelectCandidate(MatchedSubcatforHeadVerbal[],
CountMatchedSubcat);
}
}

```

Figure 3. The Algorithm of the Head Structure Selection

First, the governing element is extracted in a sentence. isHeadVerb() is the procedure that decides whether or not the element is a governing one. The governing element is a predicate that includes, and adjectives and any suffix that could make a word predicate such as ‘되/SF’, ‘하/SF’, ‘스럽/SF’, ‘답/SF’. Auxiliary predicates could be excluded in a way that the word in a left side only can be considered as a head element. When there is isHeadVerbal(), sentence element can be put under the HeadVerbals[]. HeadVerbals[] is an array to represent the governing element in a sentence. isDoubleQuote() detects a quotation mark in the process of finding a head-element in a sentence. When a quotation mark exists, the extent of the quotation mark is identified using the process of FindNextDoubleQuote(). SelectProcess() delegates the Clausal Segmentation in the range of the double quotation regardless of a sentence.

After detecting a governing element, its complements are extracted. SearchSubcatDict() finds a pattern of each governing element from the subcategorization dictionary and allots it in an array of the SubcatforHeadVerbals[]. MatchSubcatandThesaurus() compares the particle of the pattern in the subcategorization dictionary and the particle of a sentence. Semantic elements of the thesaurus of a noun in the sentence are checked vertically with the sense information in the subcategorization dictionary then when these two are identified, it is selected as a complement element. In this process, the candidate to be checked is limited as a substantive (noun, pronoun, numeral, dependent noun) that is located on the left side of the governing element. This process is repeated as many times as the number of CountHead of the governing element found in SearchSubcatDict(). When a complement follows a predicate, as in the adnominal clause, it is processed at the stage of expansion of the dependent structure.

When the information of the meaning in the subcategorization dictionary and thesaurus are matched, the method of using the conceptual distance can be considered instead of a vertical check of meaning. The

nodes of the thesaurus are encoded hierarchically, as in Figure 2, to measure the distance and hierarchy easily using the thesaurus network. The accumulation of lexicon and code in the database enables us to determine the IS-A relation and depth, sibling relation, and distance from a certain node of two lexicons by simple string matching. However the experiment of using sibling nodes in sense matching was ineffective. It is necessary to define the conceptual distance depending on the method of utilization. However, in this paper it is considered valuable that there is a IS-A relation between the meaning of the hypernym in the subcategorization dictionary and the meaning of the subordinate word in thesaurus.

SelectCandidate() is the process of selecting an optimal candidate that has the most identified elements among the extracted ones. When two or more candidates have the same numbers of elements, then the weight of the subject is lowered and it takes one candidate. This process includes the selection of the complement element of the governor. The remaining element is regarded as a non-head feature. When there is a quotation mark in a sentence, the governing element finds the complement within the quotation mark and applies the head structure selection algorithm.

3.3 The Algorithm of Expansion of the Dependent Structure

The process of including a non-head feature element in addition to the complement of the predicate in the head-feature selection algorithm and extracting a simple sentence can be incorporated in this algorithm. Figure 4 shows the algorithm of the expansion of the dependent structure. This algorithm expands the element of the complement by selecting a complement element considering the non-head feature sentence. It processes the non-head feature, except the complement element of the governing element, by searching from the right side of the sentence.

First, isComplement() determines whether or not the element of the sentence is a complement. If it is a complement, it is taken as the Current Complement. When the governing element has an adnominal clause, its pattern and meaning are compared with that of the subcategorization dictionary at the stage of ornnProcesstoRight() and its noun element is included in the complement element. isOrn() is the Boolean function that checks the case where the governing element of the sentence is a simple adnominal clause like ‘governing element + [~는/EM, ~ㄴ/EM, ~ㄹ/EM, ~되/SF]’.

isConnector() decides ConnectionFlag depending on the noun-phrase that has a connection word such as ‘,’, ‘그리고(And)’ or ‘과/와/이/나(And, Or)’. When ConjunctionFlag is TRUE, SearchConjunctionNounPhrase() processes the connection word.

```

int CountSentence;
void ExpandProcess(0, CountSentence){
    int CurrentComplement;
    int ConjunctionFlag;
    for(int i=CountSentence; i>0; i--){
        if(isComplement(i))
            CurrentComplement = i;
        else if(isOrn(i)){
            for(int j=i+1; j<CountSentence; j++){
                OrnProcessToRight(i, j);
            }
            if(ConjunctionFlag)
                SearchConjunctionNounPhrase(i, CountSentence);
            else if(SearchComplexNounPhrase(i, CountSentence))
                SearchVerbalForNoun(i, CountSentence);
            if(isConnector(i)) ConjunctionFlag = TRUE;
        }
    }
    for(int i=0; i<CountSentence; i++){
        if(isAux(i)) SearchVerbalForAux(0, i);
        else if(isAdverb(i)) SearchVerbalForAdverb(i, CountSentence);
        else if(isAdject(i)) SearchNounForAdject(i, CountSentence);
        else if(isInjection(i)) SearchVerbalForInjection(i, CountSentence);
        else if(isSymbol(i)) SymbolProcess();
        else ExceptProcess();
    }
    PostProcessing();
}

```

Figure 4. The Algorithm of Expansion of the Dependent Structure

For example, if a substantive is connected with ‘,’ , the latter substantive is defined as the same element with the former one. SearchComplexNounPhrase() examines the noun that is not taken as a complement. If the following noun is a complement, the noun is checked for meaning in the complex noun dictionary, and if the following noun is a compound noun, the SearchVerbalForNoun() entitles it to a complement element as the compound noun of CurrentComplement. The SearchVerbForAdject, SearchVerbForAdverb(), SearchNounforAdject(), SearchVerbalforInjection(), and SymbolProcess() are the processes that include adverb, adnominal, interjection, and symbol in the nearest complement element. ExceptProcess()deals with exceptional cases. PostProcessing() manages the following two cases. First, when the governing element could not include the complement element that is a subject, the subject that is located in front of a sentence is included; and when the predicate is not found in the subcategorization dictionary, the nearest subject is taken as a complement.

4. Experiment and Evaluation

TLCSS is applied to a corpora to prove its validity. This experiment uses 25,000 sentences in the group of corpora ETRI-KONAN[3]. The segmented sentences are compared with the corpora, which tag the sense of

noun and the relation with predicate and noun manually, in order to evaluate TLCSS.

4.1 Extracting a Governing Element

This study defines the governing element of a simple sentence as a predicate or an element that plays a role as a predicate with affix that makes a word a predicate. The corpora used in this study extracted predicates and words with a predicative affix among 25,000 sentences.

Table 2. The Governing Element Used in This Experiment

Section	Numbers
Verb, Adjective	70762
Predicate with affix	41764
Total governors	112526

The corpora used in this study is from the group of ETRI-KONAN. As shown in Table 2 the average number of verbs and adjectives per sentence is 2.83, and that of predicates, including a word with a predicative affix, is 1.67 per sentence. That is, this experiment uses a sentence that has on average 4.5 governing elements, and 8.02 word-phrases.

4.2 The Process of Head Structure Selection

The process of head structure selection extracts a complement element that can be included by a predicate in a sentence. The subcategorization dictionary used in this study comprises 25,000 patterns out of 19,000 predicates. This dictionary also possesses information of the thesaurus that encoded 120 thousand hierarchical meanings. This information clarifies the hierarchy of a noun and this also makes it possible to know the meaning of a complement element.

In order to find a complement element from the extracted predicate, this study follows the algorithm of head structure selection suggested here. The process and result of Clausal Segmentation is described by applying the whole process of head element, that is a complement selection to a example sentence in Table 3.

Table 3. The Example of Raw Sentence

어제 밤에 집에서 바다로 고기를 잡으러 나가셨던 부모님께서 파도에 휩쓸려 돌아가셨습니다. My parents, who went to the sea from my home to catch fish, were drowned and killed last night.

In the example in Table 3, the predicate ‘나가다(Go)’ has a candidate for each pattern from the subcategorization dictionary. This is shown in

Table 4. When the list of candidates is produced, the information of sense in the subcategorization dictionary and the meaning of the substantive of a sentence are found in the thesaurus and compared. If the meanings match, the word is selected as a complement element. Through these procedures, the candidate with the most numbers of complement elements is taken as the most appropriate candidate. If there are the same numbers of complement elements, the particles are examined in the same way to single out a candidate. The case of ‘ 나가다(Go)’, select a ‘ 집에서 바다로 나가다(Go to the sea from my home)’, number 700 in the subcategorization dictionary, as the most appropriate candidate. Other predicates such as ‘ 잡다(Capture)’, ‘ 휩쓸리다(Be drowned)’, and ‘ 돌아가다(Kill)’ also select a suitable candidate through the same procedure with the ‘ 나가다(Go)’. The result of the example of head element selection is shown in Table 5.

Table 4. Producing Candidates

Subcat Pattern Number & Candidates	
나가다 (Go)	
685	바다로 나가다. (go to the sea)
686	집에서 바다로 나가다. (go to the sea from my home)
700	집에서 나가다. (go from my home), 잡다 (Capture)
5196	고기를 잡다. (capture the fishes)
5197	부모님께서 고기를 잡다. (My Parents capture the fish)
	휩쓸리다 (be drowned), 돌아가다 (Kill)

The number of governing elements used in this experiment is 112,526. Of these 486 (0.43%) are not registered in the subcategorization dictionary. Among them, 33 elements are causative and passive. ‘ 휩쓸리다(Be drowned)’ in the example is the predicate that is not registered in the subcategorization dictionary. When there are causative and passive elements, it is necessary to change the subcategory, or the transformed rule can handle these cases. However, this experiment excludes these cases.

Table 5. The Result of Head Elements Selection

Head Elements Selection
집에서 바다로 나가다. (go to the sea from my home)
고기를 잡다. (capture the fishes)
휩쓸리다 (be drowned)
부모님께서 돌아가다. (My parents are killed.)

In order to evaluate the accuracy of the complement element of the most appropriate candidates, the complements are compared with the 25,000 sentences that are tagged semantically in a semi-automatic way. The result of comparing the complement element that is tagged semantically and the most appropriate candidate, which is the result of head structure selection, is shown in the Table 6.

Table 6 only displays the governing elements that exclude the unregistered governing element. Complete matching means that the sense element of the governing element is wholly matched with the element of a sentence. Partial matching occurs when some senses are not matched with the element of a sentence. This is inevitably caused by the lack of information in the subcategorization dictionary and thesaurus. Moreover, it is a strength to use sense information to process the meaning of predicate and noun conducting Clausal Segmentation.

Table 6. The Accuracy of Head Elements Selection Compared Sense Tagged Sentence

Complete Matching	80 %
Partial Matching	20 %

4.3 The Process of Expansion of the Dependent Structure

The algorithm to expand the dependent structure is used in this experiment to process the non-complement element in head structure selection, and to process exceptions for adnominal clauses. First, the compound noun is not treated as a complement element that is located on the left side of the complement element, but is treated as a noun that is combined with another noun. Its meaning is checked in the dictionary of compound nouns and when it is registered in the dictionary, the noun, which is not an element of complement, is included as a complement element.

After dealing with the compound noun, adnominal clauses that frequently appear, and are one of the unique features of Korean sentences, are processed. The corpora used in this study has 12,154 adnominal clauses, 10.8% of all governing elements. In order to select an optimal candidate, when the candidate does not have a subject element and the two adjacent sentences include adnominal clause, the right word of the governing element in adnominal clause is semantically matched with the subject element of the subcategorization dictionary. If it is matched it is treated as a complement element. After processing adnominal clauses, 84% (10,209) of adnominal sentences are segmented correctly. The result of processing the remaining non-head element features by the algorithm expanding dependent structures is shown in Table 7. Following Clausal Segmentation, the most significant error occurs in detecting a subject element amongst complement elements, when it is located far from a predicate. Post-processing is conducted to reduce this error.

This process deals with the following cases; when the predicate can not identify the complement element

that is also a subject and there is a subject in front of a sentence, it is treated as a complement element; when the predicate is not listed in the subcategorization dictionary the closest subject is selected as a complement.

Table 7. The Result of Expansion of the Dependent Structure

Expansion of the Dependent Structure
부모님께서 어제 밤에 집에서 바다로 나가다. (My parents went to the sea from my home last night)
부모님께서 어제 밤에 고기를 잡다. (My Parents captured the fishes last night)
부모님께서 파도에 휩쓸리다 (My Parent were drowned)
부모님께서 돌아가다. (My parents were killed.)

In the example sentence, ‘ 부모님께서(My Parents)’ , which is the subject of ‘ 휩쓸리다(Be drowned)’ , is counted as a complement element. After post-processing the result is improved by 2.6%. The accuracy of TLCSS is shown in Table 8.

Table 8. TLCC Accuracy

Total Numbers of Predicates : 12526		
Before	100,373	Accuracy Rate : 89.2%
After	103,389	Accuracy Rate : 91.8%

This study could achieve the correct result of Clausal Segmentation for any corpora that include sentences omitting a particle, not only from the special corpora. Furthermore as a result of Clausal Segmentation, simple sentences are elicited and the dependency relations of major element of these simple sentences are estimated. At the stage of head structure selection, the meaning of governing and complement element could be comprehended by matching subcategorization dictionary and thesaurus.

The main cause of any failure in this experiment in head structure selection is that there is no predicate in the subcategorization dictionary (0.43%). The second reason is where there are cases where the subcategorization dictionary and thesaurus are not correctly matched, so some words can not be treated as complement elements. Third, when the complement is located far from predicate, the element of the complement can not be selected. The remaining situations can be attributed to errors in speech tagging and lack of predicate patterns in the subcategorization dictionary. These problem should be addressed in the process of dependent structure expansion and post-processing in order to enhance the success rate of Clausal Segmentation.

5. Conclusion

This study designed and implemented a TLCC system that divides a Korean sentence into simple sentences. TLCC utilizes clausal information in the process of Clausal Segmentation and the method of limiting the selection with semantic information also makes it possible to reduce the ambiguity of Clausal Segmentation. This study proposes the TLCC algorithm that has two steps; head structure selection and de-pendent structure expansion. The accuracy of semantic and clausal information in the subcategorization dictionary influences Clausal Segmentation accuracy. Therefore the accuracy of sense matching with the thesaurus and the subcategorization dictionary should be improved by modifying the dictionary. Clausal information and empirical rules should be applied to enhance the accuracy of Clausal Segmentation.

References

- [1] Y. Wilks, M. Stevenson, "Sense Tagging: Semantic Tagging with a Lexicon", Proc. of the SIGLEX Workshop, pp.74-78, 1997.
- [2] R. Grishman, C. Macleod, A. Meyers, "Complex Syntax: Build a Computational Lexicon", Proc. of COLING Workshop, pp.268-272, 1994.
- [3] K. Choo, Y. Woo, "Automatic Extension of Korean Predicate-Based Subcategorization Dictionary from Sense Tagged Corpora", Lecture Note in Computer Science 3045, pp.585-592, 2004.
- [4] Y. Wilks, M. Stevenson, "The Grammar of Sense: Using Part-of-Speech Tags as a First Step in Semantic Disambiguation", Natural Language Engineering, Volume 4, pp.135-143, 1998.

Measuring Lexical Cohesion in a Document

Kamakhya Gupta, Mohamed Sadiq, Sridhar V

Abstract— Lexical Cohesion is one of the important features in text processing for analyzing document structure and improving the accuracy of text processing. We present a hierarchical graph based model to measure cohesion by grouping lexically cohesive units together in a text. Latent semantic analysis is used to construct a relational graph to uncover the hidden semantics from the text by discovering new relations and their weights. Coherence score is derived from the learned weights that are assigned to edges in the graph and number of disjoint partitions in the graph.

I. INTRODUCTION

Lexical Cohesion is an important feature for many text mining applications, for example, grading essays, text segmentation and summarization. Lexical cohesion refers to the chains of related words that contribute to the continuity of lexical meaning. Coherence, including both sentence and paragraph coherence, lies in the deep structure of a text, and refers to the complex non-linear functional relations behind the linear sequences of words, sentences or paragraphs.

Automatic evaluation of essay grading is an active area of research. Lexical cohesion is one of the important features to grade and give feedback for an essay. Latent semantic analysis [2] has been quite a popular technique and has been used in various Intelligent tutoring systems like AutoTutor [7], Intelligent Essay Assessor [6], Summary Street [8], Apex [9]. APEX [9] is a popular system for essay grading that uses LSA[2] to measure content and cohesion between the sentences of the text. It detects a break if the similarity between the adjacent sentences using the LSA[2] space falls below the chosen threshold. Such measures which analyze the text locally are too simple and primitive and thus fail to take the global perspective. On the other hand we give overall lexical cohesion score to a given text.

We present a method to measure the lexical cohesion of text. When the input sentences in a document flow smoothly, growing out of another, the text is said to be coherent. The key observation of our methodology assumes a writing which is supposed to be coherent. This means that the sentences should be so closely connected to one another that they flow smoothly without gaps between them or jumps in logic.

The rest of the paper is organized as follows, first we discuss background on Lexical Cohesion and LSA, and then we show, insights of our data to be used for our experiment, followed by our methodology with formal representation, thereafter, experimental set up needed to implement our methodology and finally conclusion of our work.

Applied Research Group, Satyam Computer Service Limited, Bangalore

II. LEXICAL COHESION AND LSA

Lexical cohesion is due to the continuity or relatedness of lexical meaning denoted by the reiterating or collocating items, respectively. There are several factors restricting the total amount of this force[11][10].

1. Paradigmatic proximity: “There are degrees of proximity in the lexical system, a function of the relative probability with which one word tends to co-occur with another.[10]”. The more similar the paradigmatic usage regularities of words, the higher the cohesive force of their co-occurring instances.

2. Syntagmatic proximity: Paradigmatic proximity interacts with what can be called syntagmatic proximity “there is relatedness of relative proximity in the simple sense of the distance separating one item from another, the number of words or clauses or sentences in between.” [10]. In terms of a proportionality statement: The less distant concurring items, the higher their positive/negative impact on cohesive force, and vice versa.

3. Syntagmatic order: There is a third determinant of cohesive force due to the linear order of texts in the sense that “every text is also a context for itself” setting up internal expectations [11] which decrease between more distant units. In terms of a statement of proportionality. The more distant two spans of a text, the less the impact of the preceding on the subsequent span, the higher the probability that the latter manifests a change in lexical meaning.

Latent semantic analysis (LSA) is an algorithm applied to approximate the meaning of texts, thereby exposing semantic structure to computation. LSA combines the classical vector-space model, well known in computational linguistics, with singular value decomposition (SVD), a two-mode factor analysis. Thus, bag-of-words representation of text can be mapped into a modified vector space that is assumed to reflect semantic structure. LSA represents each word of a document in a high dimensional space such that proximity between the two vectors is closely related to the semantic similarity between the two corresponding words. Thus it is possible to analyze the textual content on the semantic basis.

III. DATA SOURCE

For our experiments we have chosen essay text as they have a good distribution in the cohesiveness due to samples collected from various writers on the same topic. We collected sample essays from the site <http://www.testmagic.com/test>, which contains the sample essays written in English by the students on the different topics for the TOEFL[5] exams. Each essay is graded into three scales 5, 5.5, 6 out of 6 along with some comments for a few of the essays. A part of a sample essay on a topic is shown in Table I. There are 142 different topics and for

each topic there are multiple essays which are written by different students. A total of 484 samples distributed non-uniformly amongst the different topics. Essays are primarily based on topics of interest for students such as topics related to technology and general awareness. The whole point is to justify that many topics share common themes.

Sample Text	Comments
<i>I support the view that teachers should be a part of the learning process. Below are several beneficial reasons for which I feel that teachers are needed in while learning. Below are also some of the problems that come up without their presence.</i>	<i>Grade 5.5, Very good organization and has good development. To raise her score, the author of this essay should add some clear, specific examples and make the conclusion stronger.</i>

TABLE I
SAMPLE ESSAY DATA

IV. METHODOLOGY

In this section we show a method to measure cohesion in a document. Two key contributions of our approach are:

- Discovering hidden relationships and their strength using LSA
- Building hierarchically to discover more relationships.

The method uses LSA to construct a relational graph to uncover the hidden semantics form the text. A coherence measure is derived from the learned weights assigned to edges in the graph. First text is transformed into a relational graph, where we first segment the documents into sentences and then construct a graph where each node represents a sentence and edges indicates relationship between two sentences. The vertices are ordered as per the order in the sentence they occur. The edges are created if the LSA measure is greater than chosen threshold τ . We utilize the linear order property to group the sentences hierarchically by finding new discovered edges and assign weights to the discovered edges. In doing so, we are influenced by skorochod'ko [12], shown in figure 1, which shows that semantic structure of a text by analyzing it in terms of the topology formed by the semantic interrelations found among its

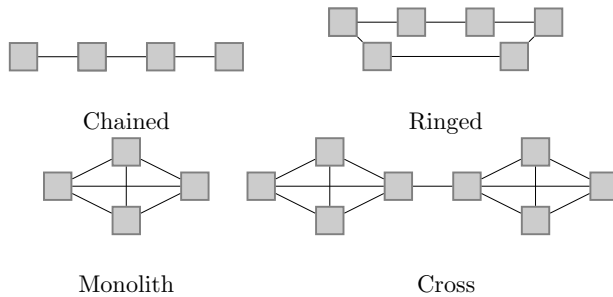


Fig. 1. Skorochod'ko text structure types. Nodes corresponds to sentences and edges between nodes indicate strong overlap between sentences

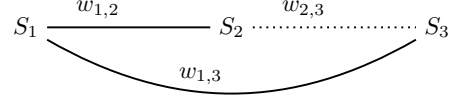


Fig. 2. Relational Graph, where solid lines existing relationship and dashed lines, the derived relationship

sentences. We have utilized the first two topology (see Figure 1), namely chained and ringed to identify partition and other two namely , monolith and cross, to measure the lexical cohesion.

We transform our problem into a relational graph, where we first segment the documents into sentences and then construct a graph where each node represents a sentence and edges indicates relationship between two sentences.

Definition 1. In a Relational order graph $G = \langle V, E \rangle$, $V = \{S_1, S_2, \dots, S_T\}$, where vertex $S_i \in V$ corresponds to sentence and if $i < j$ then S_i precedes S_j in the order it appeared in the text. The edge E between two vertices, $S_i, S_j \in V$ is present if and only if LSA similarity measure between $S_i, S_j > \tau$

From the given relationship we try to discover some new edges along with it weights. The discovery of new edges has two advantages, first to get insights of the hidden relationships and second it propagates in increasing the size of the partition. For example with three sentences, $\{S_1, S_2, S_3\}$, if S_1 is related to S_2 and S_2 is related to S_3 , then we can infer the relationship between the S_1 and S_3 (transitive relationship). See figure 2. We can now group $\{S_1, S_2, S_3\}$ as single coherent unit as between every successive vertex there is a direct path. Moreover, the symmetric and reflexivity relationship follow trivially. This make the relation an equivalence relation on the set and partition the set into disjoint sets. The elements of the disjoint sets which are the sentences, may not be in the order as they appear within the text. With the domain knowledge that neighboring sentences are more cohesive, we partition them in order as they appear in text, rather than grouping arbitrarily.

To make an order partition, we link only the adjacent vertices, if there exists a path only through their partitions and weight of the edges are assigned based on the path length. This is described in detail in the next section. The key observation is that if there is no edge in between two adjacent vertices V_a, V_{a+1} they belong to different adjacent partition.

Definition 2. In a Graph $G = \langle V, E \rangle$, an ordered partition is a sequence of $p + 1$ vertices s_k and p edges e_k of the form $S_a e_1 S_{a+1} e_2 S_{a+2} \dots S_{a+p-1} e_p S_{a+p}$ where $e_p \in E$ connects $v_{a+p-1}, v_{a+p+1} \in V$

New links are discovered using the following definition to increase the size of the partition.

Definition 3. In a Graph $G = \langle V, E \rangle$ where $P = \{P_1, P_2, P_3 \dots P_N\}$ is set of partitions on V , the relationship between adjacent vertices $S_p S_{p+1} \in V$ is only defined

if $S_p \in P_a$ and $S_{p+1} \in P_{a+1}$ and there exist a path from S_p to S_{p+1} going only through the vertices of partitions P_a and P_{a+1}

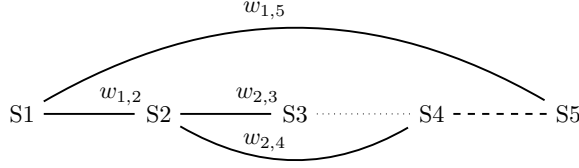


Fig. 3. Building Links



Fig. 4. Merging adjacent partition

If there is a discovery of new edge between two adjacent partitions, then the partition gets merged, as shown in Figure 4.

For example in Figure 3 initially, there are three partitions $\{\{P_1, P_2, P_3\} \{P_4\} \{P_5\}\}$ and after finding the new relationship between S_3 and S_4 shown by a dotted line makes two partitions $\{\{P_1, P_2, P_3, P_4\} \{P_5\}\}$ and then we discover the relationship between S_4 and S_5 shown by a dashed line making a single partition $\{\{P_1, P_2, P_3, P_4, P_5\}\}$

A. Updating Weights

On discovering a new relationship, appropriate weight should be assigned to uncover the hidden strength. For example between sentences, s_1, s_2, s_3 the weights $w_{1,2} = .6$ and $w_{2,3} = .8$. The basic possible ways to find the strength can be using maximum, average or minimum of weights around the path.

$$\top_{max}(w_{1,2}, w_{2,3}) = \max(w_{1,2}, w_{2,3}) = .8 \quad (1)$$

$$\top_{avg}(w_{1,2}, w_{2,3}) = \left(\frac{w_{1,2} + w_{2,3}}{2}\right) = .7 \quad (2)$$

$$\top_{min}(w_{1,2}, w_{2,3}) = \min(w_{1,2}, w_{2,3}) = .6 \quad (3)$$

If there is a path of length N , from node i to j , and weights of edges on the path are $w_{i,1}, w_{1,2}, \dots, w_{N,j}$ then weight assigned to $P_{i,j}$ is therefore the weakest link of the path.

$$P_{i,j} = \min(w_{i,j}) \quad (4)$$

In general, fixing nodes i, j , transitive affinities on different paths connecting i, j are different. For this reason, we define maximal transitive affinity between i, j as

$$h_{ij} = \max_{P_{i,j}} \top(P_{i,j}) \quad (5)$$

where $P_{i,j}$ is any possible path between i, j .

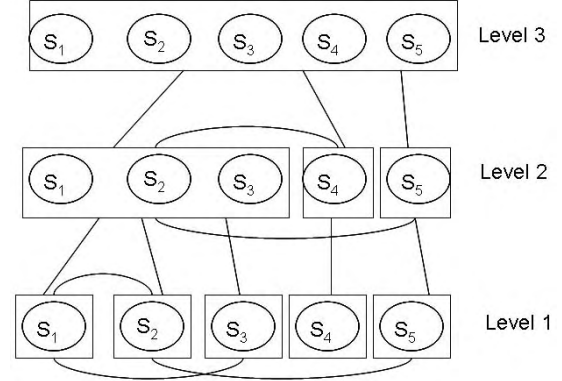


Fig. 5. Hierarchical building

B. Hierarchical building

Relational graph is constructed hierarchically as shown in the Figure 5. The motivation is to increase the size of each partition representing a lexical cohesive units together. Initially each sentence belongs to one partition and number of vertices in the graph equal to number of sentences. With some grouping made at each stage, we consider each partition as one vertex for the next pass. The weights of the graph are assigned based on the LSA measure between the sentences of the different partitions. Such an approach would tend to increase the size of the partitions on the next iteration. We stop whenever no new partitions are created or only a single partition remains.

C. Cohesive Measure

We present a method to give a cohesive score. Simple measures from the graph can be ratio of disjoint partition to number of sentences present. However this is not a good measure because due to noisy sentences, number of the partitions can increase indicating less coherence, but sentences within the partitions might be highly coherent. Similarly, on the contrary, where the number of partitions are small but within the partition sentences are not so much coherent. We combine both the principle, one which is based on the number of the partitions and another based on the number of learned weights of the graph and their corresponding weights. This seems to be good a measure as it is utilizing the skorochod'ko[12] model shown in figure 1, the monolith and cross property to measure cohesiveness in the text.

$$Cohesion = \frac{\sum_{i \in \text{links generated}} C_1 \times G_i + C_2 \times D}{\text{Number of words in the text}} \quad (6)$$

where $C_2 \gg C_1$, $C_1 + C_2 = 1$ and D is number of disjoint partitions.

V. SYSTEM ARCHITECTURE

The basic system architecture for our work is shown in figure 6. For the set of input documents are represented

Algorithm :Measuring Lexical Cohesion

Input : A set of input sentences in a document $S_1, S_2...S_N$ and threshold τ .

Output : A lexical cohesive score for different level of hierarchies

Step 1 : Initially $P_{new} = \{p_1 = S_1, p_2 = S_2...p_n = S_N\}$ and $P_{old} = \{\phi\}$

Step 2 : While (P_{old} not equals to P_{new}) {

Step 3 : Build relational graph, where vertices are elements of P_{old} and edges are created if the LSA measure is $> \tau$
step 4 : Discover the hidden relation strength along with its weight

Step 5 : $P_{old} = P_{new}$ and $P_{new} =$ Newly discovered set of partitions

step 6: Output “The cohesive score using the existing graph” }

Program 1: Measuring Lexical Cohesion in a document

in form of the sentences collection. Preprocessing modules takes all the input sentences and produces a document matrix. Using this matrix we find the weighted matrix and thereafter LSA computation is done to get the lower rank approximation of the singular value decomposition.

- Preprocessing : At first, we do auto correct for the spelling mistakes in the text as for an essay kind of text, where there are lot of spelling mistakes which can be auto corrected as they have significant impact on the LSA measure. Thereafter, we remove all the stop words and apply a stemmer to get the root keyword. Finally, representation of the sentence collection in matrix format where the cells contain the frequency, how often a particular term appears in a specific sentence. Terms are the rows, sentences are the columns.

- Weighting : We use the logarithmised term frequency as local weight and the inverse document frequency as global weight.

$$lw\_logtf(matrix) * gw\_idf(matrix) \quad (7)$$

where $lw\_logtf()$ returns the logarithmised $n \times m$ matrix. i.e, $\log(m_{i,j} + 1)$ is applied on every cell.

$gw\_idf()$ returns the inverse document frequency in a $n \times m$ matrix. Every cell is 1 plus the logarithmus of the number

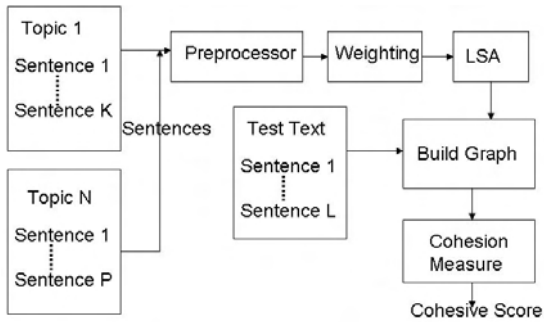


Fig. 6. System Architecture

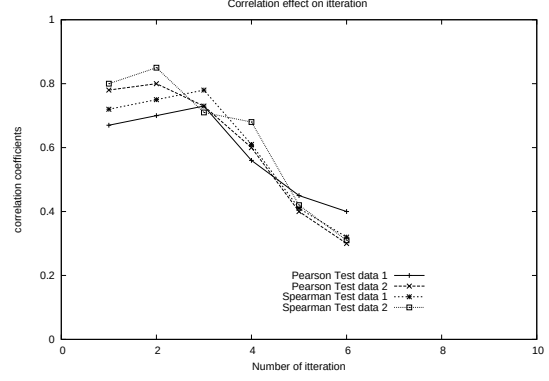


Fig. 7. Hierarchical Performance for threshold=.3

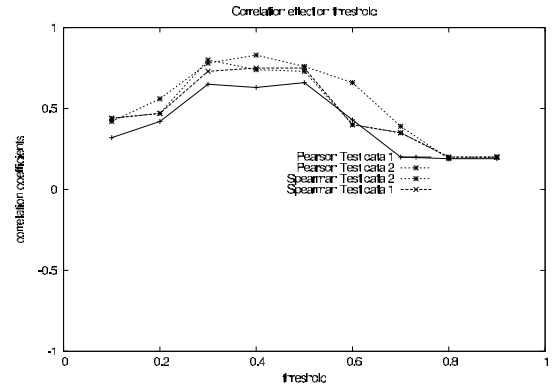


Fig. 8. Correlation measure for different values of τ at level 1

of documents divided by the number of documents where the term appears.

- LSA This matrix of the size $m \times n$ is then decomposed via a singular value decomposition, term vector matrix T (constituting left singular vectors), the document vector matrix D (constituting right singular vectors) being both orthonormal, and the diagonal matrix S (constituting singular values).

$$M = TSD^T \quad (8)$$

These matrices are then reduced to the given number of dimensions $k = dims$ to result into truncated matrices T_k , S_k and D_k , the latent semantic space.

$$M_k = \sum_{i=1}^t t_i \cdot s_i d_i^T \quad (9)$$

which is the least-squares best fit approximation of M with k singular values.

VI. EXPERIMENTAL RESULTS & DISCUSSIONS

We have implemented our methodology using *lsa* toolkit for R. Correlation coefficient measures are used to evaluate

the performance of our approach. Two popular correlation measures Pearson and Spearman are used to measure, how much predefined score (X) are related with the systems scores (Y). Pearson is a simple correlation while Spearman rank correlation is just the Pearson correlation between the ranks of two variables.

We prepared test data by using two methodologies, namely, Test data 1 and Test Data 2. Test data 1 is prepared in a semiautomated way by manually assigning score to essay based on the grade and comments written down for the essays. Test data 2, is synthesized by an algorithmic approach. We select some gold standard essays which are highly rated and jumble the sentences randomly to create a less cohesive score. Score is assigned based on the number of sentences swapped. Highest score is assigned to standard essay and score is decreased iteratively based on the percentage of swaps done in a text randomly. Initial score of 6 is assigned for a gold standard essay, and then iteratively, for each 10% swaps we decrease the score by 0.5. For example, for 20% of swaps the score assigned is 5. We do it till we reach a minimum score of 2. The idea of doing this type of data preparation is that more the number of swaps lesser will be lexical coherence.

Figure 7 shows the correlation measure on cohesive measure performance at different level of hierarchies tested over two different test data sets, Test data 1 and Test data 2. The result shows that for first few initial increase in the hierarchy level there is increase in the correlational performance and the best performance is seen for level 3 and 4 (see Figure 7). Thereafter there is a sharp decrease in the correlation measure. It may probably be because for initial few levels, the partition seems to be correctly identifying the logical semantic cohesive units together as a result the correlation is high, but after that, due to different semantic concepts getting merged resulting in decreased number of the disjoint partitions. As a result, cohesive score getting lowered then actual case leading to decrease in the correlation performance.

Figure 8 shows the cohesive measure performance for different values of threshold(τ). For the Figure 8 is apparent that for lower and higher values of τ , correlation measure is less while for thresholds in the range 0.3 and 0.4 has better correlation measures. Smaller values of τ , makes the relations graph approximately fully connected, while higher value of τ makes the graph sparse. In case of lower threshold due to more number of initial relations discovered tends to decrease the number of disjoint partition, while, sparse relations increases the number of disjoint partition. Both the lower and higher threshold, fails by overestimating and underestimating then the actual case.

From Figures 7 and 8, we observed that Spearman rank correlation gives higher correlation measure than Pearson coefficient. From the Figures 7 and 8, model performs the best for threshold τ in the range of .3 and .4 and for 3rd and 4th level of the graph in the hierarchy.

VII. CONCLUSION

In this paper we have shown using LSA and relational graph based model, a methodology to measure lexical cohesion in text. We have also shown, how hierarchically construction helps in grouping similar lexical cohesive units together, to improve the correlation results. Model best performs at certain chosen threshold and at certain hierarchical level. Hence, a fine tuning is required for the different text application domain. Future work can be focussed on exploration of the ideas to do text segmentation.

REFERENCES

- [1] P.W.Foltz, W.Kintsch and T.K.Landauer, *The measurement of textual coherence with latent semantic analysis*, *Discourse Processes*, 25:2-3, pp. 258-307, 1998
- [2] T.K.Landauer and S.T.Dumais, A solution to plato's problem: The Latent semantic analysis theory of acquisition, induction and representation of knowledge, *Psychological Review*, 104:2, ppp. 211-240, 1997
- [3] J. Morris, G. Hirst, *Lexical Cohesion by Thesaural Relations as an Indicator of the Structure of Text*, *Computational Linguistics* 17(1), March 1991.
- [4] WILD, F., *STAHl* Investigating Unstructured Texts with Latent Semantic Analysis. In: Lenz, Decker (Eds.): *Advances in Data Analysis*, Springer, 2007
<http://www.ets.org/toefl/>
- [5] P. W. Foltz, D. Laham, and T. K. Landauer. Automated essay scoring: applications to educational technology. In Proc. of the ED-MEDIA99 conference, Charlottesville. AACE., 1999
- [7] P. Wiemer-Hastings, A. C. Graesser, D. Harter, and Tutoring Research Group. *The foundation and architecture of autotutor*. In *Proc. 4th Int. Conf. on Intelligent Tutoring Systems*, Berlin. Springer-Verlag., 1998
- [8] E. Kintsch, D. Steinhart, G. Stahl, and the LSA Research Group. Developing summarization skills through the use of lsa-based feedback. *Interactive Learning Environments*, 8(2):87109, 2000
- [9] P. Dessus, B. Lemaire, and A. Vernier. *Free-text assessment in a virtual campus*. In *Proc. 3rd Int. Conf. on Human Systems Learning (CAPS2000)*, Paris, 2000
- [10] Michael A. K. Halliday and Ruqaiya Hasan. *Cohesion in English*. Longman, London, 1976
- [11] Michael A. K. Halliday and Ruqaiya Hasan. *Language, Context, and Text, Aspects of Language in a Socialsemiotic Perspective*, Oxford, 1989
- [12] Skorochodko E. Adaptive method of automatic abstracting and indexing. In C. Freiman, editor, *Information Processing 71: Proceedings of the IFIP Congress 71*, pages 11791182. North-Holland Publishing Company, 1972

Natural Language Processing Techniques for the Extraction of Semantic Information in Web Services

Maricela Bravo¹ Azucena Montes², Alejandro Reyes²

¹Morelos State Polytechnic University

mbravo@upemor.edu.mx

²National Center of Research and Technological Development

{amr, alexreyes06c}@cenidet.edu.mx

Abstract

With the increase in the number of Web services and the value they are taking, discovery of services that meet the criteria of users becomes a major challenge. Currently the discovery process lacks of well-defined semantics, therefore it is difficult to obtain the non explicit meaning of services, such as functionality. In order to provide an accurate solution to the problem described above, in this paper we evaluate two Natural Language Processing (NLP) techniques to extract semantic information from textual descriptions of Web services: linguistic patterns and extraction rules. Both techniques are implemented and compared in order to select the best alternative to our problem. This information will be useful for enhancing semantically the discovery process of services. We implemented a simulation environment for experimentation, and designed a set of experiments to show the applicability of our solution approach. Results of the tests show that the linguistic patterns are better than extraction rules.

1. Introduction

Currently Web services are published in Web-based directories, where search process is based in the occurrence of words, causing irrelevant returned services to the users. Under the service oriented platforms and architectures in use today, Web service authors describe their services using Natural Language texts, these descriptions are made in different ways and following no specific rule or format. In this context, mechanisms are necessary to identify and extract semantically relevant information from textual descriptions of Web services that expresses their meaning, responding to the question what does the service do? In this paper we present two techniques for extracting information: definitions and use of extraction rules and linguistic patterns.

The main contribution of this paper is the set of rules and patterns that identify the Web service functionality. Additionally the proposed approach presented in this paper will facilitate the automatic discovery of Web services with minimal or without human intervention.

The rest of the paper is organized as follows. In section 2 we show related research. In section 3 we present a brief overview of reported solutions. In section 4 we describe the acquisition of Web service descriptions. In section 5 we deal two information extraction (IE) techniques. In section 6 we show results which were obtained from the application of two IE techniques. Finally in section 7 we conclude and present the continuation of our work.

2. Related Research

IE has been addressed with statistical approaches to identify relevant information from texts [2, 8, 7] and linguistic approaches with machine learning to identify relevant noun phrases from documents [9, 12].

In [3] three types of linguistic patterns are presented, which are generated by machine learning algorithms and they are used to extract relevant information from documents news on terrorist attacks. In [11], authors present some semiautomatic learning tools to generate ontologies of concepts characterizing their relationships in the context of Web services. Some extraction rules are created to locate concepts in text of Web service descriptions.

In [5] authors present an approach to identify instances of events and extract semantic information related with them, as the verb that represents the event action. However in [1] and [6] authors report useful lexicon patterns to identify syntactic structures from documents and extract relevant information of them.

In contrast to the above related solution approaches, in this paper we apply linguistic analysis to extract Web service description texts, to obtain semantic

information and to identify the real meaning or functionality of the services.

3. Overview

In Figure 1 we present the overall architecture of the proposed solution. This architecture is divided in two phases, during the first phase the Web service description texts are extracted from a public service repository, the service description are coded in the standard Web Service Description Language (WSDL) that specifies a public interface to communicate with the Web service based on XML standard [4]. During the second phase, the natural language processing phase, we use tagging software to label syntactically the texts of descriptions that were acquired in order to assign them a grammatical category. Then we identify relevant semantic information that expresses the meaning of Web service action using IE techniques.

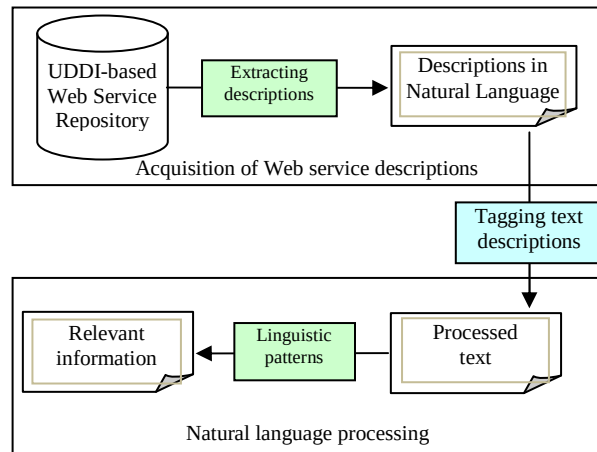


Figure 1. Architecture of the proposed solution

4. Acquisition of Web Service Descriptions

A WSDL file is used to describe in detail the operations, methods, data types, inputs, outputs and Web service descriptions. Textual descriptions are obtained from WSDL files where one of the elements in this file is the section called “documentation” that is bounded by labels `<documentation>` and `</documentation>`, which are based on XML standard. This section is related to textual descriptions made by the Web service developer and provides human-readable documentation. The descriptions in Natural Language are used as input to NLP phase.

5. Natural Language Processing

In this step we used IE techniques to identify and extract the Web service functionality from their textual descriptions tagged with FreeLing¹.

The IE task consists of obtaining relevant data in a text collection. In this paper the Web service functionality represents relevant information that is required to extract.

Some investigations as [3] and [10] claim that the “subject-verb-object” structure allows the exploitation of “verb-object” dependencies which share semantic information between main verb and object. This semantic information can be tagged as functionality due to the semantic role of verb. In this paper we regard that this relation is useful to identify functionality in textual descriptions of Web services and it can answer the question *what does the Web service do?*.

Now we present two proposed techniques to carry out the IE task. They are: a) extraction rules to locate functionalities in Web service descriptions; b) linguistic patterns that characterize the text behavior and locate functionality types that are presented in textual descriptions of services. Both methods are tested with FreeLing tagger and evaluated later. The two proposed techniques are described next.

5.1. Extraction Rules

Extraction rules are used to identify a set of words in corpus. In order to obtain the extraction rules, we make an analysis about characteristics of sublanguage that is used to describe Web services. This analysis shows that verb and objects are indicators of Web service functionality given the characteristics of such language.

Based on language analysis we obtained extraction rules to identify the most of Web service functionalities which are shown in Figure 2.

The rule shown in 2a (for one functionality) is used to extract an atomic functionality by locating first, noun phrase (NP), then identifies a simple functionality and finally finds an extended functionality detecting “VB NP FP” structures which are a verb followed by a noun phrase and prepositional phrase as structure ending. The rule 2b which is an extension of the previous rule, locates one or more extended functionalities linked by a conjunction (CC) that can only be “and”, “or”, “,”.

¹ FreeLing Home Page. <http://garraf.epsevg.upc.es/freeling/>

((DET)? (ADJ NOUN)* (NOUN)) → NP FP → Prep NP Prep → of to VB NP FP → Fun_atomic	Fun_atomic(CC Fun_atomic)* → Multi_Funct
a) For one functionality	b) For one or more functionalities
Nomenclature DET. Determinant FP. Prepositional phrase ADJ. Adjective VB. Verb NP. Noun phrase NOUN. Noun Prep. Preposition CC. Conjunction	

Figure 2. Extraction rules

5.2. Linguistic Patterns

Linguistic patterns characterize the behavior of texts for a domain; therefore they are dependent of the domain and are used to extract relevant information from corpus. Because these patterns are dependent of the domain we need to choose one to characterize it, then we choose financial domain. We carried out a deep and complete linguistic analysis of financial service texts to identify pattern types that characterize as a whole the relationship between verb and action.

The obtained patterns to extract Web service functionalities in the financial domain are presented in Table 1.

Table 1. Linguistic patterns

name: Agent-Action-Instrument noun_group (Agent) noun_group (Instrument) verb_group (Action ^ type(singular_present)) subject (Agent,Action) object (Action, instrument) → Functionality (Agent Action Instrument)	name: Action-Instrument noun_group (Instrument) verb_group (Action, (type(3rd person singular))) object (Acción, instrumento) → Functionality (Action, Instrument)
a) Pattern 1	b) Pattern 2

name: Action-Instrument-postobject noun_group (Instrument) verb_group (Action, type(singular_present)) preposition (Prep, head(to), head(of)) object (Action, instrument) postobject (Instrument, Prep) → Functionality (Agent Action Instrument postobject)	name: Funci conj Funci conjunction (conj, head (and), head (or), head (,)) Functionality (Funci) Functionality_Multiple (Funci conj Funci)
c) Pattern 3	d) Pattern 4

Patterns 1 and 2 detect structures where there is an action and an instrument, the difference between them is the presence of an agent as first part of pattern 1.

Pattern 3 detects functionalities where there is a postobject (prepositional phrase) that escorts functionality. Finally, pattern 4 detects several functionalities linked by a conjunction which can be “and”, “or”, “,”.

6. Experimental Results

We executed tests with both types of information extraction techniques using the FreeLing tagger tool. We conducted experiments in two phases: we first extracted and processed the descriptions manually, and then we used automatic tools. The aim of this experimental setup was to compare the manually extracted functionalities with the automatic extracted functionalities. We used 58 Web services descriptions from the financial domain, taken from a public repository of Web services. From these descriptions we processed and identified 69 manually because some services have multiple relevant functionalities (more than one for the same service).

Table 2 shows the resulting matrix by applying extraction rules method, and the resulting matrix for linguistic patterns. This matrix represents the amount of relevant functionalities that were extracted correctly against relevant functionalities not identified by extraction techniques presented in this paper.

Table 2. Results of experiments

Applying Extraction rules	Extracted functionalities	Not extracted functionalities
Relevant functionalities	55	14
Applying Linguistic patterns	Extracted functionalities	Not extracted functionalities
Relevant functionalities	59	10

The Receiver Operating Characteristic (ROC) curves assess the value of diagnostic tests by providing a standard measure of the ability of test. Also determines the overall discriminatory ability of different potential indicator variables by generating a common metric for comparison between them.

In figure 4 we show an analysis by ROC curves where we present the obtained results with both methods. Figure 4 shows clearly that the linguistic pattern technique is better than the extraction rules, because it has greater number of functionalities extracted correctly, 85.50% against a 14.50% of functionalities not extracted. In contrast with the extraction rules which have an effectiveness of 79.70 % of functionalities extracted correctly.

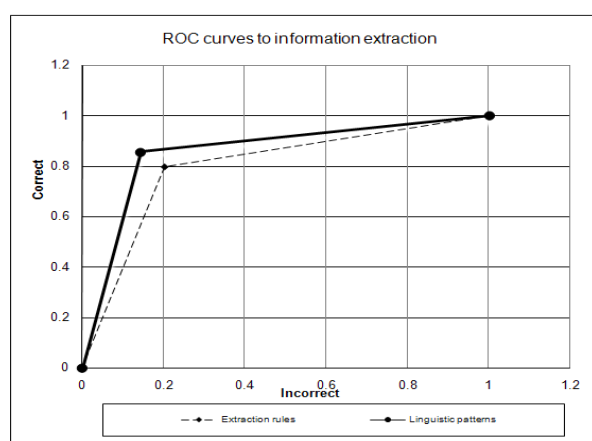


Figure 3. ROC curves

7. Conclusions

In this paper we have presented the application of two extraction techniques for obtaining semantic information from Web service descriptions: extraction rules and linguistic patterns. In particular, we have selected the financial domain for experimental purposes. Experimental results show that the use of linguistic patterns was better for discovering functionalities with 85.50% of efficiency, while extraction rules had a 79.71% of efficiency. The level of efficiency achieved by the linguistic patterns technique is due to the detailed characterization of the domain, which in turn becomes a dependency to it. The semantic information extracted will benefit the automatic discovery and composition of Web services, requiring minimal or no human intervention.

The solution approach reported in this paper represents a promising research area, because there are many Web service repositories that require semantic extraction of their characteristics in order to facilitate

Web service automatic composition. However, we need to further extend our work to incorporate a pragmatic analysis of the service logic. We are currently analyzing mechanisms to extract logic from a Web service interface (WSDL).

We are developing a hybrid approach which combines both techniques for extracting semantic information from the Web service functionality. We will evaluate if the mixture of both techniques provides better functionality discovery than each technique isolated.

We will also consider the use of NLP techniques to treat linguistic text phenomena, such as: verbs as nouns, functionalities with conjunctions, periphrasis, among others; with the aim at achieving more efficiency. As a future work we are designing an ontology to support the automatic classification of Web services, considering the semantic information extracted from their textual descriptions. This ontology will be useful for an agent-based automatic composition and discovery of Web services.

8. References

- [1] Berland M. and Charniak E., "Finding Parts in Very Large Corpora", *Proceedings of the 37th annual meeting of the Association for Computational Linguistics on Computational Linguistics*, USA, 1999.
- [2] Borkar V., Deshmukh K. and Sarawagi S., "Automatic segmentation of text into structured records", *In ACM SIGMOD Conference*, USA, pp. 175–186, 2001.
- [3] Buchholz S., "Memory-based grammatical relation finding". *In Proceedings of the Joint SIGDAT Conference EMNLP/VLC*, 2002.
- [4] Cerami E., *Web Service Essentials*, Editorial O'Reilly & Associates, pp. 119-140, USA, 2002.
- [5] Harsha V., Balakrishnan N. and Ramakrishnan K. R., "Event Information Extraction Using Link Grammar", *In Proc. of the 13th International Workshop on Research Issues in Data Engineering: Multi-lingual Information Management*, 2003.
- [6] Lehnert W., "Symbolic/Subsymbolic Sentence Analysis: Exploiting the Best of Two Worlds", *Advances in Connectionist and Neural Computation Theory*, Vol. 1. Ablex Publishers, Norwood, NJ, pp. 135–164, 1991.
- [7] Marthi B., Milch B. and Russell S., "First-order probabilistic models for information extraction", *IJCAI 2003 Workshop on Learning Statistical Models from Relational Data*, Mexico, 2003.
- [8] McCallum A., Freitag D. and Pererira F., "Maximum Entropy Markov Models for Information Extraction and

Segmentation”, *Proceedings of ICML 2000*, USA, pp. 591–598, 2000.

[9] Muresan S., Tzoukermann E. and Klavans J. L., “Combining linguistic and machine learning techniques for email summarization”. In *ConLL '01: Proceedings of the 2001 workshop on Computational Natural Language Learning*. Association for Computational Linguistics, USA, pp. 1–8, 2001.

[10] Reinberger M. and Daelemans W., “Unsupervised Text Mining for Ontology Extraction: An Evaluation of Statistical Measures”, *Scientific Literature Digital Library*, Alemania 2004.

[11] Sabou M., Wroe Ch., Goble C. and Stuckenschmidt H., “Learning Domain Ontologies for Semantic Web Service Descriptions”, *Journal of Web Semantics-Elsevier*, vol. 3, 2005.

[12] Tzoukermann E., Muresan S. and Klavans J. L., “Gist-it: summarizing email using linguistic knowledge and machine learning”, In *Proceedings of the workshop on Human Language Technology and Knowledge Management*. Association for Computational Linguistics, USA, pp. 1–8, 2001.

Classic Chinese Automatic Question Answering System Based on Pragmatics Information

Ye Yang

Graduate School of Advanced Science
Technology Education
The University of Tokushima
Minamijyosanjima 2-1, Tokushima, Japan
yang@is.tokushima-u.ac.jp

Fuji Ren

Graduate School of Sociotechno Science
The University of Tokushima
Beijing University of Post and Telecommunications
ren@is.tokushima-u.ac.jp

Peilin jiang

Graduate School of Advanced Science
Technology Education
The University of Tokushima
jiang@is.tokushima-u.ac.jp

Seiji Tsuchiya

Graduate School of Sociotechno Science
The University of Tokushima
tsuchiya@is.tokushima-u.ac.jp

Abstract

In this paper, we propose a practical question answering system for the “Analects of Confucius” based on the pragmatics information. The “Analects of Confucius” is a classic Chinese literature by the Confucian during the Warring States Period (476 B.C-221 B.C). But the literature can not be categorized by topics and it has comprehensive meanings, not only the literal meanings and explicitly communicated meanings of the passages, but also deeper meanings and implicitly communicated meanings(Here, we call the deep meaning and implicitly communicated meaning as pragmatics information). Therefore we constructed a QA system, which aims to help understanding the “Analects of Confucius” correctly. According to the experiments, the pragmatics information based retrieval results are more accurate than the results based on the interpretations in modern language.

1. Introduction

The “Analects of Confucius” is a representative of the Chinese culture and it is the best work of the Confucianism. It is occasionally quoted in the conversation of the politicians in China, Japan and Russia. For example, when President of Russia has given a lecture in Peking University, he quoted the words of Confucius “Xue2Er3Shi2Xi2Zhi1Bu2Yi4Yue4Hu1(To learn and then practise it time and again is a pleasure, isn’t it?)”.

In this way, the speaker often quotes words from the ancient literature to make the talk more accessible to the listener. However, the meaning that the “Analects of Confucius” expresses are not exclusive. For example, the sentence 1 in Table 1, its literal meaning is how to put into practice what one has heard, but it also includes some education thought of Yin1Cai2Shi1Jiao4 (The educational method should change according to the learner).

When the user retrieves the Confucius’ statement concerning with Yin1Cai2Shi1Jiao4, they cannot find out the sentence 1 in Table 1 by the previous researches[1],[2]. Because those works focus on the literal meaning of the word, the literal meanings and explicitly communicated meanings of the passages. But the literal meaning of the sentence 1 is not related to Yin1Cai2Shi1Jiao4.

In this paper, a QA system for the “Analects of Confucius” is proposed, which aims to help the user finding answer accurately and quickly in the literature.

2. About the “Analects of Confucius”

As the representative thought in China, Confucianism that started from the Confucius feeds China society, neighboring countries and areas, and many parts of the world for more than 2000 years. In the “Analects of Confucius”, around three fourths is occupied by the words (utterance) of the Confucius. It consists of 20 chapters, about 500 verses, and the theme is not decided on each chapter. The title of each chapter adopted the first two characters at the beginning of each chapter so it is not the real theme.

Table 1. The Verses of the “Analects of Confucius”

number	verse(English)
1	Zi Lu asked,“Should one put into practice what one has heard?” Confucius replied,“Your father and siblings are still alive, it is best you consider first before committing what you heard into practice.” Ran You asked,“Should one put into practice what one has heard?” Confucius replied,“Yes, you should.” Gong Xi Hua was puzzled and asked,“Teacher, both asked the same question but why the answers are different?” Confucius replied,“Ran Qiu is meek in nature, thus I encourage him to move forward; Zhong You is almost fearless, thus I restrained him.”
2	Confucius said,“The people may be made to follow, but should not be made to know.”
3	Confucius said,“One must rule with virtue and moral as the main consideration, like the North Star staying in its place and other stars revolve around it.”

Additionally, there are various opinions for the interpretation of the semantics of the “Analects of Confucius”. For example, the sentence 2 in Table 1, has two quite different ways to interpret, “Preaching the people one by one is somewhat difficult, but instructing them to be subject to the governing is possible” and “Because the people should obey the orders, it does not have to explain a principle and policy to them”.

3. Related work

Up to now several restricted domain Chinese Question Answering Systems have been constructed. For example, [3]proposes a Question Answering system which concerns with sightseeing information and combines the question answering database retrieval and the document retrieval of the web text by integrating the statistical technique and the analytical base. And [4] is about human relationships in “Dream of the Red Chamber” using the technology of a shallow inference structure mode. But these methods only

address the literal meaning and explicitly communicated meaning of the word and the passage. [5] observes the features of the pragmatics in the anaphora and constructs a QA system that makes use of the access dialog containing the features.

4. Outline of the QA system

Our goal is answer to any question about the thought of Confucius by the natural language of Chinese automatically. The complete architecture of the system is shown in Fig 1. The system consists of three modules: sentence analysis module, question analysis module and document retrieval module.

We compute similarity between the question and extracted pragmatics information of each verse with VSM model. Each element of the vector indicates the frequency of the word appeared in the document, but not tfidf[6], because there are not so much sentences of the “Analects of Confucius”.

For example, a question “Jiao4Yu4De0Zhong4Dian3 Shi4Shen2Me0(what is the key of education.)” is processed by the system. The first step is morphological analysis and the result is “Jiao4Yu4 (education)/De0 (of)/ Zhong4Dian3(the key)/Shi4 (is)/Shen2Me0(what)”. The POS of each segment is also assigned. Then we remove the “De0(of)” and “Shi4(is)” according to the unnecessary word list, and extract the keyword of this question, such as “Jiao4Yu4(education)”, “Zhong4Dian3(the key)” etc. After that, using the HowNet, the keywords are extended. Because there is a word “Jiao4Yu4(education)” in it, the question is deemed to be related to education. In the document retrieval, the system search the documents which have same type and category to the question. Then the system calculates the similarity, sorts the query results and return correlative answers.

5. Question analysis

For the characteristic of the “Analects of Confucius”, we cannot use the existing test collection. Therefore we made the test collection of the “Analects of Confucius” originally. 50 subjects were asked to make the questions of 20 verses per person. These questions were checked by the other ten subjects and then composed a test collection. The test collection includes 199 questions.

All the collected questions are clustered by k-medoids algorithm and most of them can be compartmentalized into 6 types: Method type(ask the method), Attitude type(asked the belief and views), Reason type(ask the reason), Difference type(ask the difference of two things), Content type(ask the contents and the purpose and the role, etc) and others such as alternative question.

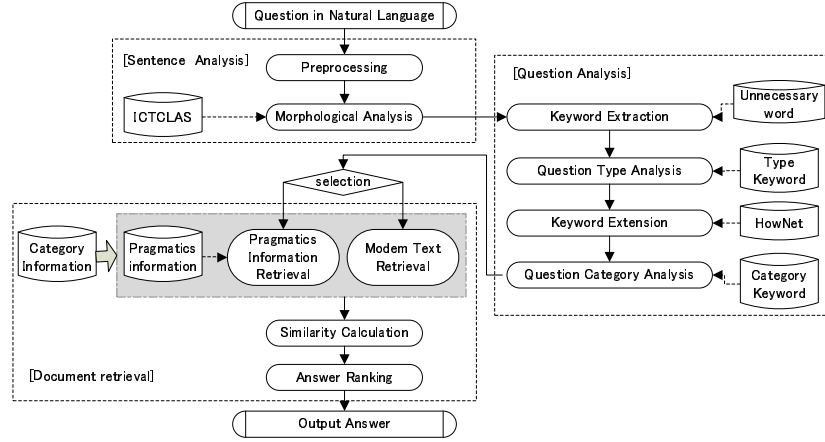


Figure 1. Architecture of the system

We make keyword collection to judge question category. According to the categories, we extract the important expression as keyword from pragmatics information and the interpretations in modern language of the “Analects of Confucius” after the morphological analysis using ICTCLAS[7]. Then, we expand the keyword of each category. During expanding, the segment that does not directly express the content of keyword is deleted and the one plays an important role of the content is signed as keyword.

6 Approach based on pragmatics information

6.1 Semantics and pragmatics

Semantics is the approach that takes no count of a speaker and a listener, the utterance’s environment and the background knowledge, etc, and merely addresses the meaning that purely expressed by language[8].

On the other hand, the pragmatics, unlike the semantics, is the approach that addresses the meaning that infers from the knowledge, process, and the situation about a speaker and a listener, the utterance’s environment, and the background knowledge, etc[8].

6.2 Pragmatics information

The pragmatics is the approach that is mainly used in a field of utterance and discourse analysis. The utterance is a sentence to be emitted from a real context but different from the sentence mentioned in the semantics.

For example, “the student of the Tokyo University is excellent” can not only be interpreted literally as “the student of the Tokyo University is excellent”, but also ironically

means “the student of the Tokyo University is not excellent”. In such utterance, the literal content is not enough, the listener need to infer the true content which obtains directly appear from various information but not only language. Therefore the pragmatics information is extracted by using the relevance theory to establish the “Analects of Confucius” QA system.

6.3 Pragmatics information extraction method

In relevance theory[8], there are three handlings: decision of explicature, decision of high-level explicature and decision of implicature for interpretation of the utterance.

Explicature is the proposition expressed by an utterance and is an explicitly communicated meaning. In Pragmatics, explicature can be derived by two principal methods: Disambiguity and Resolution of ellipsis.

For example, “Gentleman” in the 1 chapter 14 verse and 12 chapter 19 verses of the “Analects of Confucius” shown in Table 5 can be interpreted as “The cultivated person” and “the rulers” respectively. In addition, it is necessary to supplement the meaning “Xiang1Si4(be similar)” to expression “be someone like brothers” in the “Analects of Confucius” in 13 chapter 7 verse to make it more accessible to the reader.

High-level explicature means the propositional attitude of speaker in utterance. It is generated by adding speech act verb or propositional attitude verb’s objective to base-level explicature.

For example, from the content mentioned in the “Analects of Confucius” in 2 chapter 9 verse of and the expression “Hui(a person’s name) is not stupid”, the meaning “Confucius admires Hui” can be understood.

Implicature is the implicitly communicated meaning. It is generated from context and assumption by inference of Pragmatics. For example, the sentence in 1 chapter 7 verse of the “Analects of Confucius” can be interpreted by “The emphasis of education is to cultivate personality” although it is not mentioned directly.

In this paper, we define the meaning provided by these three handlings as pragmatics information. Pragmatics information in each verse of the “Analects of Confucius” have been manually extracted referring to the existing literatures[9],[10],[11],[12] related to the “Analects of Confucius” (A part of verses where pragmatics information does not exist are excluded).

In order to exclude the possibility to giving pragmatics information arbitrarily, 30 subjects were required to read the pragmatics information and only the pragmatics information agreed with by more than 10 subjects was submitted to the system.

7 Experiment and evaluation

In this section, the evaluation experiment is carried out for the proposed QA system. The number of questions of each category used in the experiment are shown in Table 2. Category “person” is a gathering of the verses about the gentleman, the scholar, and the woman, 30 questions of “Confucius” include both the question of category “Confucius in life” and category “Confucius who saw from others’ eyes”. 19 questions of “Others” include four categories: “Funeral and Sacrifice”, “Art and Literature”, “Destiny” and “Others”.

7.1 Experiment and Discussion of Adaptability of Method Using Pragmatics Information

The proposed method is compared with the method without using pragmatics information. For each category, Recall(R), Precision(P) and macro average F measure are used for evaluation. Fig.2 is the result of precision and recall under each threshold in category “Politics” obtained by using the pragmatics information. 0.18 is the optimal threshold in the result. Here, the threshold value indicates the similarity between the question and search object. The value is 0.18 denotes that every result whose similarity is greater than 0.18 will be returned. Fig.3 is the result that investigated precision and recall of each threshold in category “Politics” obtained with the interpretations in modern language. From this result, the precision rate reaches the top when threshold is 0.08.

The comparative results between proposed method and the method without using pragmatics are shown in Fig.4 and 5 (Poli: Politics; Sted: Study and education; Mora: The

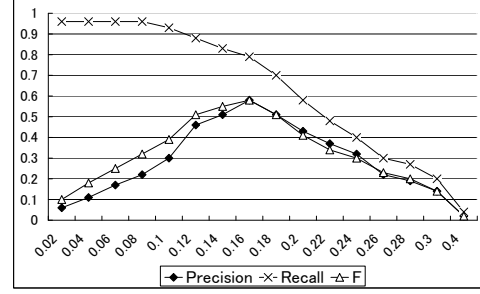


Figure 2. The threshold experimental result of the answer extraction based on pragmatics information.

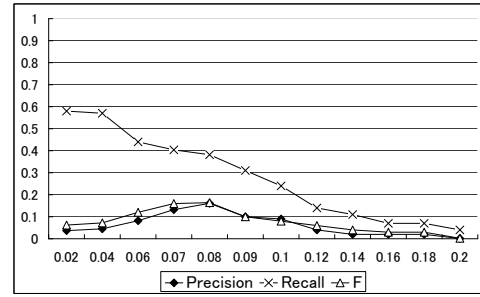


Figure 3. The threshold experimental result of the answer extraction based on the interpretations in modern language.

moral; Conf: Confucius; Peop: People; Othe: Others). P in figure presents the result by using pragmatics information, and C presents the result without using pragmatics information. The proposed method improved precision rate and recall rate in every 7 categories.

Furthermore, the results are evaluated by macro average F measure from each category and micro average F measure from all the categories[13]. The results are shown in Table 3. For the method based on pragmatics information, micro average F measure and macro average F measure are about 50%, higher than the ones without using pragmatics information. It is because there are many important words with low appearance frequency contributing to the retrieval in pragmatics information.

7.2 Experiment and discussion of rank determination of correct answer document

Next, we evaluate the system with $ANMRR$ (Average Normalized Modified Retrieval Rank) and AR (Average

Table 2. The number of questions for each category

Politics	Morals	Study and Education	Conduct	People	Confucius	Others	Sum
30	30	30	30	30	30	19	199

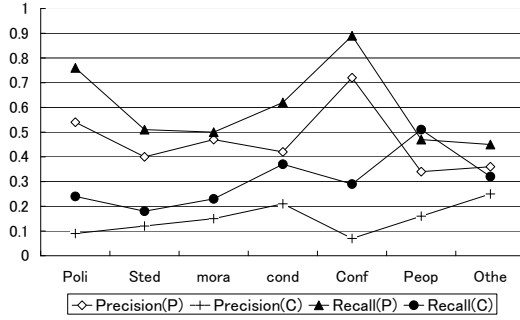


Figure 4. Precision and recall tests on seven categories.(*P* presents the results using pragmatics information, *C* presents the results using the interpretations in modern language.)

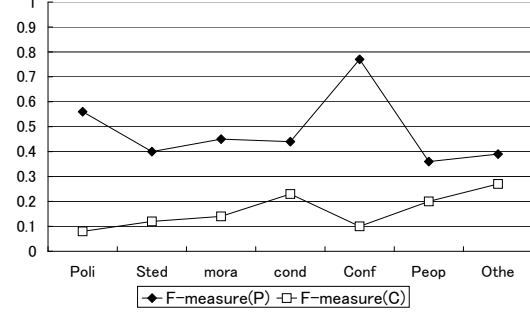


Figure 5. F-measure tests on seven categories.(*P* presents the results using pragmatics information, *C* presents the results using the interpretations in modern language.)

Recall)[14]. The ranks of all the answers are clarified by *ANMRR* (1). It is shown that the correct extracted answer ranks forward, when the *ANMRR* becomes small.

$$ANMRR = \frac{1}{Q} \sum_{q=1}^Q \frac{MRR(q)}{K - \frac{ng(q)}{2} + 0.5} \quad (1)$$

$$AR = \frac{1}{Q} \sum_{q=1}^Q R(q) \quad (2)$$

Here, *q* is question, and *Q* is the test collection. *MRR*(Modified Retrieval Rank) states the modified rank,

Table 3. Comparative results between the method without using pragmatics information and our method.

method	micro average <i>F</i> (%)	macro average <i>F</i> (%)
using pragmatics information	52.4	48.1
not using pragmatics • • information	20.2	16.2

ng(q) shows the number of correct answers of one questions in the test collection. *GTM* means the maximum values of the number of correct answers in the test collection, and is defined as $GTM = \max\{ng(q); g\}$. The number of answers returned by the system is assumed to be $K = \min\{4 \times ng(q); 2 \times GTMg\}$. *AR* is defined as the average value of *R(q)*, which is a fraction of the *K* relevant documents to the correct retrieved documents.

Fig.6 shows the comparative experiment results of the correct answer rank between the method with pragmatics information and the one without it. It is understood that the rank of the correct answer extracted by using the proposal method is much higher. Additionally, the proposal method has larger *AR* that the correct retrieved documents in first *K* retrieved documents is much more than the one without pragmatics information.

Table 4 shows the comparison result of average *ANMRR* and average *AR* without categorization. The proposed method derived correct documents with higher rank and much more correct documents in first *K* retrieved documents.

8 Conclusion and future work

In this paper, in order to implement the QA system, the pragmatics information for the verses of the “Analects of Confucius” were extracted and a practical method to extract

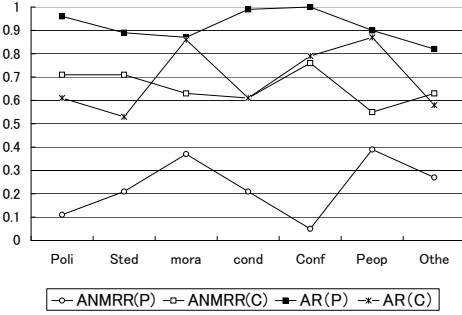


Figure 6. The experimental result of the rank of each category.

Table 4. Average ANMRR and average AR Comparison of no distinction between categories.

method	average ANMRR(%)	average AR(%)
using pragmatics information	23	92
not using pragmatics information	66	69

the deeper meanings and implicitly communicated meanings in the verse was proposed. The experiments verified that the method based on pragmatics and category information has better retrieval performance than the method without using pragmatics information at all.

By now, the pragmatics information extraction was performed manually. we will consider the semi-automatic pragmatics information extraction method in order to exclude the possibility to giving pragmatics information arbitrarily.

References

- [1] Harabagiu, Sanda M.: Deriving Metonymic Coercions from WordNet. In: Workshop on Usage of WordNet in Natural Language Systems, COLING-ACL '98 (1998)
- [2] Kurohashi, S. and Higasa, W.: Dialogue Helpsystem based on Flexible Matching of User Query with Natural Language Knowledge Base. In: Proceedings of 1st ACL SIGdial Workshop on Discourse and Dialogue, pp. 141-149, Hong Kong (2000)
- [3] Hu, H., Ren, F., Kuroiwa, S.: Chinese Automation Question Answering System of Specific-domain based on Vector Space Model. *EEJ Trans. EIS*, vol.125, no.5, pp.698-706 (2005)
- [4] Wang, S., Liu, Q., Bai, S.: A study on Question Answering system. In: ICCPOL2003, Shenyang, China, pp.498-506 (2003)
- [5] Kato, T, J. Fukumoto, F. Masui, N. Kando,: Are Open-domain Question Answering Technologies Useful for Information Access Dialogues?. *IPSJ SIG Notes*, 2004-NL-162, pp.145-150 (2004)
- [6] Matsuo, Y. • M. Ishizuka,: Keyword Extraction from a Document using Word Co-Occurrence Statistical Information. *Transactions of the Japanese Society for Artificial Intelligence*. vol.17,no.3-D, pp.217-223 (2002)
- [7] Zhang, H., Yu, H., Xiong, D. and , Q., HMMM based Chinese lexical analyzer ICTCLAS. In: Proceedings of 2nd SigHan Workshop, pp.184-187 (2003)
- [8] Leech, Geoffrey N.: *Pragmatics Information*. Kinokuniya SHOTEN (1987)
- [9] Kong, S.: *Analects of Confucius in Japanese, Chinese and English*. Shogakkukan Publishers, Japan (2000)
- [10] Nan, H.: *Original Ideas about Analects of Confucius*. Fudan University Press, Shanghai, China (1990)
- [11] Yang, B.: *Analects of Confucius translation and annotation*. Zhonghua Book Company, Beijing, China (1980)
- [12] Yang, S.: *Analects of Confucius Annotation and textual research*. Shanghai Guji Publishing, Shanghai, China (1982)
- [13] Fukumoto, F., Suzuki, Y.: Using Synonyms and Their Hypernymy Relations of WordNet for Text Categorization. *Transactions of Information Processing Society of Japan*, vol.43, no.6, pp.1852-1865, Jun (2002)
- [14] Manjunath, B. S., Ohm, J.R., Vasudevan, V. V. and Yamada, A.: Color and texture descriptors. *IEEE Trans. Circuits and Systems for Video Technology*, vol. 11, no. 6, pp.703-715, Jun (2001)

Table 5. Examples of systematization of “Analects of Confucius”

Chap-ter/ Verse	Category by Pragmatics information	Content of Analects of Confucius	Pragmatics information
1/7	Implicature	Zi Xia said, “I would grant that a man is indeed schooled if he revered virtues instead of beauty, devoted in serving his parents, exert himself to the utmost in serving his lord, and behave with trustworthiness with his friends, even though he is said to be not schooled.”	The emphasis of education is to cultivate personality.
1/14	Disambiguity	Confucius said, “The gentleman does not seek to satiate himself in eating, does not seek ease in living, quick in action, cautious in his speech and always use others who has achieved the high morality to put himself right. He can be seen as conscientious in his learning.”	The cultivated person satisfies with a poor living, and enjoys the road of the moral.
2/19	High-level explicature	Confucius said, “I conversed with Yan Hui and throughout the conversation, he did not ask, disagree or contradict, similar to someone who is slow to learn. But in private, his actions reflect what we have conversed, he is indeed not stupid at all.”	Confucius praised YanHui
12/19	Disambiguity	Ji Kang Zi asked Confucius about governance, “Should I use capital punishment on those that are immoral?” Confucius replied, “Is there really a need for the use of capital punishment? As long as you set a good example, others will follow. The virtues of the lord is like the wind while those of the commoners are like the grass. When the wind blows in a direction, the grass will bend towards that direction.”	The influence that the rulers has on the people is large.
13/7	Resolution of ellipsis	Confucius said, “The public policies in State of Lu and the State of Wei are brothers.”	The public policies in State of Lu and the State of Wei are similar.

A Japanese Language Model with Quote Detection by Using Surface Information

Manuel Medina González and Hirosato Nomura

Kyushu Institute of Technology

680-4 Kawazu, Iizuka, Fukuoka, 820-8502, Japan

manuel@dumbo.ai.kyutech.ac.jp, nomura@ai.kyutech.ac.jp

Abstract

In natural language processing, quotes are an important grammatical category which needs consideration. In this paper, we propose a Japanese language model that includes quotes as a category. The quotes are recognized by using surface information and dependencies between the words. Then, they are divided into direct and indirect speech. Finally, we extract the quotes and create a relation between them and the original text. After the text has been analyzed, we obtain a tree structure with all the elements hierarchically categorized. We have experimentally tested the accuracy of the parsing process by creating a prototype system. The results show a 67.29% overall correct quote detection.

1. Introduction

Correct identification of the parts that conform a text is an important task in natural language processing. In machine translation, the recognition of high-level grammatical categories helps to create heuristics and rules that improve the accuracy of the output. Thus, language models that consider elements beyond parts of speech, phrases or clauses are required.

Our ongoing research involves *subjunctive mood* analysis and decision in machine translation from Japanese to Spanish, and recognition of grammatical categories in Japanese texts is an important phase of our work. These categories include embedded sentences and *quotes* [2]. A Japanese language model created from this perspective is presented in [16], but it only considers sentences already extracted and lacks information about quotes, requiring human interaction to obtain the needed information.

We propose an improvement to that model by adding information about quotes to identify and extract them from large blocks of text. Our model uses dependency grammar and surface information to identify quotes even if the related elements are not in the same sentence or paragraph.

The rest of this paper is structured as follows: In section 2 we present previous works on the subject. In section 3 we describe the categories used in the model and the criteria used to separate the elements. Section 4 explains quotes in detail. Section 5 includes the algorithms used for parsing text. We empirically tested the accuracy of the model by creating a prototype system that analyzed articles from The Mainichi Shimbun and Yahoo Japan, and in section 6 we present the results obtained and discuss about them. Finally, in section 7 we present our conclusions and our future work.

2. Related works

The analysis of long Japanese sentences is a topic that has been widely researched. Kurohashi et al.[11] describe how the scope of conjunctive structures can be calculated by analyzing the similarity of *bunsetsu*. Takakura et al.[19] use grammatical and semantic features to find semantic relations between the clauses in complex sentences. In greater detail, Doi[6] performs sentence splitting to overcome the translation of long sentences, and Mathieu[15] uses extraction to retrieve the keyphrases in documents written in Japanese. However, in these studies quotes are not considered as a category. We are not aware of the existence of works who specifically deal with quote extraction in Japanese.

3. The model

Our model relies on two basic concepts: *Bunsetsu* (the smallest meaningful element in Japanese)[3] and *dependency grammar*[17].

3.1. Description.

The model consists of 6 hierarchical categories:

1. **Document.** It consists of a list of paragraphs. This is the top category.

2. **Paragraph.** A structure that contains:
 - (a) A list of Bunsetsu
 - (b) A list of quotes.
 - (c) A list of *embedded structures*.
 - (d) A list of *conjunctions*.
3. **Quote.** A *type of document* that expresses a message said or written by someone. It is explained in detail in Sect. 4.
4. **Embedded structure.** Any predicative element that modifies a noun or a pronoun becomes, along with its own modifiers, an embedded structure. This structure can be as simple as a sole element (for instance, an adjective) or as complicated as a complete *paragraph*. For instance, in Fig. 1a the verb “買った” (past tense of “to buy”) is properly identified as an embedded sentence.

Medina et al.[16] state that this category is necessary for the analysis of subjunctive mood. In our research, we consider these structures as paragraphs instead of sentences. They can also be considered as documents if necessary, although it is unlikely to have a series of paragraphs modifying a single bunsetsu.
5. **Conjunction.** An element that joins two sentences, called *left* and *right*. If there are two or more conjunctions within a paragraph, the right sentence of the conjunction in the position i is also the left sentence of the conjunction in the position $i + 1$. A paragraph with n conjunctions has $n+1$ sentences.

The separation of conjunctions (and hence, of sentences) is based on the process shown in [11]. The following modifications were made for our model:

 - (a) The types of conjunctions we consider are 接続助詞 (subordinate conjunctions), 接続詞 (coordinate conjunctions)[3, 14], and predicates in *Renyou-kei* (the ending form of certain parts of speech which allows them to modify predicative elements).
 - (b) When a coordinate conjunction is found at the beginning of a paragraph, we attach the last sentence of the previous paragraph as its left sentence, creating a relation between different paragraphs.
6. **Sentence.** Any set of bunsetsus related directly or indirectly to the same predicate. It is possible that a sentence contains only one element (the predicate itself). In Japanese, there are 4 elements that can become predicates: *verbs*, *adjectives-i*, *adjectives-na* and the copula だ (da) and its derivations です (desu) and である (dearu).

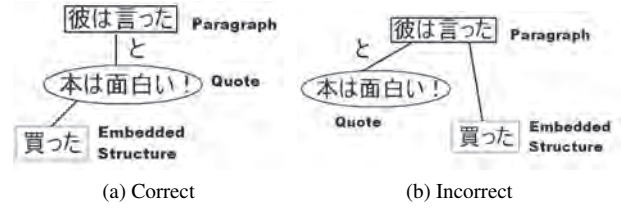


Figure 1: Importance of the order of the analysis

When analyzing texts, it is important to maintain this hierarchy in order to avoid incorrect detections. For instance, analyzing the text “彼は「買った本は面白い！」と言った” (He said “The book I bought is interesting!”) will result in figure 1a. Here, the analysis is correct because the quote is extracted first and then the embedded structure is extracted and attached to the quote. However, extracting the embedded structure first will result in figure 1b. This is incorrect because the embedded structure is attached to the paragraph, losing all the relations to its real container (the quote).

4. Quotes

A quote is essentially a passage referred to, repeated, or adduced. For instance, “Hello!” is considered a quote in the sentence ヒルさんは私に日本語で「こんにちは!」と言いました (Mr. Hill said to me in Japanese: “Hello!”).

In Japanese, quotes are identified with: 1) a particle that indicates quotation, and 2) a verb of *communication* or *report*.

4.1. Quotes in Our model

We extend the definition of quote by allowing the use of a wider set of verb categories. As we can verify in [2] and [9], there are cases of *utterances* that can be analyzed almost the same way as quotes.

Kudo[10] defines a series of categories for verbs in Japanese. According to them, we create a list with verbs from the following categories: *Human recognition activities*, *language activities*, *representation activities*, *thought*, *emotion*, *perception*, *sensation*, *relation*, and *performance*. The list is not definitive and can be adjusted as necessary.

4.2. Issues with quotes in Japanese

It is well known that Japanese is a language where many elements are normally omitted in both oral and written conversation. Quotes are no exception. However, there are cases where surface information provides clues to determine whether if a given text is or has a quote. We identify the following scenarios:

- The verb does not belong to a quote-related category.
- The elements are not directly related to the verb.
- The verb is in the same paragraph, but not in the final position.
- The verb is not in the same paragraph.

The next subsections explain in detail our approach.

4.3. Particles

1. The element that contains the quote particle must not be an adverb. Example (1) shows this case.

(1) はっきりと 言いなさい!
Say it clearly!

2. If the quote particle is “と”, it must not have *conjunctive properties*. This occurs mainly when it is used as a conditional marker, as illustrated in example (2)

(2) タイヤは 古いと 危ないですよ。
Tires are dangerous if they are old.

4.4. Dependencies and parts of speech

1. If the dependency of the element containing a quote particle is with a noun, identify whether if the noun derives from a verb in the list, or if it is a *サ変名詞* (*sahen noun*). *Sahen* refers to nouns that can become verbs by adding the verb “する” after them, as in “勉強”(study) ⇒ “勉強する” (to study).

- If it derives from the list, we have a quote.
- If it is a *sahen noun*, verify if there is a verb in the list conformed by the noun stem + “する”. In example (3), there’s no main verb, but we observe that “説明” (explanation) has a relationship with the quote candidate (in this case, the text between quotation marks). The verb “説明する” exists and it is included in the list. Thus, we conclude there is a quote.

(3) 彼女は現在主婦業を特訓中で「カレーの作り方を勉強しています」と説明。
She is now in special training to become a housewife, and explains, “I’m studying how to make curry”.

2. If the dependency is with an adverb, as in example (4), the quote is identified only if the adverb is the last element in the list of *bunsetsu*.

(4) ダルビッシュは「まだやらなきゃいけないことがあるので、日本一に突き進みたい」とキッパリ。
Darvish clearly stated: “As there are still many things I have to do, I want to rush for becoming Japan’s number one”.

3. Every time we find a verb included in the list, we remember its position in the list of *bunsetsu*, the paragraph in which it is contained, and the position of the paragraph in the document. It does not matter if it is not being used to quote. In case there is no verb when analyzing a possible quote, we create a relationship between the last element of the possible quote and the last identified verb contained in the list. This technique allows us to identify quotes even if the related elements are not in the same paragraph.

4.5. Direct and indirect speech

The types of speech in Japanese is a topic still discussed. Florian[4] presents the existence of both direct and indirect speech. Maynard[5] introduces two more concepts: “Direct-oriented indirect style” and “Content-oriented indirect style”. In this research, we consider only the former, because we do not need a more specific separation for the analysis of mood in our present studies. Some of the cases we consider are studied in [2].

Direct speech refers to mention exactly, word for word, what someone is saying or has said. In contrast to English and Spanish, the use of quotation markers in Japanese (“ ” and “ ”) is not always necessary[4]. A quote is in direct speech if: 1) it has quotation marks or 2) the predicate uses the auxiliary verb “ます”, particles used at the end of a sentence (like “よ” or “ね”), or is in *imperative* mode. Examples (5) and (6) show cases with and without quotation marks.

- (5) 彼女は「お前と踊らない」と言いました。
She said “I won’t dance with you”.
- (6) 兄は早く帰った方がいいよと言ってくれました。
My elder brother said to me “It’s better for you to come back home quickly”

Indirect speech refers to *report* what someone said. There is no need to repeat the message word by word, and thus quotation marks are also unnecessary. Most of the paragraphs that do not use communication-related verbs (utterances) fall in this category. A quote is in indirect speech when: 1) It does not have quotation marks, and 2) the predicate of the quote does not have the auxiliary verb “ます”, particles used at the end of a sentence and is not in *imperative* mode. Indirect speech is shown in examples (7) and (8).

- (7) 彼女は私と踊らないと言いました。
She said she will not dance with me.
- (8) 彼女は頭がいいと彼らは思っていない。
They do not think she is intelligent

5. Parsing algorithms

First, we divide the input text in *chunks* (plain strings of text), using linefeeds and the Japanese period (“.”) as delimiters except when a quotation-start marker (“「”) is encountered. In this case, as long as the quotation-end marker (“」”) is not found, we do not separate chunks to avoid separating the elements inside a possible quote. Then, for each chunk we perform morphological and syntactical analysis, obtaining a list of bunsetsu as result. Next, we apply to that list the process explained in Algorithm 1. We delegate to other functions the recognition of quotes and embedded sentences. Those functions will call this main function recursively to analyze the elements that each category contains, and then they extract it from the original text. Last, we search for conjunctions in the remaining elements.

We perform two special steps to make the analysis easier:

1. If the particle “の” (no) appears as a 形式名詞 (*keishiki meishi*, a special type of noun that accompanies predicative elements and make them function as non-predicative elements), we extract it from the original bunsetsu, then we create a new bunsetsu in the position $originalbunsetsu + 1$ add the element “こと” to it, and modify all the relationships in the paragraph.
2. If a sahen verb appears in *potential mode* (verb stem + “できる” (can)), we change できる for “する” (to do), and create two new bunsetsus: one in the position $sahenverb + 1$ that includes the keishiki meishi “こと” and the particle “が”, and the other in the position $sahenverb + 2$ including the verb “できる” to it. Finally, we modify all the relationships in the paragraph.

The recognition process for quotes is divided in two steps: To identify quotes using *direct speech* and to identify quotes using *indirect speech*. The order of the analysis is important because there might be quotes inside other quotes, specially in the case where quotation marks are present. We analyze a list of bunsetsu sent from Algorithm 1 and return a list containing all the quotes found, with their elements properly analyzed. The analysis of direct speech is also divided in the two cases presented above. In Algorithm 2 we present the case when there are quotation markers in the text, and Algorithm 3 shows the case without them. As for indirect speech, the recognition process is the same as in Algorithm 3, excluding line 12.

Algorithm 1: Analysis and extraction. The algorithm is applied to all the chunks obtained from the original text, after morphological and syntactical analysis. Created paragraphs are attached to a document structure.

```

1 function Analyze (List of Bunsetsu B) returns Paragraph
  /* Handle_quotes calls Algorithms 2 and
  3. Found quotes are analyzed
  recursively and extracted from B */
2 List of Quote Q ← Handle_quotes(B)
  /* All found embedded structures are
  analyzed recursively and extracted
  from B */
3 List of Embedded Structure E ←
  Handle_embedded_structures(B)
  /* With the remaining elements B */
4 List of Conjunction C ← Handle_conjunctions(B)
5 Add Q, E and C to Paragraph pb
6 return pb
7 end function

```

Algorithm 2: Quote detection: Direct speech with quotation markers

```

1 function Handle_direct_speech(List of Bunsetsu B) returns
  List of Quote
2 Q : List of Quote, bi : Bunsetsu in position i
3 i ← 0
4 while i < |B| do
5   bs ← Find bunsetsu with quotation-start marker,
   starting from bi
6   be ← Find bunsetsu with quotation-end marker,
   starting from bs
7   Create a List of Bunsetsu Btemp that contains all the
   bunsetsus between (and including) bs and be
8   predicate ← Find predicate for Btemp
9   if Btemp is a quote then
10    Attach to quote the result of
       Analyze(Btemp) /* recursive call
       to Algorithm 1 */
11    Extract from B all the elements contained in
       Btemp
12    Create relation between quote and predicate
13    Attach quote to Q
14  end
15  i ← (Position of be in B) + 1
16 end
17 return Q
18 end function

```

Algorithm 3: Quote detection: Direct speech without quotation markers

```

1 function Handle_direct_speech(List of Bunsetsu B)
  return List of Quote
2   Q : List of Quote, bi : Bunsetsu in position i
3   i ← 0
4   while i < |B| do
5     predicate ← Find predicate for B
6     if predicate ∈ B then
7       end ← Position of predicate in B
8     else
9       end ← last position in B
10    end
11    be ← Search in B for a bunsetsu that contains a
        quote particle, up to position end
12    Check if be satisfies the conditions to consider
        this a quote with direct speech
13    Create a List of Bunsetsu Btemp and add the
        result of getModifiers(be)
        /* Btemp has all the bunsetsus
           we need to analyze */
14    if Btemp is a quote then
15      Attach to quote the result of
        Analyze(Btemp) /* recursive
        call to Algorithm 1 */
16      Extract from B all the elements in Btemp
17      Create a relation between quote and
        predicate
18      Attach quote to Q
19    end
20    i ← (Position of be in B) + 1
21  end
22  return Q
23 end function

```

Example. Expressing the text “昨日見た映画でニューヨークが綺麗に映っていたので、娘はアメリカに行きたいと言っている” (My daughter is saying she wants to go to the United States because New York looked beautiful in the movie we saw yesterday) in our model, results in the structure illustrated in figure 2.

6. Experiments and results

In order to test the accuracy of our model, we implemented it in a prototype system. The conditions for the experiments were as follows:

Morphological and syntactical analysis. In order to perform the necessary morphological and dependency analysis, we use two external tools: JUMAN and KNP. The former is a morphological analyzer and the latter is a sentence struc-

Table 1: Results

		Expected	Correct	%
Direct Speech	(A)	91	50	54.94%
	(B)	193	134	69.43%
Indirect Speech	(A)	50	37	74%
	(B)	146	102	69.86%
Total	(A)	141	87	61.70%
	(B)	339	236	69.61%

ture analyzer. Both were created by the University of Kyoto. Our system’s input is the output obtained from those systems.

List of verbs. We obtained a list of words (including nouns and verbs) from *Bunrui Goiho*[8], which is one of the most used Japanese thesauri. A total of 784 words were extracted.

Test sets. We performed the experiments in two data sets: (A) 50 articles from the database of the *Mainichi Shimbun* from 1992, and (B) 84 articles obtained randomly from Yahoo Japan Topics (<http://dailynews.yahoo.co.jp/fc/>), from October 2007. The contents were converted to raw text in UTF-8 encoding. The headline and the date were ignored.

Output. The system returns a document with the parsed contents of the article, the number of paragraphs, embedded structures, and quotes found. Optionally, it can count the number of quotes in direct and indirect speech, and can output a JPEG or PDF image of the obtained structure, as in Fig. 2.

Table 1 shows the obtained results. All the articles were previously read and examined manually. Then, the results were compared against the output of the system. An element is considered to be correct only if it matches exactly with the results obtained by humans. The difference between the results are due to how the articles are written: some include quotes that are more easily identified by the system than others. The general average of correct detection was 67.29%

6.1. Discussion

The incorrect results include quotes not detected because there was a verb in the paragraph and a relation existed between the quote and it, but a communication verb is omitted. For instance, in 「金を出せ」と脅した (He/She threatened me saying “Give me the money”), it is clear that the verb “to say”, used in the English translation, is not in the original text. If we only want to make these quotes *detectable*, we could use a rule similar to the one created for quotes ending with adverbs. However, in both cases, we can not decide correctly the omitted verb without giving knowledge to the system. In the previous example, “to shout” could

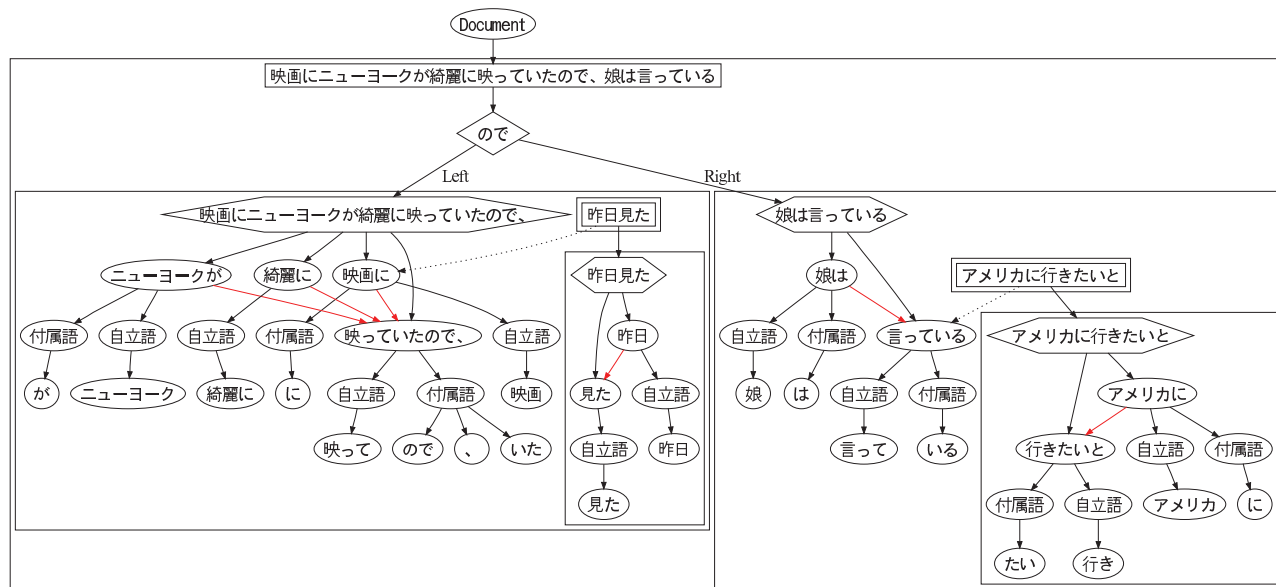


Figure 2: The input text after our model was applied.

have been used instead of “to say”.

The system performed well on recognizing quotes where the elements were not in the same paragraph. However, independent quotes were not detected when there was no previous verb of communication. Expressions like “we received the following comments” were found in some cases: as the verb “to receive” does not fall on the categories considered for this research, the quote is not detected.

As we considered only verbs and sahén nouns as predicates for possible quotes, when expressions like 明らかにする (to make clear) appear, the quote was not recognized. This issue can be solved easily by adding these expressions to the list of verbs. Also, the use of the expression ～という (called ～) caused the system to incorrectly detect quotes in sentences like 山田という人 (A person called Yamada). The solution can be implemented by adding special rules when these expressions are encountered.

7. Concluding Remarks and Future Work

We proposed a model of Japanese language that is capable of identifying quoting structures in large blocks of text using dependencies between the elements and surface information. It distinguishes if the quotes are using direct or indirect speech. Based on the structures obtained from our model and the accuracy achieved in the experiments, it will become a framework for future works. By considering quotes as a grammatical category, mood analysis could become easier and more precise due to the creation of rules

based on this model. We also believe our technique could be used in other works related to natural language processing, like text summarization.

References

- [1] A. Ahern and M. Leonetti. The Spanish subjunctive: Procedural semantics and pragmatic inference. In *Current Trends in the Pragmatics of Spanish*, pages 35–57. John Benjamin Publishing Company, 2004.
- [2] J. Borrego, J. Asencio, and E. Prieto. *El subjuntivo. Valores y usos*. Sociedad General Española de librería, S.A., 9 edition, 2003.
- [3] Conference on Research about Junior-High-School Japanese Education in the area of Kyushu, editor. *Beautiful Japanese* (美しい日本語). Conference on Research about Junior-High-School Japanese Education in the area of Kyushu, 1998.
- [4] F. Coulmas. *Direct and Indirect Speech*, chapter Direct and Indirect Speech in Japanese, pages 161–178. Mouton de Gruyter, 1986.
- [5] F. Coulmas. *Direct and Indirect Speech*, chapter The particle -o and content-oriented indirect speech in Japanese written discourse, pages 179–200. Mouton de Gruyter, 1986.
- [6] T. Doi and E. Sumita. Input Sentence Splitting and Translating. In *Proceedings of the HLT-NAACL*, 2003.
- [7] M. Jary. Mood in Relevance Theory: A Re-analysis Focusing on the Spanish Subjunctive. *UCL Working Papers in Linguistics*, 14:157–187, 2002.
- [8] W. Kawashino, T. Nakano, and I. Masahiko. “bunrui goi-hyo” as a resource of information processing : Compilation

- strategy and treatment of technical terms. *Journal of Japan Society of Information and Knowledge*, pages 12–28, 2000.
- [9] K. Koike. *Methodology of Spanish Composition* (スペイン語作文の方法). Dai San Shobo (第三書房), 2002.
 - [10] M. Kudo. *System and Text of Aspect and Tense - Expressions of Time in Modern Japanese* (アスペクト・テンス体系とテキスト-現代日本語の時間の表現). Hitsuji Shobo (ひつじ書房), 1995.
 - [11] S. Kurohashi and M. Nagao. A Syntactic Analysis Method of Long Japanese Sentences Based on the Detection of Conjunctive Structures. *Computational Linguistics*, (20):507–534, 1994.
 - [12] S. Kurohashi and M. Nagao. KN Parser : Japanese Dependency/Case Structure Analyzer. In *Proceedings of the Workshop on Sharable Natural Language Resources*, pages 48–55, 1994.
 - [13] S. Makino and M. Tsutsui. *A dictionary of Basic Japanese Grammar*. The Japan Times, 1995.
 - [14] S. Makino and M. Tsutsui. *A dictionary of Intermediate Japanese Grammar*. The Japan Times, 1995.
 - [15] J. Mathieu. Adaptation of a Keyphrase Extractor for Japanese Text. In *Proceedings of the 27th Annual Conference of CAIS-99*, pages 182–189, 1999.
 - [16] M. Medina González and H. Nomura. An Approach to Spanish Subjunctive Mood in Japanese to Spanish Machine Translation. In *AI2007: Advances in Artificial Intelligence*, pages 744–748, 2007.
 - [17] J. Nivre. Dependency Grammar and Dependency Parsing. Technical report, Växjö University: School of Mathematics and Systems Engineering, 2005.
 - [18] T. Takagaki. Noun Modification in Japanese and Spanish (日本語とスペイン語の名詞修飾). In *Japanese and Spanish* (日本語とスペイン語), pages 1–28. The National Institute for Japanese Language, 1994.
 - [19] S. Takakura, D. Han, and T. Furugori. An Experiment for Determining Semantic Relations Between Main and Subordinate Clauses in Complex Sentences. In *Proceedings of the WISICT*, 2004.
 - [20] N. Tsujimura. *An introduction to Japanese Linguistics*. Blackwell publishing, 2002.
 - [21] H. Ueda. The Japanese Particle “wa” and Spanish Subjunctive Mood (日本語の「は」とスペイン語の接続法). *The Society for Japanese Linguistics*, 21, 2002.

Machine Learning and Data Mining

Genetic Support Vector Classification and Feature Selection

Iván Mejía-Guevara
Instituto de Investigaciones en Matemáticas
Aplicadas y Sistemas,
Universidad Nacional Autónoma de México,
Circuito Escolar S/N, CU,
04510 D. F., México
imejia@uxmcc2.iimas.unam.mx

Ángel Kuri-Morales
Departamento de Computación,
Instituto Tecnológico Autónomo de México,
Río Hondo No. 1,
01000 D. F., México
akuri@itam.mx

Abstract

An important issue regarding the design of Support Vector Machines (SVMs) is considered in this article, namely, the fine tuning of parameters in SVMs. This problem is tackled by using a self-adaptive Genetic Algorithm (GA). The same GA is used for feature selection. We validate our results implementing some statistical tests based on single domain benchmark data sets, which are used for comparison with other traditional methods. One of these methods is commonly used for the selection of parameters in SVMs.

1. Introduction

The main purpose of this paper is to deal with the optimal design of Support Vector Classifiers (SVC). This method has been used in many practical applications with remarkable success [1] [2] [3].

When Vapnik and his co-workers [4] developed the theory behind the SVMs it seemed that we had an approach that necessitated of less casuistic participation during the design stage. However, the system's efficiency and efficacy relies on the proper selection of its free parameters; particularly, kernel and regularization ('C') parameters. Hence, we were faced with a shift from heuristic decisions of one kind to another, however subtler. This realization was one of the main motivations behind our work. We decided to attempt the automatic determination of C. This we have achieved using an efficient optimization algorithm, namely a GA. But GAs also require the careful insertion of parameters. In order to avoid a new shift of heuristics we introduced the use of self-adaptive GAs. Even if we still have to establish a range for the GA's parameters, such choice is simpler and less critical. In fact, unless the designer determines

obviously improper ranges, the GA in question will converge to like values. Another important task which is possible by using GAs is feature selection, which is also considered in this paper.

The article is organized as follows. We discuss in section 2 some theoretical characteristics of SVC. In section 3, we describe a self-adaptive Genetic Algorithm approach, which is combined with the SVC to define a Genetic Support Vector Classifier (GSVC). In section 4 some discussion regarding other methods used in the past for tackling this problem is presented. In section 5 we describe some experiments based on some statistical tests for single domain benchmark data sets. The comparison is performed against other statistical methods widely used in the past. Finally, some conclusions are presented in section 6.

2. Support vector classification

A SVM is a system for efficiently training linear learning machines in a high-dimensional feature space, nonlinearly related to input space. A SVM constructs models that are complex enough, yet it is simple enough to be analyzed mathematically [5]. It allows us to construct functions from a set of labeled training data, where these may either correspond to classification or nonlinear regression.

For binary classification problems, given the training set $\tau = \{\mathbf{x}_i, y_i\}_{i=1}^N$, the objective of a SVM is to find an hyperplane or linear decision function which separates the observations in a training set which belong to a class from those in the other class. The SVM solves this problem by maximizing the separation margin among the closest points of both classes. For linearly separable patterns, a linear function can separate the two classes and the margin of separation is defined through the application of the

SVM. In the nonlinearly separable case, the problem cannot be solved with a linear discriminant function in the input space. Therefore, a set of variables $\{\xi_i\}_{i=1}^N$, which are called slack variables, is introduced to penalize the points that do not satisfy the linearly separation criterion. In other words, a slack variable is added for each misclassified point. These points are found in the wrong side of the class that they represent, in or outside the separation margin.

The last problem can be tackled with a SVM by solving a quadratic optimization problem. The representation that results from the geometric analysis is called primal form. The primal form is as follows:

$$\begin{aligned} \text{Min} \quad & \frac{1}{2} \mathbf{w} \cdot \mathbf{w}^T + C \sum_{i=1}^N \xi_i \\ \text{s.t.} \quad & y_i(\mathbf{w} \mathbf{x}_i - b) \geq 1 - \xi_i \quad i=1, \dots, N \end{aligned} \quad (1)$$

In practice, however, it is better to deal with the Wolfe dual problem; derived from the Lagrange function of the primal problem and the Karush-Kuhn-Tucker (KKT) conditions [5]. The dual form is as follows:

$$\begin{aligned} \text{Max} \quad & \sum_{i=1}^N \alpha_i - \frac{1}{2} \sum_{i=1}^N \sum_{j=1}^N \alpha_i \alpha_j y_i y_j K(\mathbf{x}_i, \mathbf{x}_j) \\ \text{s.t.} \quad & \sum_{i=1}^N \alpha_i y_i = 0 \\ & 0 \leq \alpha_i \leq C \quad i=1, \dots, N \end{aligned} \quad (2)$$

The elements of vector $\mathbf{a}$ in (2) are known as Lagrange Multipliers (LMs) and their number corresponds to the number of restrictions in the primal form (N).

The so-called regularization parameter, C, is an upper bound of the LMs and it is used to penalize the points which are misclassified. There is a trade-off between the magnitude of the separation margin and the number of admissible misclassified points which is controlled by C. If C is low the classification capacity of the machine is sacrificed and the margin of separation is large. On the other hand, for a large C the classification capacity becomes a priority and the margin is smaller. In particular, when C tends to infinity the problem is equivalent to one with linearly separable patterns. The proper determination of this parameter is, as may be inferred, vital to the adequate design of a SVM and its analysis is one of the main objectives of this paper.

The $K(\mathbf{x}_i, \mathbf{x}_j)$ functional is called a kernel and it is used to project the input space to a higher dimensional feature space. In this new space a nonlinearly separable patterns set has a high probability to become a linearly

separable one. It is important to notice that there are infinitely many functions which can be used as kernels but only those which fulfill Mercer's conditions [5] should be used. Other important issue regarding the design of SVMs is the decision about which kind of kernel to be used during the training process. In practice, very popular kernel functions are: polynomial, radial basis function (RBF) and perceptron. Regarding the selection of kernel function, one or more parameters should be defined in order for the SVM to work properly. We illustrate, for instance, the functional form of the RBF kernel, as follows:

$$K(\mathbf{x}_i, \mathbf{x}_j) = \exp\left(-\frac{1}{2\sigma^2} \|\mathbf{x}_i - \mathbf{x}_j\|^2\right) \quad (3)$$

In (3), the width σ has to be properly selected in order for the SVM to operate efficiently. Typically, the best choice of this parameter has depended on subjective criteria. Its automatic determination is, therefore, desirable and is also dealt with in this paper.

The corresponding discriminant function (which corresponds to a classification surface) is constructed by using the LMs resulting from the optimization process and is given by the following expression:

$$f(\mathbf{x}) = \sum_{i=1}^N \alpha_i y_i K(\mathbf{x}_i, \mathbf{x}_j) + b \quad (4)$$

b in (4) is the “threshold” and it can be computed from $\mathbf{a}$. From (4) it is clear that the values for which $\alpha_i > 0$ are the ones that really matter in the definition of the DF and, for this reason, they are called “support vectors”. A fundamental result of this technique (from which its name is derived) is that the proper choice of a subset of the training set (which depends of the said support vectors) allows the best determination of the parameters of this function. Geometrically, the support vectors are located in the border of the separation margin on both sides of the plane. The kernel function is also used in (4) where the resulting surface, although nonlinear in the original input space, is linear in the space projected by $K(\cdot, \cdot)$.

3. Genetic support vector classification

3.1. Self-adaptive genetic algorithm

Genetic Algorithms were formally introduced in the 1970s by John Holland and his students at the University of Michigan. Their characteristics have made them attractive for some types of optimization. In particular, GAs work very well on mixed (continuous and discrete) combinatorial problems. They are less

susceptible to getting stuck at local optima than gradient search methods [6].

When using GAs the following issues must be taken into account to ensure their proper functionality: a) population size, b) genome representation, c) fitness function, d) selection method, e) probabilities for crossover and mutation and f) termination criteria [7].

We considered a population of size $nP=30$. This is a relatively small population size and we emphasize that our method (as will be discussed in the sequel) performed quite adequately with it. The initial population was randomly generated. GAs may be modeled by finite homogeneous Markov chains [8]. Their transition matrix is known to converge to the same fixed point when repeatedly multiplied by itself. Since the transition of a GA from one generation to another may be equated to such repeated matrix product, a corollary is that the GA will converge to the same results regardless of the initial population. Weighted binary fixed point representation was used. A number is represented by a sign bit ('1' corresponds to a negative sign); I bits for the integer part and D bits for the decimal part. For example, if $I=2$ and $D=6$ the binary string '101010000' corresponds to -1.25. With this representation the range of possible values for a real r is $-2^I + 2^{-D} \leq r \leq 2^I - 2^{-D}$. It is obvious that search and optimization when attempted (in a digital computer) over a discrete landscape is optimal when such space is encoded in binary, for the grain of the resulting discrete grid cannot be made smaller. In other words, assuming that our programs are running in a digital binary computer, any non binary encoding will ultimately miss part of the usable (binary) space. The matter of the best binary encoding, though, is a different matter. There have been several attempts to show that codes which 'try' to avoid the so-called Hamming cliffs will outperform more conventional ones. When representing real variables, the disadvantage (in terms of the Hamming cliffs) of using floating point vs. fixed point representation is obvious. Floating point variables may jump several orders of magnitude as the result of changing a single bit.

Once the initial population is generated, the so-called Vasconcelos' Genetic algorithm (VGA) is used. This approach is intended to capture [9] the main characteristics of the Idealized Genetic algorithm (IGA) [7]. Its main characteristics are: a) Total elitism (equivalent to the $\mu + \lambda$ scheme in evolutionary strategies), b) Deterministic coupling during selection and, c) Annular Crossover. The genome is considered to be a ring of size ℓ . Individuals i and $n-i+1$ are deterministically selected as candidates for crossover during selection. A random number is generated; if it is smaller than P_c (the probability of crossover) then a

semi-ring of size ℓ is taken from the same position for each of the two parents; the resulting genomes pass on to the next population. Otherwise, the individuals are passed to the next population untouched. Uniform mutations occur with probability P_m . Given annular crossover, the genome is a ring in which the first bit and last bits in the string are contiguous. The exchange is given between semi-rings, which is equivalent to the switch of two not adjacent portions in the genetic string. It is easy to see that annular crossover is equivalent to the well known two-point crossover. The schemata in the chromosome are disrupted with higher probability than in the typical 1-point crossover scheme. However, annular crossover makes the algorithm less susceptible to the way the encoding scheme affects the dynamics of the evolutionary process.

The most important characteristic of this kind of GA is the use of deterministic selection together with strong elitism. This combination results in a genetic pool richer in new schemata (because "best" and "worst" individuals are purposely combined) while retaining old schemata which have already been identified by the algorithm as the most adequate (because the best overall individuals are always retained) [9].

The two recognized reasons which hamper the efficiency of a GA are known to be deceptive functions and spurious correlation (also known as "hitchhiking"). The VGA, in this regard, is related with the IGA because a) The possible problematic (deceiving) *locus* are destroyed with high probability because of the best-worst crossover approach, b) Spurious correlation is avoided since the mentioned crossover is impervious to hitchhiking bits, c) Total elitism always preserves the desired schemata and d) Over the generations the worst individuals are eliminated from the candidate population whereas the best ones are retained in the $1/k$ percentile.

The VGA has proved its superiority over other optimization algorithms (several GA variations and a random mutation hill climber) as discussed in [9] where its efficiency was statistically tested over a large (more than 70,000) set of problems.

The self-adaptive mechanism used here consists on including the crossover and mutation parameters in the genome. These values are subjected to the same operators as those of the problem proper and, therefore, are optimized along with the main problem itself. The idea is to leave the problem of their determination to the GA. This principle has two main avenues of implementation. The one chosen here is known as population self-determination. It was described and experimentally tested in [10]. It follows a democratic

principle where each individual of the population contributes with a proposal for the values of the parameters to be used during each generation. Thereafter the values derived from all individuals are averaged and are adopted by all the individuals in the population. Under this approach the crossover probability (P_c) for each generation is estimated by:

$\sum_{i=1}^{N_p} (p_c)_i / N_p$, where $(p_c)_i$ represents the crossover probability for an individual i in the population, $i = 1, \dots, N_p$. The same approach is used for mutation probability (P_m), where it is determined by $\sum_{i=1}^{N_p} (p_m)_i / N_p$ being $(p_m)_i$ the mutation probability for individual i in the population.

3.2. Genetic support vector classifier

In this article we propose a GA variation for the selection of parameters in SVMs. With our approach, it is possible to tackle the following problems simultaneously: a) The selection of parameters in the SVM, b) Feature selection and c) The calibration of the genetic probabilities of mutation and crossover. This last problem, in particular, deals with the matter of how to tune the GA's parameters dynamically; it is important because it has been shown that a GA's convergence rate is dependent on their proper adjustment during the evolutionary process.

The genetic approach we propose for the specification of parameters and variable selection in SVC was called a Genetic Support Vector Classifier. Its detailed implementation is as follows:

- 1) Define the genome for the representation of the parameters: C , σ , P_c , P_m and fS ,
- 2) Randomly generate the initial population,
- 3) Compute the fitness for each individual of the population,
- 4) Apply the GA (self-adaptive Vasconcelos' genetic algorithm),
- 5) If a termination criterion is reached, finish. If not, return to step 3.

Parameter fS in step 1) is a binary mask that will be used for the implementation of feature selection. For a deep analysis of this issue the reader may see [11]. Here we introduce a binary string of size equal to the number of independent variables of τ , where a '0' in position i means that the variable i must be dropped during the training process and a '1' means the opposite. Given the strong elitist property of VGA the problem of sacrificing fitness when some variables of the training set are not considered is avoided since the

evolutionary process begins with all variables, but a reduced subset of those variables is always kept.

Different alternatives may be considered for the fitness evaluation. We use the accuracy function which is defined as the proportion of well-classified observations in the test set. The accuracy function is computed for each individual in the population and for the new individual generated in the evolutionary process through the application of SMO. The kind of algorithm that is chosen in this phase is very important since most of the training time depends on its selection. The use of traditional optimization methods here is out of the question (the Chunking algorithm [12], for instance) because of the time these algorithms would need for training. Therefore, although some other algorithms can be used for training the SVM, we strongly suggest the use Sequential Minimal Optimization, an approach discovered by [13] which is an efficient and fast way to train SVMs.

Our stopping criterion is simply based on the number of generations. Because of the efficiency of SVGA we will show later that only 50 generations were needed to obtain a competitive machine. We confirm this by applying also the stopping criteria were either a) The computation of 300 generations or b) The fitness value does not improve during the last 50 generations; in contrast with [14], where the last criteria was proposed but using 600 generations in a) and 100 in b) instead. Moreover, 500 individuals were used in that application in comparison with the 30 that we used here. These differences imply a very significant decrease on the computational cost.

Once all the variables that will be specified in the genome have been defined the size of the genome for the GSVC is equal to $n_C + n_p + n_{p_c} + n_{p_m} + m$, where n_C , n_p , n_{p_c} y m are the number of bits used for the codification of C , p , P_c , P_m and the number of characteristics, respectively.

4. Related research

We have mentioned before that the accuracy of a SVM model is largely dependent on the selection of the model parameters. Therefore, we discuss in this section other approaches proposed in the past to deal with the parameter selection problem. In what follows we make a very brief review of some of them.

In [15] an algebraic approach is proposed for the determination of C and ε (which is used to define the so-called ε -insensitive loss function) in Support Vector Regression (SVR). In [14], [16], [17] genetic approaches are used to deal with the determination of parameters in classification tasks. A genetic method for

the solution of regression problems is treated in [18]. In [19] the use of grid search is considered. A grid search tries values for each parameter across the specified search range using geometric steps. However, grid searches are computationally expensive because the model must be evaluated at many points within the grid for each parameter. In [18] the use of grid method is reported as being very time consuming in comparison with a genetic method. Finally, we mention that simulated annealing has also been used for this purpose as in [20].

Similar genetic approaches have been used in [14] [21], but new characteristics are considered here that have proven their efficiency in the past: a) Fixed-point encoding, b) The use of a very efficient genetic strategy (VGA) and, b) The self-adaptive mechanism where P_c and P_m are determined as part of the evolutionary process.

In [22] a GA is used for feature selection and the optimization of C in SVMs. An important contribution of this approach is the use of theoretical bounds on the generalization error for SVMs instead of performing cross-validation. Although the authors mentioned the capability of their approach to optimize the kernel parameter, they use a linear kernel where no parameter is considered in all the experiments reported. This approach was found to achieve better and faster results when a fitness function based on these bounds was used, but if we observe the bound expressions used, the kernel function used during the training process is involved in the estimation of these bounds. Therefore, if the kernel parameter is optimized by means of the GA, the values of the bounds must be computed taking into account the changes of the kernel function every time the parameter of the kernel is changed by the evolutionary process and, therefore, its effect on the time and performance of this approach could be different. Another important issue is the use of decimal encoding when the number of features to be selected is known beforehand and a binary encoding otherwise. In our genetic approach we use binary encoding since, in general, the number of features to be selected is unknown. Another difference between both proposals is that, even though both GAs are elitist, ours is self-adaptive. Additionally, both GAs consider different strategies for recombination and mutation.

A good discussion regarding this topic may be found in [23], where the authors showed that although RBF appears to be the most popular kernel it does not always achieve the best performance. The authors reach that conclusion applying Genetic Programming (GP) to evolve a kernel for a SVM classifier (they call it GK SVM). Three commonly used kernels are used, Polynomial, RBF and Sigmoid, where large variation in accuracy among these kernels is reported according

to the experiments performed by the authors. Finally, the authors report the ability of the GK SVM to automatically find kernel functions that perform competitively in comparison with the widely used kernels, without requiring manual parameter insertion to achieve optimal performance. Although, it is important to note that parameter C was set to 1 for all the experiments. It is clear that GSVC has not been designed to deal with the optimal selection of the kernel functions. However, it can actually determine the kernel parameters, C parameter and feature selection.

5. Experimentation

Three classification problems are used here to illustrate the performance of the GSVC. The first dataset is the so-called German credit assignment problem (**german**) which consists of 1000 observations with 24 features. The second dataset has 690 instances and 14 attributes. This dataset is known as the Australian credit problem (**australian**). Finally, the **sonar** dataset is considered where the task is to train a network to discriminate between sonar signals bounced off a metal cylinder and those bounced off a roughly cylindrical rock. This dataset has 208 instances and 60 attributes. The **sonar** dataset was taken from UCI Machine Learning Repository while **german** and **australian** datasets were taken from the Statlog Project. All training sets, however, can be downloaded from [25].

Some classical statistical classifiers were used to compare the performance of the GSVC. We also applied the grid method (**grid**) for the selection of parameters C and σ . The statistical classifiers we used are: 1-nearest neighbor (**1-nn**), linear discriminant analysis (**lda**) and k-nearest neighbor (**k-nn**).

The validation was implemented a statistical approach proposed by [24], where a taxonomy of statistical questions in machine learning is presented, focused on single domain datasets. One of the tests proposed by the author, which is used here, is known as *5x2cv paired t test*, based in 5 iterations of 2-fold cross-validation. In each replication of the test, the available data are partitioned into two equal-sized sets τ_1 and τ_2 . The comparison of two different algorithms is made by subtracting their corresponding accuracy rates in the application of 2-fold cross-validation in each replication. The means and variances s_i^2 are also successively estimated and finally a t statistic is

defined as: $\tilde{t} = p_{(1)}^1 / \sqrt{\frac{\sum_{i=1}^5 s_i^2}{5}}$. This is called 5x2cv

$\tilde{t}$ statistic and it has an approximate t distribution with 5 degrees of freedom. The value of $p_{(1)}^1$ represents the difference between the estimated errors for the two algorithms in the first replication. The comparison of two algorithms is based on the null hypothesis that there is no difference between their performances. The

null hypothesis is rejected if $\tilde{t} > t_5$. In this work, we compare the accuracy resulted from the GSVC against the ones obtained from the other approaches proposed here. We used a RBF kernel, where σ is genetically determined. The results are reported in Table 1.

Table 1. Hypothesis tests for measuring classification accuracy

method	50 generations			>50 generations		
	german	australian	sonar	german	australian	sonar
1-nn	1.9501(<)	10.5799(>)	5.2768(>)	1.8897(<)	9.5250(>)	5.6037(>)
k-nn	2.3671(>)	13.1663(>)	6.0657(>)	2.3671(>)	13.1663(>)	6.0657(>)
lda	0.9796(<)	2.4561(>)	5.6414(>)	0.9796(<)	2.4561(>)	5.6414(>)
grid	1.6836(<)	0.3083(<)	1.4338(<)	1.6836(<)	0.3083(<)	1.4338(<)

In Table 1, the $\tilde{t}$ values resulted from the application the statistical tests are displayed, where “(>)” means that there is statistical evidence (with 95% confidence) to reject the null hypothesis and “(<)” means the opposite. In this Table we also show the results from the application of two terminating criteria for the GSVC. In the left hand side of the Table, the results using 50 generations are shown; whereas in the right side the criterion with more than 50 generations explained in section 3.2 are shown. We can appreciate that results concerning rejection of the tests are the same for both termination criteria, which indicates that our genetic approach is efficient since only 50 generations are sufficient to get competitive results. This is important since valuable computation time is saved. We can also appreciate from these results that the GSVC resulted superior in most of the cases in comparison with the other classifiers. The performance of GSVC was in general superior from 1-nn, k-nn and lda. For the case of grid method, however, there is no statistical difference between them, but grid method resulted computationally more expensive. Moreover, with this method feature selection could be possible but computationally unmanageable. In the case of GSVC, feature selection was implemented with no cost in performance.

Finally, we want to point out that all experiments were performed using the MATLAB interface of LIBSVM [25].

6. Conclusions

The SVC is presented in this paper, where the parameters included for their evolutionary optimization were: a) The regularization parameter, b) The kernel parameter, c) The probabilities of mutation and

crossover, d) Feature selection (from a string mask used to represent the independent variables). Once again, we stress the fact that we used an ad hoc genetic algorithm which does not display the limitations of more conventional ones.

The factors which make this method commendable are: 1) The calibration of parameters is automatic and efficient, 2) The average accuracy was statistically proven to be superior in comparison with other classifiers; 3) the number of independent variables during the training phase is reduced without sacrificing accuracy; 4) the use a fast self-adaptive GA. The evolutionary process is enhanced by dynamically varying the probabilities of crossover and mutation during the optimization process.

Although we only used RBF kernel in our experiments, we point out that the genetic approach is also useful for any kernel function that might be employed during the training phase of a SVM.

7. References

- [1] C. Cortes, and V.N. Vapnik, “Support-vector networks”, *Machine Learning* 20, Springer, 1995, pp. 273.
- [2] B. Schölkopf, C. Burges, and V.N. Vapnik, “Extracting support data for a given task”, *Proc. KDD*, 1995, pp. 252-257.
- [3] C.J.C. Burges, and B. Schölkopf, “Improving the accuracy and speed of support vector machines”, *Proc. NIPS*, 1996, pp. 375-381.
- [4] B.E. Boser, I. Guyon, and V.N. Vapnik, “A training algorithm for optimal margin classifiers”, *Proc. COLT*, 1992, pp. 144-152.

- [5] Schölkopf, B., Smola, and K. Müller, *Learning with kernels*, MIT Press, 2002.
- [6] Holland, J.H., *Adaptation in natural artificial systems*, University of Michigan Press, Ann Arbor, 1975.
- [7] Mitchell, M., *Introduction to Genetic Algorithms*, MIT Press, 1996.
- [8] G. Rudolph, "Convergence analysis of canonical genetic algorithms", *IEEE Transactions on Neural Networks* 5(1), 1994, pp. 96-101.
- [9] A.F. Kuri-Morales, "A methodology for the statistical characterization of genetic algorithms", *LNCS* 2313, 2002, pp. 79-88.
- [10] J. Galaviz, and A. Kuri, "A self-adaptive genetic algorithm for function optimization", *Proc. ISAI/IFIPS*, 1996, pp. 156.
- [11] I. Guyon, and A. Elisseeff, "An introduction to variable and feature selection", *Journal of Machine Learning Research* 3, 2003, pp. 1157-1182.
- [12] Vapnik, V.N., *Estimation of Dependencies based on Empirical Data*, Springer-Verlag, Berlin, 1982.
- [13] J. Platt, "Fast training of support vector machines using sequential minimal optimization", *Schölkopf, B., Burges, C.J.C., Smola, A.J., eds.: Proc. Advances in kernel Methods-Support Vector Learning*, MIT Press, Cambridge, MA, 1999, pp. 185-208.
- [14] C.L. Huang, and C.J. Wang, "A GA-based feature selection and parameters optimization for support vector machines", *Expert Syst. Appl.* 31(2), 2002, pp. 231-240.
- [15] V. Cherkassky, and Y. Ma, "Practical selection of SVM parameters and noise estimation for SVM regression", *Neural networks* 17(1), January 2004, pp. 113-126.
- [16] J.H. Min, and Y. Lee, "Bankruptcy prediction using support vector machines with optimal choice of kernel function parameters", *Expert Syst. Appl.* 28(4), 2005, pp. 603-614.
- [17] A. Kuri-Morales, and I. Mejía-Guevara, "Evolutionary training of SVM for multiple category classification problems with self-adaptive parameters", *Proc. IBERAMIA/SBIA'06, LNAI*, 2006, pp. 239-338.
- [18] I. Mejía-Guevara, and Á. Kuri-Morales, "Evolutionary feature and parameter selection in support vector regression", *Proc. MICAI-07, LNAI, Springer*, 2007, pp. 399-408.
- [19] C.W. Hsu, C.C. Chang, and C.J. Lin, "A practical guide to support vector classification".
- [20] P.F. Pai, W.C. Hong, "Software reliability forecasting by support vector machines with simulated annealing algorithms", *Journal of Systems and Software* 79(6), 2006, pp. 747-755.
- [21] S.H. Min, J. Lee, and I. Han, .C. Hong, "Hybrid genetic algorithms and support vector machines for bankruptcy prediction", *Expert Syst. Appl.* 31(3), 2006, pp. 652-660.
- [22] H. Fröhlich, "Feature Selection for Support vector Machines by Means of genetic Algorithms", *Master's thesis*, University of Marburg, 2002.
- [23] T. Howley, M.G. Madden, "The genetic kernel support vector machine: description and evaluation", *Artificial Intelligence Review* 24(3-4), 2005, pp. 379-395.
- [24] T.G. Dietterich, "Approximate statistical test for comparing supervised classification learning algorithms", *Neural Computation* 10(7), 1998, pp. 1895-1923.
- [25] C. Chang, and C. Lin, "LIBSVM: a library for support vector machines", 2001.

Cascading Decomposition and State Abstractions for Reinforcement Learning

Chung-Cheng Chiu¹ and Von-Wun Soo^{1,2}

¹Department of Computer Science, National Tsing Hua University
101, Section 2 Kuang Fu Road, Hsinchu, Taiwan, R.O.C
g944345@oz.nthu.edu.tw

²Department of Computer Science and Information Engineering,
National Kaohsiung University
700, Kaohsiung University Rd, Nan Tzu Dist., 811. Kaohsiung, Taiwan, R.O.C.
soo@cs.nthu.edu.tw

Abstract

Problem decomposition and state abstractions applied in the hierarchical problem solving often requires manual construction of a hierarchy structure in advance. This work is to provide some automatic algorithms for dimension reduction in problem solving. We propose cascading decomposition algorithm based on the spectral analysis on a normalized graph Laplacian to decompose the problem into several sub-problems and conduct parameter relevance analysis on each subproblem to perform dynamic state abstraction. In each decomposed subproblem, only parameters in the projected state space related to its subgoal are reserved, and identical sub-problems are integrated into one through feature comparison. The whole problem is transformed into a combination of projected sub-problems, and problem solving in the abstracted space is more efficient. The paper demonstrates the performance improvement on reinforcement learning based on the proposed state space decomposition and abstraction methods using a capture-the-flag scenario.

1. Introduction

In dealing with daily routine tasks, we could find solutions efficiently and make things done in a proper manner. But when it comes to autonomous problem solvers, even a simple task might be a challenge. One of the most important factors results in this difficulty is the dimensionality of the state space. To represent the properties of an object or an environment in real world requires so high dimensions that make most of effective methods in a low dimension lose their effectiveness. The limitation is known as *curse of dimensionality*. But in practice, many effective

algorithms for dealing with problems with high dimensionality have been proposed. The breakthrough is mainly based on the studies of dimension reduction. With these reduction methods, a problem is projected onto a low dimension space, and the new representation allows low-dimension algorithms available to work. The fundamental principle of dimension reduction is based on the features of data. A complex data could be composed of several ordered components. In some cases, determining the order and relations among the features provide a basis to represent the data in low dimension.

In this work we focus on the complexity reduction for reinforcement learning. Reinforcement learning is an effective learning framework for solving problems without much prior knowledge. Although it automates the learning process to find the solution and relieves the efforts to establish actual action policies for any given state, it often suffered from its high dimensional state space. This leads to the research field of hierarchical reinforcement learning. Prior work in hierarchical reinforcement learning like hierarchies of abstract machines (HAMs) [11], options framework [15], MAXQ [5], and programmable HAMs [2] rely on manual constructions of hierarchies. The major processes for hierarchical framework construction include problem decomposition and state abstractions, and many works have been proposed to automate the processes ([6], [7], [8], [9], [10], [13], [14]).

Previous approaches in problem decomposition often consider its processes as subgoal identification. Some approaches identify a subgoal as either a state that tends to have a high visited frequency or reward gradient [6], or a high visited frequency based on all successful trajectories [9]. Another kind of subgoal identification applies the concept of graph cut so that a subgoal can be defined as an *access state* that provides

an easy access to its neighboring regions [8], [10], [13]. Some performed graph cut [10] via conventional network flow analysis [1], and others applied normalized cut [13] by using spectral clustering algorithm [12] over an undirected graph. The graph that corresponds to problem solving can be derived from the entire state space [10], or constructed based on only those states updated through newly observations in order to reduce the problem complexity [13]. Many approaches provided binary separation that decomposed a problem into multiple subsets [10], [13], and applying these methods needs further implementation such as sampling data several times [13] or executing recursively on the decomposed components [10]. However, in data sampling, the required sampling time multiplies as the problem scales up and thus may result in high computation complexity. In recursive execution, it requires manual analysis on the separation results in order to determine whether a decomposition is appropriate for problem solving and if further execution is required. Other approaches [6], [8], [9], [14] that could find multiple separations usually need further empirical testing and analysis to design executable algorithms in practice. Due to the limitations, we propose a new method called cascading decomposition algorithm that automatically divides the problem into multiple subsets from a large-scale structure of the state space. The algorithm is based on spectral graph theory, and is described in later sessions.

For state abstractions, [7] proposed a robust approach to find state abstractions by applying statistical hypothesis-testing to test the relevance of state parameters. A limitation of their method is that their statistical hypothesis-testing analyzes each state individually. It restricts the analysis to a mere single state and ignores the relevance among the overall states, and it also requires many trails of learning algorithms to provide sufficient samples, which leads to high computation cost and could thus make it impractical. We adopt statistical hypothesis-testing in performing state abstraction in the framework but apply a different approach on selecting samples to relieve such problems. The details would be discussed in the session 5.

We propose a cascading decomposition algorithm, a modified irrelevant parameter identification method, and a shared-subtask learning algorithm, which correspond to problem decomposition, state abstraction and associated learning respectively. The cascading decomposition algorithm applies the spectral graph analysis on the state transition graph of a task and separates the graph into multiple components. Since the transition graph is a Markov network, we define the decomposition criterion based on the property of

conditional independence in the graphical model. Our irrelevant parameter identification method evaluates the relevance of parameters by using the statistical hypothesis testing on the projected states. The design aims to provide an analysis that is robust to the variance from multiple optimal policies. The cascading decomposition algorithm finds subtasks and irrelevant parameter identification that reduces their dimensionality, the shared-subtask learning algorithm constructs a control structure and models policies to combine their results and finds the optimal solution. We design a capture-the-flag scenario to study the overall performance of proposed algorithms in the complexity reduction for the reinforcement learning.

2. Spectral Graph Theory

Let $G = (V, E, W)$ denotes an undirected graph where V is the set of vertices, E is the set of undirected edges, and W denotes the set of edge weights. The *combinatorial Laplacian* L is defined as $L = D - W$ where D is a diagonal matrix whose entries are the row sums of W that is called a degree matrix. The projection of a function on the eigenspace of the Laplacian provides a smoothness measure on the underlying graph which is called *Dirichlet sum*

$$\langle f, Lf \rangle = \sum_{u \sim v} (f(u) - f(v))^2 w_{uv} \quad (1)$$

where $u \sim v$ indicate u and v are neighboring nodes. This fundamental smoothness property of the graph Laplacian implies that its measurement takes underlying manifolds into consideration instead of analyzing merely in Euclidean space. An alternative operator is the *normalized Laplacian* [4] and is defined in equation (2) as

$$\mathcal{L} = D^{-1/2}(D - W)D^{-1/2} \quad (2)$$

where D is a diagonal matrix with its entry $d(u, u)$ as d_u while its operation on a function f on a graph is defined in equation (3)

$$\mathcal{L}f(u) = \frac{1}{\sqrt{d_u}} \sum_{v \sim u} \left(\frac{f(u)}{\sqrt{d_u}} - \frac{f(v)}{\sqrt{d_v}} \right) w_{uv} \quad (3)$$

According to [4], the eigenvalues of normalized Laplacian relate better than standard Laplacian to some graph invariants for a general graph. Thus, we adopt the normalized Laplacian operator in cascading decomposition algorithm, and the following description focuses on the normalized Laplacian.

Referring to the *Dirichlet sum*, the eigenvalues and eigenvectors of the operator provide a solution to a sequence of minimization problems, where the eigenfunctions correspond to minimizing functions, and the eigenvalues are referred to the *Rayleigh*

quotient [4]. For the *normalized Laplacian* operator with a function $g: V(G) \rightarrow \mathbb{R}$, the *Rayleigh quotient* is defined in equation (4) as

$$\frac{\langle g, \mathcal{L}g \rangle}{\langle g, g \rangle} = \frac{\langle g, D^{-1/2} L D^{-1/2} g \rangle}{\langle g, g \rangle} = \frac{\sum_{u \sim v} (f(u) - f(v))^2 w_{uv}}{\sum_u f^2(u) d_u} \quad (4)$$

where $f = D^{-1/2}g$. With the orthonormality constraint, the quotient gives eigenvalues and the corresponding functions $g: V(G) \rightarrow \mathbb{R}$ that are orthogonal to the first eigenvector. The second eigenvector captures the geometry structure of the underlying graph such as *bottleneck* [3]. This property can also be expressed in terms of the *Cheeger* constant of a graph G

$$h_G(S) = \min_S \frac{|E(S, \tilde{S})|}{\min(\text{vol}_S, \text{vol}_{\tilde{S}})} \quad (5)$$

where S is a subset of vertices, $\tilde{S}$ is the complement of S , and $E(S, \tilde{S})$ is the set of edges between S and $\tilde{S}$, vol is the sum of the degrees of the vertices in a set. According to [4], the first non-zero eigenvalue of the *normalized graph Laplacian* operator $\mathcal{L}$ on a graph has

$2h_G \geq \lambda_1 > \frac{h_G^2}{2}$. This theorem shows that the

eigenfunction corresponding to λ_1 captures the geometry structure of the manifold and provides a binary classification, and is the fundamental property of spectral clustering for graph cut.

3. Cascading Decomposition

For performing dynamic abstractions, the purpose of a decomposition method is to provide a separation such that all sub-problems are as much independent as possible. We define the independent criterion based on the Markov property. In a Markov network, the probability of a state depends only on its neighboring states, and is independent on the rest of states if the probabilities of the neighboring states are given. This property is applied to reinforcement learning and is the basis of the problem solving framework. Two subtasks are independent given their connected states, thus to find separable sub-problems in reinforcement learning, the decompositions should be performed on the weaker connection edges between components in the state transition graph. This kind of notion is similar to that in the graph cut [1]. Previous works [10], [13] took graph cut to decompose the task environment. But for problems that should be decomposed into more than two sub-problems, these approaches would become more complex to apply. We propose the cascading decomposition method and its variant k-cascading

decomposition to decompose the graph into multiple components.

In the conventional spectral clustering, the algorithm finds the symmetric point in the second smallest eigenfunction and takes its value as separation criterion. The vertices with greater values than the separation value are classified into one class, and the others are classified into another class. In multiple clustering, cascading decomposition algorithm identifies *bottlenecks* of the state space to decompose the state transition graph into multiple components based on the smoothness measure in the eigenfunction. The *bottlenecks* are identified by finding relatively higher differences between neighboring vertices. For each pair of connected vertices, its edge is assigned as their value difference. An edge whose difference value is no less than any other edges connecting to its end vertices is identified as *bottleneck*. In some cases, the designer might want to specify the cluster number of a problem based on his prior knowledge or preference. We propose k-cascading decomposition algorithm to suffice such requirement. K-cascading decomposition algorithm is a slight modification of the cascading decomposition algorithm. It keeps disconnecting higher difference edges in the remaining edge set until the component number is no less than defined cluster number.

4. State Abstractions

State abstractions find state parameters that are irrelevant for determining the optimal policy within the state space. When the irrelevant parameters are determined and removed, the projection of a new policy function should provide the same optimality. To find irrelevant parameters, the analysis method should determine whether the projection of a state parameter is a sound abstraction. Jong & Stone [7] define a criterion for irrelevance of a state parameter x in terms of action-value functions Q as

$$\exists_a \forall_{s'=[s]x} \forall_{a'} Q^*(s', a) \geq Q^*(s', a') \quad (6)$$

and apply statistical hypothesis-testing to evaluate the relevance of each state parameter. They define the null hypothesis as

$$\bar{Q}(s, a) \geq \bar{Q}(s, a') \quad (7)$$

the test produces p -value $p_{s,a,a'}$ for the value function. The relevance of a state parameter x is evaluated via taking

$$p = \max_a \min_{s'=[s]x} \min_{a' \neq a} p_{s',a,a'} \quad (8)$$

The parameter x is determined to be irrelevant on the projected state s' when the hypothesis is evaluated as true.

The method evaluates the action-action pair on each state individually, and it requires many samples of value functions to perform such statistical test. For example, in the experiment of [7] they ran 25 independent trails of Q-learning to obtain samples of value functions. Furthermore, the relevance of a parameter is represented by the state with minimum p -value among all the projected states. This definition restricts the analysis to the most relevant case and ignores the effects on other states. There are two limitations. The first is that if there are multiple optimal policies, then the statistical test would find possible irrelevance only when the value function is exactly calculated. The second, for a large scale domain, the irrelevance might be rejected by a single rarely visited state even it is applicable on all other states.

The relevance of a state parameter is determined by the policy derived from a learning algorithm. Different problems in the same environment could result in different parameter relevance, and even in the same problem it could find different results from the analysis. The possibility happens when optimum policy is not unique. With a learning algorithm that derives only one optimal policy at a time, it could find different policy at each trial. Then the two results generate ambiguity of state-policy relation. Since the method in [7] takes the minimum p -value to represent the projection result, it often underestimates the irrelevance of a state parameter.

To relieve the limitations and reduce computation cost, we modify their irrelevance analysis method from applying on each state to applying on each projected state. The hypothesis is expressed as

$$\forall_{s' \in [s]_x} \bar{Q}(s', a) \geq \bar{Q}(s', a'). \quad (9)$$

With this hypothesis, the statistical hypothesis test evaluates the relevance of a parameter from all the projected states. The reduction of the sampling time is proportional to $O(n_x)$ where n_x denotes the number of projected states with parameter x .

5. Shared Subtasks Learning Algorithm

In the projected state space, several subtasks would apply a shared subtask. Among these subtasks, although they have the same set of optimum policies, the value functions are still different. This kind of difference within projected value function requires a different model for the shared subtask learning. There are two parts need to be modeled, the shared subtask policy and the expected utility when invoking a

shared subtask. A shared subtask would be invoked as different subtasks and terminates at various states. This property disturbs the learning of shared subtask policy. In this reason, the learning algorithm encapsulates the shared subtask to find optimal policy within. The terminal conditions are assigned with relative pseudo utilities based on the relativity in the original utility. In other words, if the original utility of terminal condition t_1 is greater than another terminal condition t_2 , then we would assign pseudo rewards with t_1 greater than t_2 . The invocation of a shared subtask receives only the rewards of executing the subtask policy, and to learn optimal policy (hierarchical optimality in the notion of hierarchical reinforcement learning), the learning algorithm includes the utility after the subtask terminates. In a policy with a sharable subtask, if a sharable subtask ζ is executed and terminates after N steps, the expected utility can be written as

$$V(s) = E \left[\sum_{t=0}^{N-1} \gamma^t R(s_t) + \sum_{t=N}^T \gamma^t R(s_t) \right] \quad (10)$$

The first term in the expectation function at the right hand side of equation (10) is the cumulative rewards of executing ζ at state s' which terminates after N steps. Assume the termination of ζ results in state s'' , and the second term in the expectation function is the expected utility at state s'' . Let s' denotes the previous state of s'' , the termination state after executing ζ , the transition probability of executing ζ in state s results in state s' is $P(s'|s, \zeta)$, and the transition probability of s' to s'' is $P(s''|s')$. Let V^ζ denotes the value function of ζ , we can rewrite equation (10) as

$$V(s) = \sum_s P(s'|s, \zeta) V^\zeta(s') - V^\zeta(s) + \sum_{s''} P(s''|s') V(s'') \quad (11)$$

The first two terms at the right hand side of the equation are cumulative rewards of executing ζ , while the last term is the cumulative reward after executing ζ till the agent terminates at some goal. The update of value function for invoking a shared subtask ζ is stated as

$$V^i(s) + = \alpha \{ R(\zeta) + \gamma [\gamma^{N+1} V^i(s''') + V^\zeta(s') - \gamma^N V^\zeta(s'')] - V^i(s) \} \quad (12)$$

where γ is the discount ratio, s denotes the state invoking ζ , V^i is the value function of a sub-problem which s is in, V^ζ is the value function of ζ , and s' denotes the start state of ζ , s'' denotes the end state of ζ , s''' denotes the resulted state after ζ terminates. Then

the corresponding action-value function update is expressed as

$$Q^\zeta(s, \zeta) \leftarrow \alpha \{R(s, \zeta) + \gamma [\gamma^{N+1} \max_{a'''} Q^\zeta(s''', a''') \quad (13)$$

$$+ \max_{a'} Q^\zeta(s', a') - \gamma^N \max_{a''} Q^\zeta(s'', a'')] - Q^\zeta(s, \zeta)\}$$

6. Experiments

We set up a capture-the-flag scenario and verified in both single agent and two-agent cases to analyze the performance of cascading decomposition and state abstraction algorithms. The scenario is illustrated using a 10×10 grid world, which is divided into four rooms, and there are doors connecting two neighbored rooms as shown in Figure 1(a). There are two flags A and B in the grid world, one is located at the left bottom corner, and the other is at the right bottom corner. The missions of the agents are to find and capture the flags at different target locations, and to plug these flags at the specific locations. Each agent starts at S(0,0). The flag A at (0,9) should be plugged at the location of A'(4,4), and the flag B at (9,9) should be plugged at the place of B'(5,4). When both flags are plugged at the correct target locations, the agent must go to the final goal G at the right upper corner (9,0), and the whole task is completed. There are six primitive actions for the agents: *west*, *north*, *east*, *south*, *unplug* and *plug*, and each agent must move one grid at a time from one location to the other one. When an agent goes to the grid where a flag is located, it can decide whether to capture the flag or not. Each agent could hold one flag at a time, and when it reaches the specified destination of a flag, it can choose whether to plug the flag or not.

If an agent holds a flag and goes back to the flag's origin location, then the agent can choose to put the flag back to its origin. Either flag could be captured by both agents, but when a flag has been held by an agent, the other agent cannot snatch the flag back from it. The flag can be available only when it is put back to the original location. The transition graph of this scenario is illustrated in Figure 1(b), and the second smallest eigenvector is illustrated in Figure 2.

In the single agent case, performing irrelevant parameter analysis finds shared subtasks. The subtask sharing reduces the complexity of a learning process linearly, and the problem complexity becomes 3/4 of the original problem in the scenario. After the shared subtasks are identified, the decomposed learning framework is applied to learn the action-value function. The overall performance is compared against the method using the spectral clustering and a conventional Q-learning algorithm, and the result is illustrated in Figure 3(a). Since the decomposition result of spectral clustering alone does not provide dimension reduction, the performance of learning based on it is identical to conventional Q-learning. In two-agent environment, irrelevant parameter analysis on each separated component after cascading decomposition finds the parameters of other agent's location that can be reduced. The reduction after subtask sharing is only linear, while the reduction after irrelevant parameters projection is exponential and results in significant performance improvement. In the experiment with the two-agent case, since the other agent is equally competent to accomplish the tasks of the problem, it takes fewer epochs to complete the whole task than it did in the single agent case. The result is illustrated in Figure 3(b).

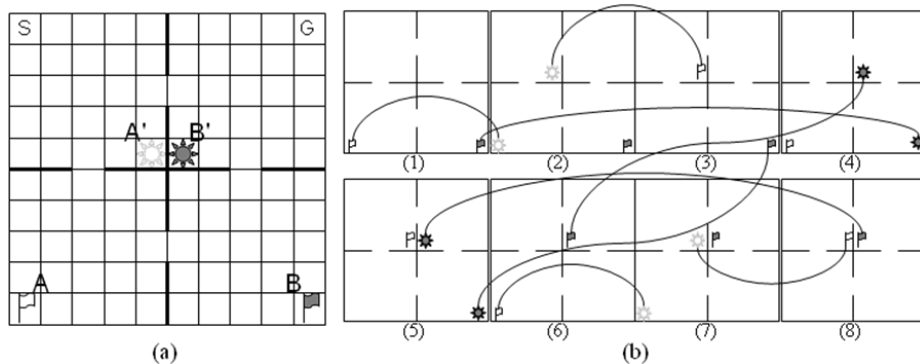


Fig 1. (a) The capture the flag scenario. (b) The transition graph of the environment. The * marks on both the origin and destination of a flag indicate that the flag is held by an agent and can be put onto one of these places.

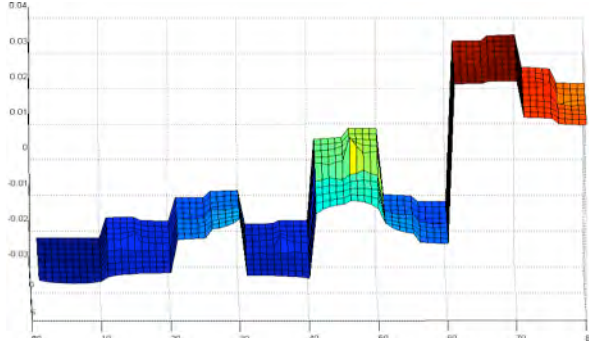


Fig 2. 3D view of the eigenvector

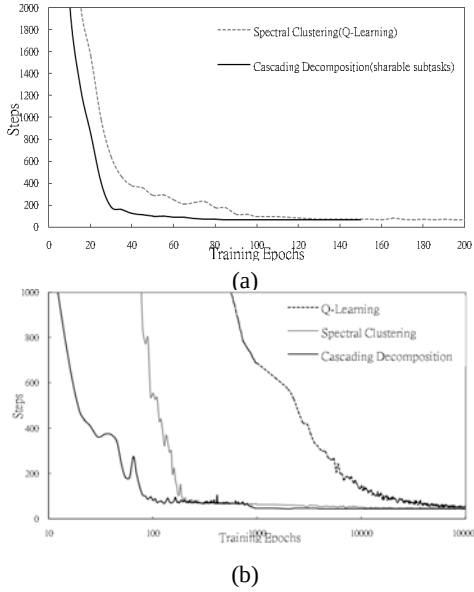


Fig 3. (a) A single agent capture-the-flag scenario. The horizontal axis represents number of training epochs while the vertical axis represents the average steps to goal. (b) The two-agent case.

References

- [1] Ahuja, R. K., Magnanti, T. L., and Orlin, J.B. Network Flows: Theory, Algorithms, and Applications. Prentice Hall (1993)
- [2] Andre, D., and Russell, S. Programmable Reinforcement Learning Agents. In Advances in Neural Information Processing Systems 13, MIT Press (2001)

- [3] Cheeger, J. A lower bound for the smallest eigenvalue of the Laplacian. Problems in analysis, 195-199. Princeton University Press. (1970)
- [4] Chung, F. Spectral Graph Theory. Number 92. CBMS-AMS. (1997)
- [5] Dietterich, T. Hierarchical reinforcement learning with the MAXQ value function decomposition. Journal of Artificial Intelligence Research. **13** (2000) 227-303
- [6] Digney, B. Learning Hierarchical Control Structure for Multiple Tasks and Changing Environments. In Proceedings of the Fifth Conference on the Simulation of Adaptive Behavior. (1998)
- [7] Jong, N. K. and Stone, P. State Abstraction Discovery from Irrelevant State Variables. Proceedings of the 19th International Joint Conference on Artificial Intelligence. (2005)
- [8] Mannor, S., Menache, I., Hoze, A., and Klein, U. Dynamic abstraction in reinforcement learning via clustering. In Proceedings of the Twenty-First International Conference on Machine Learning. ACM Press (2004) 560-567
- [9] McGovern, A., and Barto, A. G. Automatic discovery of subgoals in reinforcement learning using diverse density. In Proceedings of the Eighteenth International Conference on Machine Learning. Morgan Kaufmann (2001) 361-368
- [10] Menache, I., Mannor, S., and Shimkin, N. 2002. Q-Cut - Dynamic discovery of sub-goals in reinforcement learning. In Proceedings of the Thirteenth European Conference on Machine Learning. LNCS 2430. Springer (2002) 295-306
- [11] Parr, R., and Russell, S. Reinforcement Learning with Hierarchies of Machines. In Advances in Neural Information Processing Systems 10, MIT Press (1998)
- [12] Shi, J., and Malik, J. Normalized cuts and image segmentation. IEEE Transactions on Pattern Analysis and Machine Intelligence **22**(8) (2000) 888-905
- [13] Şimşek, Ö., Wolfe, A. P., and Barto, A.G. Identifying Useful Subgoals in Reinforcement Learning by Local Graph Partitioning. In Proceedings of the Twenty-Second International Conference on Machine Learning. ACM Press (2005) 816-823
- [14] Şimşek, Ö., and Barto, A. G. Using relative novelty to identify useful temporal abstractions in reinforcement learning. In Proceedings of the Twenty-First International Conference on Machine Learning. ACM Press (2004) 751-758
- [15] Sutton, R. S., Precup, D., and Singh, S. P. Between MDPs and Semi-MDPs: A framework for temporal abstraction in reinforcement learning. Artificial Intelligence **112**(1) (1999) 181-211

Topic Calculation and Clustering: an Application to Wikipedia

Sara E. Garza

Ramón F. Brena

Eduardo Ramírez

Tecnológico de Monterrey, Centro de Sistemas Inteligentes
Eugenio Garza Sada, 2501 Monterrey, México
{A00592719, ramon.brena, eduardo.ramirez}@itesm.mx

Abstract

Wikipedia is nowadays one of the most valuable information resources; nevertheless, its current structure, which has no formal organization, does not allow to always have a useful browsing among topics. Moreover, even though most Wikipedia pages include a “See Also” section for navigating through those articles’ related Wikipedia pages, the only references included here are those which authors are aware of, leading to incompleteness and other irregularities. In this work, a method for finding related Wikipedia articles is proposed; this method relies on a framework that clusters documents into semantically-calculated topics and selects the closest documents which could enrich the “See Also” section.

1. Introduction

The amount of text and quantity of documents that exist in large or medium-sized corpora can be quite impressive, causing information retrieval tasks, such as search and browsing, to become a real challenge. While the size of the WWW is estimated to be over 27 billion indexed web pages [1], the Wikipedia corpus—which has gained acceptance for its extensive topic coverage and its openness to accept contributions from anyone—currently consists of a collection that surpasses 2.5 million documents in its English version [2].

However, even though Wikipedia contains a vast number of articles where information about a considerable amount of topics can be found, it is also true that navigating in the middle of this broad pool of data can become a difficult endeavor. A specific example of the former can be observed when browsing through an article’s related articles (which are included inside a section called “See also”, “Related Topics”, or “Related to”). On one hand, this section is not mandatory, so it can be the case that certain articles simply do not contain it. On the other hand, and more important, contributors (or a community of them) are not expected to

be aware of all the articles related to theirs, specially in a corpus where content publication is allowed at any time. As a consequence, many pages can miss important links.

All of this arises the issue of providing a better linking schema, and, ultimately, exploring automatic alternatives for a more useful browsing. The latter is being considered because, if the links for an article’s related pages were automatically generated, every article could have a more accurate set of related documents, and this set would be determined by having a *complete* view of the Wikipedia corpus. Moreover, an automatic approach could help reduce human bias while enforcing a semantic-based navigation.

Therefore, an automatic approach is being proposed with the aim of finding and selecting sets of related Wikipedia articles. This approach consists of using “Semantic Contexts”, a quantitative framework for calculating the main topics inside a collection and clustering the documents according to these. If documents are organized into the found topics, then it may be easier to determine which ones are strongly related and suggest an enhanced “See Also” section.

The remainder of the paper is organized as follows: section 2 defines “Semantic Contexts” and related work. Section 3 describes the approach used for finding Wikipedia related pages. Section 4 contains the experiments that were performed for testing the proposed approach. Finally, Section 5 offers conclusions and future work.

2. Background

2.1 Semantic Contexts Framework

“Semantic Contexts” [25] [24] [23] (which will be referred to just as “SC’s” from now on), at a first glance, might be considered as an effort to enforce the so-called Semantic Web [12] by means of a soft-computing initiative [30]. They can be intuitively thought of as *conceptual areas* around which a *family of keywords* (document terms) appears frequently inside the text of web pages. For example, a context referring to palm plants may contain terms

such as “leaf”, “tree”, and “beach”.

In order to describe how SC’s are obtained from a corpus, the notion of “k-cores” has to be introduced. A k-core is a group of k words that *best* represents a topic; i.e., the “OO programming” could be described by the {object, oriented, programming, encapsulation, polymorphism} 5-core. The notion of “best” implies that a force equation, which contrasts the cohesion of each candidate core as a whole (co-occurrence of terms) against the individual popularity of each word, has to be maximized. For achieving this, the ratio between the *joint document frequency* of the core words and the sum of the *disjoint frequencies* of each isolated core term is obtained. Equation 1 shows this definition:

$$f(\{w_1, \dots, w_k\}) = c \frac{JF(\{w_1, \dots, w_k\})}{DF(\{w_1, \dots, w_k\})} \quad (1)$$

where $f(\{w_1, \dots, w_k\})$ represents the force of a set of k keywords, c is a suitable constant (like 10^{12}), $JF(\{w_1, \dots, w_k\})$ (“joint frequency”) is the number of pages that contain all keywords from w_1 to w_k , and $DF(\{w_1, \dots, w_k\})$ (“disjoint frequency”) is the number of pages that happen to have one of the terms ranging from w_1 to w_k but not all of them at once.

To find the k-cores within a corpus, an n-round hill-climbing algorithm, which takes as candidate cores all the groups of k words (one group per document) with highest tf-idf [3] values, is run. After successive word replacements, the candidate cores “evolve” into k-cores by keeping the groups that lead to a peak force. [25] offers a more thorough explanation of the algorithm, the parameters that have been found to work better, and the characteristics of the k-cores that finally emerge.

The found k-cores of the corpus are to be used as conjunctive queries over the collection. This is done with the purpose of constituting groups of documents that contain all the k words of the core, and that, thus, belong to the topic represented by it (note that this implies *overlapping* clusters, since a document can pertain to more than one topic). Consequently, the aforementioned groups build the conceptual areas that intuitively define an SC. However, depending on the application, different formal definitions can be accommodated to describe an SC. Regarding browsing and finding semantically close documents purposes, an SC might be described as a tuple $S = (K, D)$, where:

- K is a set that represents the k-core
- D is a group of document vectors of the form $\vec{d}_j = \langle w_{1j}, w_{2j}, \dots, w_{Nj} \rangle$, where:
 - N represents the length of the corpus vocabulary
 - $w_{ij} \in [0, 1]$

- $\forall w_{ij}, t_i \quad t_i \in K \Rightarrow w_{ij} > 0$, meaning that every core word must be present in the document.

2.2 Related Work

Regarding works that seek to create semantic tools for Wikipedia, in [13] research was focused on clustering Wikipedia music related articles by means of a Self-Organized Map (SOM). Fragments of text and the titles of each article were used, as well as text belonging to hyperlinks included in these articles. The final result consisted of a topological map where similar documents were grouped together, according to discovered categories. A similar work, but also devoted to the area of automatic linking, has been done in [19]. The main goal was to discover missing hyperlinks inside Wikipedia, and this was accomplished by performing two tasks: first, clustering pages with related content (including titles and links) and co-citations [27], and second, obtaining from these clusters the most related articles, for finally determining which links should be inserted in the queried page. Another approach can be found in [26], which tries to identify document topics in order to aid clustering and classification. This is done by specifically using Wikipedia’s category network and document titles.

An effort for constituting a semantic Wikipedia, although with a “hard”, ontologic approach, can be found in [28]; the inclusion of typed links, attributes, and the specification of typed data inside the Wikipedia corpus is described. This was done by using RDF and RDFS. On the other hand, [17] analyzes the current English Wikipedia data for identifying its semantic structure and presenting a pertinent visualization. The analyzed data includes existing categories and authors’ production. Other works in this area include [21], which uses Wikipedia as a *source* for semantic information.

Concerning document clustering works applied to Information Retrieval, [15] uses a dynamic, iterative approach for browsing; this method “scatters” the document collection into small disjoint clusters for the user to pick, and “gathers” the selected groups into a sub-collection of topics, which are presented again. Furthermore, [10] combines NLP and a document clustering algorithm based on Principal Component Analysis and k-means to build a retrieval system where information can be searched and browsed within “contexts”.

With respect to works dealing only with semantic hyperlinks and automatic hyperlinking, on one hand, [14] presents an approach that intends to create semantic links through the use of clustering (which is done with the Wordnet English lexicon) and Latent Semantic Linking. On the other hand, [29], does automatic linking of plain text Chinese news by utilizing Self-Organized Maps in order to create groups of words and groups of documents that allow to

establish different relationships among the existing documents of the corpus; intra-cluster, inter-cluster, and aggregate links are finally constructed.

3 Method

As it was stated earlier, since Semantic Contexts by nature group documents that are similar, it is possible to apply them for finding the related articles for a Wikipedia page. This requires two processes, which intuitively consist of 1) clustering similar pages into topics and 2) providing the most related pages out of the clusters that the page belongs to. Therefore, first it is necessary to do Semantic Context extraction and then related document selection. The extraction process is done offline, while selection can be done online.

3.1 Semantic Context Extraction

This task basically consists of creating the document clusters that will serve for defining which sets of documents are within the same topic—and thus, related to each other. For achieving this purpose, two sub-tasks have to be carried out: a) topic calculation and b) cluster creation. The former consists of finding the corpus k -cores, and is the most laborious step; afterwards, the latter is a straight-forward procedure.

In addition to running the hill-climbing algorithm that is currently used for finding the k -cores, pre-processing [9] and post-processing operations have to be done to seek that the resulting cores will be meaningful. Pre-processing steps include pruning common Wikipedia words and phrases (i.e. “The free encyclopedia”), whereas in post-processing 0-force cores are deleted, as well as cores that show not to have maximum force.

Semantic Contexts are created by using each core as a conjunctive query over the corpus, as it was already stated. The information that relates documents to cores is to be stored in a database for providing a faster access.

If Semantic Context extraction were to be seen as a clustering task, its components (according to [20]) could be summarized as follows:

- **Objects.-** The objects to be clustered are *documents*, whose feature vectors are *tf-idf weights* (quantitative components).
- **Principle for grouping.-** Objects are grouped according to a conceptual principle, which is conformed by k selectors that are taken in conjunction; consequently, objects complying with all the selectors belong to the same group. In this case, as it was already stated, every weight w_{ij} corresponding to a core term t_i must be greater than 0.
- **Type of cluster structure.-** *Overlapping* clusters are produced. Intra-cluster similarity can be measured by calculating cosine distance [11] between pairs of document vectors. Inter-cluster similarity can be measured by either calculating set similarity (Jaccard’s coefficient [4]) between the SCs’ documents or cosine distance between SC vectors.
- **Cluster representation.-** Clusters are represented as Semantic Contexts of the form $S = (K, D)$.

3.2 Related Document Selection

With respect to the process of selecting the related documents of an article, it mainly consists of choosing and presenting the documents (which we will refer to also as *target documents*) that are within the same contexts that the article (which we will address as the *source* document) belongs to. As a consequence, the first step is to retrieve the contexts that the source corresponds to; moreover, besides retrieving them, it is necessary to select the most related ones (specially if the source document belongs to several contexts). In order to achieve this goal, the contexts are ranked according to a score which is assigned by summing up the amount of occurrences of the context’s core words inside the source. For example, if the document has *word*₁ 4 times, *word*₂ 3 times, and *word*₃ 6 times, then the context with core $\{word_1, word_2, word_3\}$ will be given a score of 13. Once sorted according to this criterion, the first n (i.e., the two with highest score) are to be selected. The documents belonging to these SC’s are retrieved, and their degree of similarity with regard to the source is computed for each one (in our case, this is done by obtaining the cosine distance between the source and each target document vector). Those targets with a similarity greater than a certain threshold are included in the related documents set. Finally, the targets are ranked according to the cosine previously obtained, and are presented as the related articles. It is important to note that the purpose of ranking and selecting is to avoid presenting an extensive list of results that can be overwhelming and lead to user disorientation [7] (ultimately, a document may be related to almost every other document inside a given collection—for instance, by tangent links [8]—, but the aim is to show to the user the *most related* ones).

4 Experiments

The goal of the validation experiments we present below is to evaluate the performance of the related-pages generation method and to compare it against human-made related pages; we also take a random page selection as a baseline reference point. In addition, we take an independent measure, completely unrelated to our method, in order to avoid

an unwanted bias. This measure, as explained below, is drawn from a hierarchical topic structure taken from Wikipedia’s List of Academic Disciplines; pages distances are defined as the number of links needed to go from a point to another one in the hierarchy.

4.1 Setup

Corpus. The corpus was conformed by 1429 documents, which were extracted (title and text) from the January 2008 XML dump file of Wikipedia articles (which has a cleaner text than the HTML version but still contains special markup [5] to be dealt with). The selected articles belong to the Academic Areas [6] that are established on the Wikipedia Contents section. The articles were indexed using Lucene Wikipedia indexer [18].

SC extraction. We considered cores composed of 4 words; this value was chosen because, empirically, it has shown to return good results [25]. Regarding the hill-climbing algorithm, it was set to run with 2 rounds (as a consequence, for the last round, the cores found were introduced as candidates). Stemming was made with Snowball [22].

Document and core selection. Considering that the most relevant related documents are probably found within the first pair of a document’s k-cores, a value of $n = 2$ (where n is the number of cores chosen) was set. The threshold for selecting documents within the k-cores was set to 0.1—that is, documents showing to have a degree of similarity greater than 0.1 were considered into the final list of related articles. To be able to make a fair comparison with other sets of related pages (these sets are described in the next section), only the top five results were considered for each article.

Test set. 50 source documents were randomly chosen from the corpus for retrieving their related pages.

4.2 Result Evaluation

In order to know if the obtained related documents are in fact associated to the test set’s source articles, the results were compared against two approaches: a) the *random* approach and b) the *human* approach. For the random approach, a set of random documents was selected. With respect to the human approach, a set of documents regarded by human authors as related was chosen (these can be found on the “See also” section). These two comparisons define a lower and an upper bound, respectively; as a consequence, it is expected that the generated results surpass the random approach and show to be competitive with the human approach.

For establishing a comparison among the different approaches (random, human, and automatic), it was necessary

to use a distance metric that could tell—by means of a numerical value—which technique shows the “closest” documents given a source article. Now, due to the fact that Wikipedia’s List of Academic Disciplines is arranged into a hierarchy, it was possible to define this metric in terms of the number of links or “hops” needed to get from one page to another. And, in addition to solely using this number, a weighted scheme had to be considered for providing a fairer measurement. That way, pages with short paths that showed to be not very related could be awarded a greater distance than those that showed to be semantically closer but had longer paths because they fell into deeper levels. For achieving this, a “discounting factor” was to be used when going down the hierarchy; this factor takes half the distance of the previous level and is illustrated on Table 1. An example of how this distance would be measured is shown in Table 2. Moreover, it is important to highlight the various reasons behind the motivation to use this evaluation method: 1) this measurement is *independent* with respect to the tested approaches, and thus provides *objectivity*, 2) since it is independent to the automatic approach, the former is not subject to any element of the evaluation method either—i.e., the discipline hierarchy, 3) because the method is *tailored* for the studied corpus, it can be considered as complementary with regard to standard measures (precision, recall, F_1), and 4) the documents belonging to the hierarchy have not been assessed as relevant by judges under an artificial setup, but instead by a community of users who have created and edited this page of academic disciplines according to their actual needs.

Having established the distance metric, as well as the test sets, all distances from the test source articles to the presumed related pages were computed. After this was done, the distances belonging to the same source were summed up and normalized by dividing this sum by the maximum possible distance from that particular node to any other in the hierarchy. Finally, the normalized results for each test page were averaged. The final average obtained by the SC’s method was compared to the one obtained with the random approach and the human approach.

4.3 Results

Document clustering. A total of 633 4-cores, out of more than 750 that were obtained before refinement, were found inside the corpus. This achieved a total coverage of 77%—which indicates that most documents fell into an SC. With respect to the resulting cluster structure, more than 20% of the contexts showed to be overlapping, less than 1% showed to subsume or be subsumed by other contexts, and the rest

Table 1. Score according to hierarchy levels.

$$\text{Level 0} \xrightarrow{1} \text{Level 1} \xrightarrow{0.5} \text{Level 2} \xrightarrow{0.25} \text{Level 3} \xrightarrow{0.125} \text{Level 4} \xrightarrow{0.0625} \text{Level 5} \xrightarrow{0.03125} \text{Level 6}$$

Table 2. “Hop” distance from *Machine Learning* to *Functional programming*

$$\begin{aligned} &\text{Areas} \xrightarrow{1} \text{Comp. Science} \xrightarrow{0.5} \text{AI} \xrightarrow{0.25} \text{Cognitive Science} \xrightarrow{0.125} \text{Machine Learning} \\ &\text{Areas} \xrightarrow{1} \text{Comp. Science} \xrightarrow{0.5} \text{Prog. Languages} \xrightarrow{0.25} \text{Functional Programming} \\ \\ &d(\text{Machine Learning, Functional Prog.}) = d(\text{Machine Learning, Comp. Sciences}) + \\ &\quad d(\text{Comp. Sciences, Functional Prog.}) \\ \\ &= d(\text{Machine Learning, Cognitive Sciences}) + \\ &\quad d(\text{Cognitive Sciences, AI}) + \\ &\quad d(\text{AI, Computer Sciences}) + \\ &\quad d(\text{Computer Sciences, Prog. Languages}) + \\ &\quad d(\text{Prog. Languages, Functional Prog.}) \\ \\ &= 0.125 + 0.25 + 0.5 + 0.5 + 0.25 \\ &= \mathbf{1.625} \end{aligned}$$

conformed disjoint clusters; these data were acquired by analyzing the database where the k-cores and other SC information were stored. Finally, concerning the clusters that were formed, each one contained an average of ten documents. Even though this can be considered as a relatively small number of elements, for our purpose it was a fair amount.

Related pages. These are the main results and ultimately the presented validation. Recalling that distances near 0 are smaller, the random approach obtained an average distance of 0.846, while the automatic approach yielded a weighted average distance of 0.252. Finally, the human approach had an average of 0.275, that we can see is not as good as the machine one.

4.4 Result Discussion

With respect to the related pages results, it can be stated that these complied with what was expected. On one hand, they widely surpassed the random approach, and, on the other hand, they showed to be at least as good as the human approach. And, even though more extensive tests could be more conclusive, these results enable to establish that an automatic approach for obtaining Wikipedia related articles via Semantic Contexts is possible, and might produce an acceptable quality.

Regarding SC extraction, after the cores were obtained

from the corpus, an analysis was performed with the intent of finding a structure—if any—that they could fall into. It was discovered that, at least for the academic disciplines section of Wikipedia, the cores were arranged in a plain hierarchy; that is, the majority of them were near to be mutually exclusive (thus each containing a different fragment of the corpus documents), and just a few were shown to subsume others. Furthermore, since the created document clusters were very small, it could be stated that the corpus was highly fragmented. Probably this result was yielded because every Wikipedia article can be considered as a topic by itself; as a consequence, clustering with SC’s in this case became a “meta-topic” or “super-topic” search.

5 Conclusions and Future Work

An automatic approach for finding related pages within Wikipedia articles has been presented. This approach consists of two main processes: Semantic Context extraction and related document selection. On one side, Semantic Context extraction serves the purpose of creating document clusters, which are gathered by semantically discovering the existing topics inside the corpus; every topic is represented by a 4-core (the four most important words regarding the subject). Related document selection, on the other hand, consists of retrieving and ranking the most relevant clusters

(SC's) for a given document and presenting the documents inside these clusters that appear to be the most similar.

The aforementioned approach has been tested on documents belonging to Wikipedia's List of Academic Disciplines; the obtained results show that the approach—as expected—is by far superior than selecting random pages, and that is also competitive when compared to human hand-made lists. Despite the small test set and selected portion of the corpus, the obtained outcome enables to conclude that this approach deserves to be further explored. Additionally, these results may lead to consider using this clustering process as a “shortcut” to finding the most similar documents instead of brute-force alternatives, like calculating cosine similarity for every pair of documents and ranking the results according to the source.

Future works include refining the method (i.e., final document ranking and selection criteria), experimenting with different parameters, and including a larger portion of Wikipedia articles. Also, regarding this last point, an alternative evaluation method is being considered; a possibility would be to use link clustering and graph-based methods [16], and validate results against this schema. On the other hand, if the application concerns browsing, it would also be desirable for the current prototype to reflect this task.

References

- [1] <http://www.worldwidewebsite.com/>.
- [2] <http://en.wikipedia.org/wiki/Wikipedia>.
- [3] <http://en.wikipedia.org/wiki/Tf-idf>.
- [4] http://en.wikipedia.org/wiki/Jaccard_index.
- [5] <http://en.wikipedia.org/wiki/Wikipedia:Cheatsheet>.
- [6] http://en.wikipedia.org/wiki/List_of_Academic_Disciplines.
- [7] M. Agosti, M. Melucci, and F. Crestani. Automatic authoring and construction of hypermedia for information retrieval. *Multimedia Systems*, 3(1):15–24, 1995.
- [8] J. Allan. Automatic hypertext link typing. *Proceedings of the the seventh ACM conference on Hypertext*, pages 42–52, 1996.
- [9] N. Andrews and E. Fox. Recent developments in document clustering. Technical report, Virginia Tech, 2007.
- [10] P. Anick and S. Vaithyanathan. Exploiting clustering and phrases for context-based information retrieval. *ACM SIGIR Forum*, 31:314–323, 1997.
- [11] R. Baeza-Yates. *Modern Information Retrieval*. Addison-Wesley, New York, 1999.
- [12] T. Berners-Lee, J. Hendler, and O. Lassila. The semantic Web. *Scientific American*, 284(5):28–37, 2001.
- [13] S. Bloehdorn and S. Blohm. A self organizing map for relation extraction from wikipedia using structured data representations. *Proceedings of the International Workshop on Intelligent Information Access (IIA-2006)*, 2006.
- [14] J. Camacho-Guerrero, A. Carvalho, M. Pimentel, E. Munson, and A. Macedo. Clustering as an approach to support the automatic definition of semantic hyperlinks. *Proceedings of the 18th conference on Hypertext and hypermedia*, pages 81–84, 2007.
- [15] D. Cutting, D. Karger, J. Pedersen, and J. Tukey. Scatter/Gather: a cluster-based approach to browsing large document collections. *Proceedings of the 15th annual international ACM SIGIR conference on Research and development in information retrieval*, pages 318–329, 1992.
- [16] L. Getoor and C. Diehl. Link mining: a survey. *ACM SIGKDD Explorations Newsletter*, 7(2):3–12, 2005.
- [17] T. Holloway, M. Bozicevic, and K. Börner. Analyzing and Visualizing the Semantic Coverage of Wikipedia and Its Authors. *ArXiv Computer Science e-prints, cs/0512085*, 2005.
- [18] L. W. indexer. <http://schmidt.devlib.org/software/lucene-wikipedia.html>.
- [19] M. Krotzsch, D. Vrandecic, and M. Volkel. Wikipedia and the Semantic Web-The Missing Links. *Proceedings of Wikimania*, 2005, 2005.
- [20] R. Michalski and R. Stepp. Automated Construction of Classifications: Conceptual Clustering versus Numerical Taxonomy. *IEEE Trans. on Pattern Analysis and Machine Intelligence*, Vol. PAMI-5, pages 396–410, 1983.
- [21] R. Mihalcea and A. Csomai. Wikify!: linking documents to encyclopedic knowledge. In *CIKM '07: Proceedings of the sixteenth ACM conference on Conference on information and knowledge management*, pages 233–242, New York, NY, USA, 2007. ACM.
- [22] M. Porter. An algorithm for suffix stripping. *Program*, 14(3):130–137, 1980.
- [23] E. H. Ramirez and R. F. Brena. Semantic contexts in the internet. In *LA-WEB '06: Proceedings of the Fourth Latin American Web Congress (LA-WEB'06)*, pages 74–81, Washington, DC, USA, 2006. IEEE Computer Society.
- [24] E. H. Ramirez and R. F. Brena. A soft semantic web. In *IEEE Workshop on Hot Topics in Web Systems and Technologies (HOTWEB 06)*, 2006.
- [25] E. H. Ramirez and R. F. Brena. Semantically mapping the web. In *8th Conference on Computing, CORE 2007*, 2007.
- [26] P. Shönhofen. Identifying document topics using the wikipedia category network. In *Web Intelligence Proceedings of the 2006 IEEE/WIC/ACM International Conference on Web Intelligence*, pages 456–462. IEEE Computer Society, 2006.
- [27] H. Small. Co-citation in the scientific literature: A new measure of the relationship between two documents. *Journal of the American Society for Information Science*, 24(4):265–269, 1973.
- [28] M. Völkel, M. Krötzsch, D. Vrandecic, H. Haller, and R. Studer. Semantic Wikipedia. *Proceedings of the 15th international conference on World Wide Web*, pages 585–594, 2006.
- [29] H. Yang and C. Lee. A text mining approach for automatic construction of hypertexts. *Expert Systems With Applications*, 29(4):723–734, 2005.
- [30] L. A. Zadeh. Soft computing and fuzzy logic. *Software, IEEE*, 1994.

Using linguistic information to classify Portuguese text documents

Teresa Gonçalves
Departamento de Informática
Universidade de Évora
7000-671 Évora, Portugal
tcg@di.uevora.pt

Paulo Quaresma
Departamento de Informática
Universidade de Évora
7000-671 Évora, Portugal
pq@di.uevora.pt

Abstract

This paper examines the role of various linguistic structures on text classification applying the study to the Portuguese language. Besides using a bag-of-words representation where we evaluate different measures and use linguistic knowledge for term selection, we do several experiments using syntactic information representing documents as strings of words and strings of syntactic parse trees.

To build the classifier we use the Support Vector Machine (SVM) algorithm which is known to produce good results on text classification tasks and apply the study to a dataset of articles from the Público newspaper. The results show that sentences' syntactic structure is not useful for text classification (as initially expected), but part-of-speech information can be used as a term selection technique to construct the bag-of-words representation of documents.

1. Introduction

Current Information Technologies and Web-based services need to manage, select and filter increasing amounts of textual information. Text classification allows users, through navigation on class hierarchies, to browse more easily the texts of their interests. This paradigm is very effective both in filtering information as in the development of online end-user services.

As the number of documents involved in these applications is large, efficient and automatic approaches are necessary for classification. Standard Machine Learning approaches use the bag-of-words representation to deceive the classification target function, where the only features are document word statistics. Typical linguistic structures such as morphology and syntax are usually neglected in the learning process. Moreover, almost all studies have been conducted on texts written in the English language.

In order to evaluate common Information Retrieval representation techniques (that possibly use some morphologi-

cal information) for the Portuguese language, a set of initial experiments was made. The most effective document representation served as a starting point for the other analyses.

Then, the use of morphosyntactic and syntactic information was appreciated. While using morphosyntactic information as a term selecting technique, a structured representation based on parse trees was used to represent the syntactic information. To build the learners we used the Support Vector Machine (SVM) algorithm since it can support structured representations and is known to produce good results on text classification tasks.

This paper is organized as follows: Section 2 presents concepts and tools related to linguistic information and Machine Learning, Section 3 describes the used document representation and Section 4 presents and Section 5 evaluates the experiments. Conclusions and future work are pointed out in Section 6.

2. Linguistic Information and Machine Learning

This section introduces the used linguistic concepts and tools and the chosen Machine Learning algorithm and software.

2.1. Linguistic information

The Portuguese language is morphological rich: while nouns and adjectives have 4 forms (two *genders* – masculine and feminine and two *numbers* – singular and plural), a regular verb has 66 different forms (two *numbers*, three *persons* – 1st, 2nd and 3rd and five *modes* – indicative, conjunctive, conditional, imperative and infinitive, each with different number of *tenses* ranging from 1 to 5).

2.1.1 Representation

Morphological information includes word stem and its morphological features, like grammatical class and flexion.

While some natural language processing tasks use word stem, others use its lemma.

Most syntactic language representations are based on the context-free grammar (CFG) formalism introduced by [4] and, independently, by [1]: given a sentence, it generates the corresponding syntactic structure usually represented through a tree structure known as sentence *parse tree*. It contains its constituents structure (such as noun and verb phrases) and words' grammatical class.

2.1.2 Tools

We applied a Portuguese stop-list (set of non-relevant words such as articles, pronouns, adverbs and prepositions) and POLARIS, a lexical database [9], to generate the lemma for each Portuguese word.

PALAVRAS [2] parser was used to obtain words' POS tags and sentences' parse tree. It was developed in the context of the VISL project by the Institute of Language and Communication of the University of Southern Denmark. This parser is robust enough to always produce an output even for incomplete or incorrect sentences and has a comparatively low percentage of errors (less than 1% for word class and 3-4% for surface syntax) [3].

Its output is the syntactic analysis of each phrase including the POS tag associated with each word. Possible POS tags and are: adjective (adj), adverb (adv), article (det), conjunction (conj), interjection (in), noun (n), numeral (num), preposition (prp), pronoun (pron), proper noun (prop) and verb (v).

2.2. Learning Algorithm

To build the learners we used the Support Vector Machine (SVM) algorithm (with different kernel functions) since it can support structured representations and is known to produce good results on text classification tasks.

2.2.1 Support Vector Machine

Support Vector Machine (SVM) is a learning algorithm introduced by Vapnik and coworkers [16], which was motivated by the theoretical results from the statistical learning theory. It joins a kernel technique with the structural risk minimization framework. A *kernel technique* comprises two parts: a module that performs a mapping into a suitable feature space and a learning algorithm designed to discover linear patterns in that space.

The *kernel function* (or simply the kernel), that implicitly performs the mapping, depends on the specific type and domain knowledge of the data source. The *learning algorithm* is general purpose and robust; it's also efficient, since the amount of computational resources required is polynomial

with the size and number of data items, even when the dimension of the embedding space grows exponentially [15]. Its key aspects can be highlighted as follows (see Figure 1):

- Data items are embedded into a vector space called the feature space.
- Linear relations are discovered among images of data items in feature space.
- Algorithm is implemented in a way that the coordinates of the embedded points are not needed; only their pairwise inner products.
- Pairwise inner products can be computed efficiently directly from the original data using the kernel function.

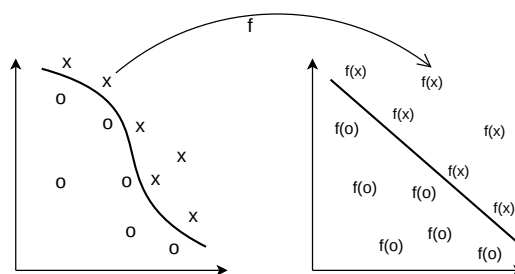


Figure 1. Kernel function: data nonlinear pattern transformed into linear feature space.

The *structural risk minimization* framework creates a model with a minimized VC (Vapnik-Chervonenkis) dimension. This developed theory [17] shows that when a model VC dimension is low, the expected probability of error is also low, which means good performance on unseen data.

2.2.2 Kernel functions

Most approaches to text classification use the basic vector space model (VSM) to represent documents. The simplest measure that takes into account word frequency in each document can be naturally reinterpreted as a kernel method. Normalization and term reduction approaches can also be interpreted as kernel functions (see [15]) and other standard kernels (like the polynomial one) apply non linear transformations to the usual VSM approach.

The *convolution kernel* [7] is the most well-known kernel for structured objects. A structured object is an object formed by the composition of simpler components; frequently, these components are, recursively, simpler objects of the same type. It's the case of strings, trees or graphs. The convolution kernel definition is based on kernels defined over structure components.

For tree structured objects, the feature space is indexed by subtrees and similarity is based on counting common subtrees. The subset tree kernel [5] is one of such kernels that uses ordered labelled trees counting subsets of common trees between two trees. They have produced good results on parse tree ranking [5] and predicate argument classification [11], [20].

2.2.3 Software

For the morphological and morphosyntactic levels we used WEKA SVM algorithm. WEKA [19] is a software package developed in New Zealand Waikato University implementing a large collection of ML algorithms.

For the syntactic level, the SVM algorithm was run using SVM^{light} -TK [10]. This software is an extension to SVM^{light} [8], that uses convolution kernels to represent tree structures. It implements two different kernels for tree structures: the subtree kernel [18] and the subset tree kernel [5]. Intuitively, the first counts all common n -descendants until the leaves (being n the root node) and the second adds to that counting all trees considering as leaves all internal tree nodes.

3. Document representation

Most text classification research is applied to English written documents and even if there are some that work on the sub-word or multi-word levels, the most used indexing term is, without doubt, the word. Next, we present the document representations used in this work.

3.1. Morphological information

As a baseline for this study, we first considered the traditional bag-of-words representation using the word and its lemma as indexing terms. We considered several filtering and weighting measures. We also used word grammatical class (morphosyntactic information) as a term selecting method. Figure 2 illustrates a two sentence document (“Mother observes her daughter Carlota. Carlota plays with the doll.”) and the corresponding representation.

3.2. Syntactic information

Since a parse tree is an ordered tree, each document, that is a sequence of sentences, can be represented as an ordered tree of ordered trees. In this way, a document is a tree where each root child is the parse tree of a sentence and the leaves are its word lemma. This representation was named *syntactic-tree* representation.

However, we did not use the complete parse tree, but only the nodes of the following word classes: noun, proper

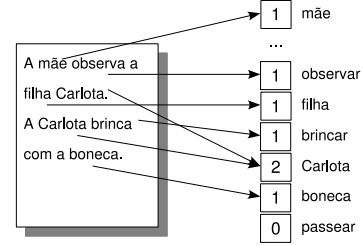


Figure 2. The original document and the corresponding bag-of-words representation.

noun, adjective, verb, pronoun, and adverb. Figure 3 illustrates a two sentence document (“Mother observes her daughter Carlota. Carlota plays with the doll.”) and the corresponding representation.

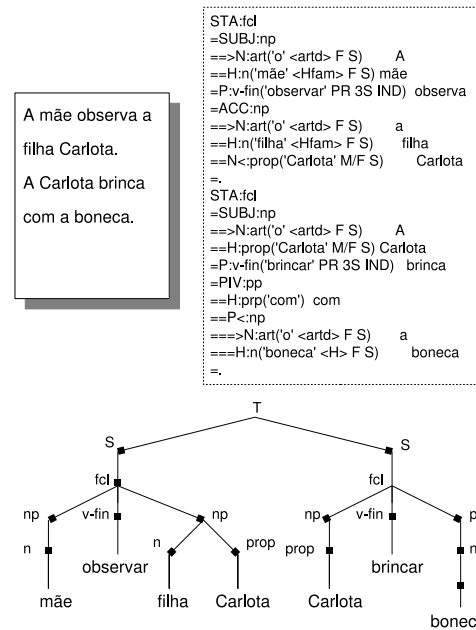


Figure 3. Original document, PALAVRAS output and syntactic-tree representation.

4. Experiments

This section introduces the used dataset, describes the experimental setup and presents the obtained results.

4.1. Dataset description

Publico corpus contains the Público newspaper daily news (from the years of 1994 and 1995) taken from 9 dif-

ferent sections (used as semantic classes). It totals 101646 documents, where there are 282657 distinct words, and, on average, 512 running words (tokens) and 254 unique words (types) per document.

For the syntactic experiments, a subset of **Publico** corpus with the October 1995 news was used. **Pub9510** has 4290 documents, with 70743 distinct words, and, on average, 215 tokens and 124 types per document. Table 1 shows the semantic classes and proportion of documents for each dataset.

	Publico	Pub9510
section	doc %	doc %
ciências, tecnologia e educação (science, technology, education)	6.2	6.7
cultura (culture)	15.5	14.5
desporto (sports)	9.9	10.3
diversos (diverse)	8.2	8.1
economia (economy)	13.3	10.5
local (local)	17.2	21.3
mundo (world)	9.4	9.3
nacional (national)	9.2	10.3
sociedade (society)	11.2	9.1

Table 1. Publico and Pub9510 corpora: classes and proportion of documents.

4.2. Experimental setup

Bag-of-words representations used a linear kernel while the syntactic-tree one was run with the subset tree kernel. WEKA was run with default parameters (normalized training data and $c=1$, the trade-off between training error and margin) and SVM^{light} -TK with $L=0.001$ (decay factor) and $c=10$. A train-and-test procedure was applied with 33% of documents used for testing.

Learner performance was analyzed through precision (π), recall (ρ) and F_1 (f_1) measures [13] of each category (obtained from classification contingency table: prediction vs. manual classification). For each one, we calculated the micro- (μ) and macro-averages (M) and made significance tests regarding a 95% confidence level.

4.3. Morphological information

First we considered morphological information with the traditional bag-of-words representation. It's the typical representation used in Information Retrieval techniques, it serves as baseline experiment and allows to verify lemmatization role in the classification process.

We made experiments using original words and their lemma (lm). Word selection was made using three classes

of experiments: stop-word elimination¹ (st), filtering function (word frequency: fr and mutual information: mi) and threshold value (t). To weight the selected terms we used the three usual components: document (term frequency: t), collection (without component: x , and inverse term frequency: f) and normalization (co-sin: c). All these options can be graphically represented in a 3-dimensional space with normalization, selection and weighting axes. In turn, selection and weighting techniques can also be represented in other three-dimensional spaces (The marked point on Figure 4 corresponds to stop-word elimination, using lemmas as indexing terms, mutual information as filtering function, threshold value equals to one and $tfidf$ as weighting technique).

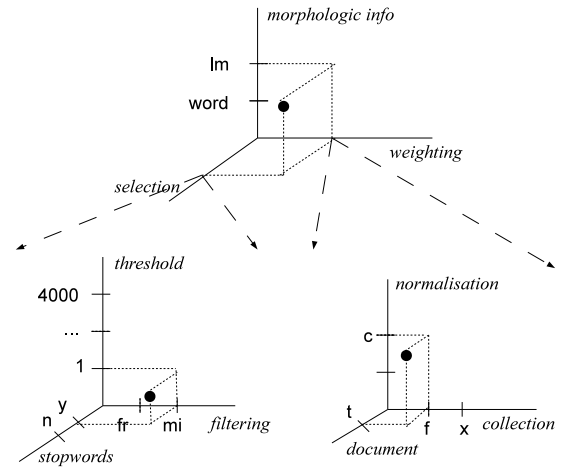


Figure 4. Graphical 3D representation of the experiments.

4.3.1 Results

We made experiments with combinations of the options described above (for **lemma** experiments we eliminated stop-words) with threshold values ranging from 1 to 4000 (this value indicates the smallest frequency above which the term is selected), in a total of 88 different runs. Table 2 shows the minimum, maximum, average and standard deviation values for the micro- and macro averages of the three performance measures.

To choose a representing experiment, we searched, for each performance measure, the ones with performance values having no significant difference with the maximum. There were 5 experiments with all 6 performance measures in the best set:

- word lemmatization (lm), with threshold value 1 (t_1);

¹The stop-list was composed of 207 words.

	π^μ	ρ^μ	f_1^μ	π^M	ρ^M	f_1^M
min	.787	.787	.787	.779	.770	.774
max	.843	.843	.843	.842	.831	.836
avg	.824	.824	.824	.821	.810	.815
std	.012	.012	.012	.013	.013	.013

Table 2. Publico minimum, maximum, average and standard deviation values for micro- and macro averages.

- stop-word elimination (st), term frequency filtering function (fr) and threshold value 50 (t_{50});
- stop-word elimination, mutual information filtering function (im), tfidf weighting technique (tfc) and threshold value 50 (t_{50}).

Table 3 shows their values (boldface values have no significant difference). From this set we chose **lm-fr-tfc- t_1** as the representative experiment. It counts 205155 distinct words with averages of 288 tokens and 189 types per document.

	π^μ	ρ^μ	f_1^μ	π^M	ρ^M	f_1^M
lm-fr-txc- t_1	.840	.840	.840	.839	.828	.833
lm-fr-tfc- t_1	.843	.843	.843	.842	.831	.836
st-fr-txc- t_{50}	.839	.839	.839	.837	.826	.831
st-fr-tfc- t_{50}	.840	.840	.840	.838	.828	.832
st-im-tfc- t_{50}	.840	.840	.840	.839	.828	.833

Table 3. Publico performance values for the morphological “best” experiments.

4.4. Morphosyntactic information

To access the discriminative power of morphosyntactic information over classes we made a set of experiments including the word classes considered more informative – name (n), proper name (prop), adjective (adj) and verb (v).

4.4.1 Results

Using the same setting as the morphological information selected experiment we tried 15 different word class combinations. Table 4 shows the performance values for these experiments. Values of the morphological experiment, **lm-fr-tfc- t_1** , are also shown. From all experiments, only the one that combines all word classes (n+prop+adj+v) has equivalent values to the morphological one. This experiment counts 201327 distinct words with 250 tokens and 169 types per document.

We also made some experiments using higher thresholds, but performance values were lower.

	π^μ	ρ^μ	f_1^μ	π^M	ρ^M	f_1^M
lm-fr-tfc- t_1	.843	.843	.843	.842	.831	.836
adj	.683	.683	.683	.671	.663	.666
n	.808	.808	.808	.806	.704	.706
prop	.784	.784	.784	.778	.766	.771
v	.635	.635	.635	.623	.614	.618
n+adj	.807	.807	.807	.803	.792	.796
n+prop	.829	.829	.829	.824	.814	.818
n+v	.803	.803	.803	.821	.789	.802
prop+adj	.804	.804	.804	.800	.788	.793
adj+v	.732	.732	.732	.726	.715	.720
prop+v	.805	.805	.805	.802	.789	.795
n+prop+adj	.833	.833	.833	.830	.819	.824
n+adj+v	.813	.813	.813	.809	.800	.804
n+prop+v	.835	.835	.835	.833	.822	.827
prop+adj+v	.816	.816	.816	.811	.801	.805
n+prop+adj+v	.839	.839	.839	.838	.827	.831

Table 4. Publico performance values for morphosyntactic experiments.

4.5. Syntactic information

Even if our initial expectations about using syntactic information for text classification were low, we made some experiments with it.

Besides using the syntactic-tree representation (tre), and aiming to access the discriminating power of the structured representation, we considered other representations with information retrieved from the parse trees: a *bag-of-words* representation (bag) and a *sequence-of-words* representation (seq, an ordered tree, where words are root children).

4.5.1 Results

For this level of information we made two different experiments: one included the all document sentences (tot) and the other only the ones considered more informative – finite clauses with subject, predicate and direct object (fcl). For the fcl setting, we also made experiments including only the first n sentences of each document (trying to access if the first sentences have all the necessary information to classify news documents), with $n \in \{1, 3, 5, 10\}$. Table 5 shows the obtained performance measures.

The **bag.tot** experiment (in italics) has the best significant values for all measures. For the other experiments, we present in boldface the values with no significant difference when compared the second best value of each measure.

		π^μ	ρ^μ	f_1^μ	π^M	ρ^M	f_1^M
tre	tot	.812	.812	.812	.810	.788	.792
	fcl	.789	.789	.789	.789	.765	.770
	fcl ₁	.675	.674	.674	.544	.526	.530
	fcl ₃	.699	.699	.699	.683	.667	.670
	fcl ₅	.734	.734	.734	.720	.702	.706
	fcl ₁₀	.782	.782	.782	.776	.757	.760
seq	tot	.829	.829	.829	.821	.811	.814
	fcl	.819	.819	.819	.814	.801	.804
	fcl ₁	.672	.671	.671	.549	.534	.539
	fcl ₃	.708	.708	.708	.698	.681	.686
	fcl ₅	.761	.761	.761	.750	.738	.741
	fcl ₁₀	.791	.791	.791	.779	.771	.772
bag	tot	.857	.857	.857	.858	.842	.847
	fcl	.824	.824	.824	.823	.809	.811
	fcl	.824	.824	.824	.823	.809	.811
	fcl ₁	.613	.613	.613	.605	.588	.594
	fcl ₃	.734	.734	.734	.725	.707	.711
	fcl ₅	.790	.790	.790	.782	.765	.768
	fcl ₁₀	.815	.815	.815	.803	.793	.793

Table 5. Pub9510 performance measures for syntactic experiments.

5. Evaluation

Looking at Table 3 one can say that it was possible to reduce the number of attributes (t_{50}) without compromising performance. However, these values were achieved only for experiments with the original words and not with lemmatization. It also seems that the mutual information filtering function should be used with $tfidf$ weighting, while when filtering by the term frequency one, the weighting function is indifferent.

Concerning the morphosyntactic information study (Table 4), it was possible to achieve an equivalent performance to the “best” morphological setup when combining all four word classes together ($n+prop+adj+v$).

Taking into account the syntactic information experiments (Table 5) it is possible to say that structured representations introduce noise to text classification problems, since the best results were obtained using the bag-of-words representation. It also seems that adding information about sentence constituents and grammatical word class (in a structured way) damages the learner.

Since syntactic information experiments were made with the Pub9510 corpus, another SVM was run with it using the morphological “best” setup ($lm-fr-tfc-t_1$). Table 6 reproduces the “best” syntactic experiment along with the obtained Pub9510 morphological performance measures.

Values from the morphological setup are significantly better than the syntactic ones. In this way, and as expected, sentence syntactic structure does not properly reveal document class.

	π^μ	ρ^μ	f_1^μ	π^M	ρ^M	f_1^M
Morphological	.855	.855	.855	.854	.840	.844
Syntactic	.812	.812	.812	.810	.788	.792

Table 6. Pub9510 morphological and syntactic performance measures.

6. Conclusions and Future Work

This paper presents a series of experiments, applied to the Portuguese language, aiming at verifying the value of incorporating linguistic information on text classification problems.

Concerning morphological information, results show that, when properly combined, word normalization, filtering and weighting functions and threshold values can sharpen performance. Comparing Publico dataset results with previous work on text classification with Portuguese written documents [6], one can conclude that the best combination depends on the dataset (or its domain).

For the Portuguese language, the use of morphosyntactic information as a term selection function generates classifiers with equal performance as the use of traditional Information Retrieval techniques. Moreover, by selecting words that belong to the name, proper name, adjective and verb word classes it’s possible to reduce the number of terms without decaying performance.

Results’ analysis show that, when using syntactic information, structured representations (syntactic-tree and sequence-of-words) harm the learner. Further more, as initially expected, sentence syntactic structure does not properly reveal document class; it could perhaps expose document writer or the kind of used language (like generic as the newspaper documents vs. specific areas of knowledge like medical or juridic ones).

Previous work made on the English language also studied the impact of linguistic processing on text classification. Moschitti and Basili [12] used linguistic tokens – nouns, verbs and adjectives, proper nouns and complex nominals and tokens augmented with their POS tag in context and concluded that SVM global performances were slightly penalized by the use of NLP-derived features. Scott and Matwin [14] also could not improve bag-of-words results while trying phrase-based and hypernym representations.

Regarding future work, we intend to perform further tests on different collections/domains and languages. It will be important to evaluate if these results are bound to the Portuguese language and/or the kind of the dataset domain.

Moreover, we intend to address the document representation problem by trying document representations that incorporate its semantic information.

References

- [1] J. Backus. The syntax and semantics of the proposed international algebraic of the Zurich ACM-GAMM Conference. In *Proceedings of the International Conference on Information Processing – IFIP Congress*, pages 125–132. UNESCO, Paris, 1959.
- [2] E. Bick. *The Parsing System PALAVRAS – Automatic Grammatical Analysis of Portuguese in a Constraint Grammar Framework*. Aarhus University Press, 2000.
- [3] E. Bick. A constraint grammar based question answering system for portuguese. In *Proceedings of the 11th Portuguese Conference of Artificial Intelligence – EPIA’03*, pages 414–418. LNAI Springer Verlag, 2003.
- [4] N. Chomsky. Three models for the description of language. *IRI Transactions on Information Theory*, 2(3):113–124, 1956.
- [5] M. Collins and N. Duffy. New ranking algorithms for parsing and tagging: Kernels over discrete structures, and the voted perceptron. In *ACL-02, 30th Annual Meeting of the Association for Computational Linguistics*, pages 263–270, 2002.
- [6] T. Gonçalves, C. Silva, P. Quaresma, and R. Vieira. Analysing part-of-speech for Portuguese text classification. In *CICLing-06, 7th international Conference on Intelligent Text Processing and Computational Linguistics*, pages 551–562, Mexico City, MX, February 2006. Springer-Verlag.
- [7] D. Haussler. Convolution kernels on discrete structures. Technical report, Department of Computer Science, University of California at Santa Cruz, 1999.
- [8] T. Joachims. Making large-scale SVM learning practical. In B. Schölkopf, C. Burges, and A. Smola, editors, *Advances in Kernel Methods - Support Vector Learning*. MIT Press, 1999.
- [9] J. Lopes, N. Marques, and V. Rocio. Polaris: PORTuguese Lexicon Acquisition and Retrieval Interactive System. In *The Practical Applications of Prolog*, page 665. Royal Society of Arts, 1994.
- [10] A. Moschitti. A study on convolution kernels for shallow semantic parsing. In *ACL-04, 42nd Annual Meeting on Association for Computational Linguistics*, pages 335–342, Barcelona, SP, 2004.
- [11] A. Moschitti. Efficient convolution kernels for dependency and constituent syntactic trees. In *ECML-06, 17th European Conference on Machine Learning*, pages 318–329, Berlin, DE, 2006.
- [12] A. Moschitti and R. Basili. Complex linguistic features for text classification: a comprehensive study. In *ECIR-04, 26th European Conference on Information Retrieval Research*, pages 181–196, Sunderland, UK, 2004.
- [13] G. Salton and M. McGill. *Introduction to Modern Information Retrieval*. McGraw-Hill, 1983.
- [14] S. Scott and S. Matwin. Feature engineering for text classification. In *ICML-99, 16th International Conference on Machine Learning*, pages 379–388. Morgan Kaufmann Publishers, 1999.
- [15] J. Shawe-Taylor and N. Cristianini. *Kernel Methods for Pattern Analysis*. Cambridge University Press, 2004.
- [16] V. Vapnik. *The nature of statistical learning theory*. Springer, NY, 1995.
- [17] V. Vapnik. *Statistical learning theory*. Wiley, NY, 1998.
- [18] S. Vishwanathan and A. Smola. Fast kernels on strings and trees. In S. Becker, S. Thrun, and K. Obermayer, editors, *Advances in Neural Information Processing Systems*, volume 15, pages 569–576. MIT Press, Cambridge, MA, 2003.
- [19] I. Witten and E. Frank. *Data Mining: Practical machine learning tools and techniques*. Morgan Kaufmann, San Francisco, US, 2nd edition, 2005.
- [20] D. Zhang and W. Lee. Question classification using support vector machines. In *SIGIR-03, 26th ACM International Conference on Research and Development in Information Retrieval*, pages 26–32, 2003.

Performance of Inductive Method of Model Self-Organization with Incomplete Model and Noisy Data*

Natalia Ponomareva,^{1,2} Mikhail Alexandrov,^{3,4} Alexander Gelbukh^{5,4}

¹ University of Wolverhampton, UK

² Polytechnic University of Valencia, Spain

³ Autonomous University of Barcelona, Spain

⁴ Social Networks Research Center, University of Central Europe in Skalica, Slovakia

⁵ Center for Computing Research, National Polytechnic Institute, Mexico

{nponomareva,dyner1950}@mail.ru, www.gelbukh.com

Abstract

Inductive method of model self-organization (IMMSO) developed in 80s by A. Ivakhnenko is an evolutionary machine learning algorithm, which allows selecting a model of optimal complexity that describes or explains a limited number of observation data when any a priori information is absent or is highly insufficient. In this paper, we study the performance of IMMSO to reveal a model in a given class with different volumes of data, contributions of unaccounted components, and levels of noise. As a simple case study, we consider artificial observation data: the sum of a quadratic parabola and cosine; model class under consideration is a polynomial series. The results are interpreted in the terms of signal-noise ratio.

1. Introduction

1.1. Inductive method of model self-organization

We study the Inductive Method of Model Self-Organization (IMMSO), an evolutionary machine-learning (ML) algorithm that has important applications in Data Mining and other ML tasks. This method aims to determine the optimal complexity of a model that could describe or explain a given set of experimental observation data. Note that the method determines the complexity of the model and not the specific values of its parameters.

By a *model* we mean a formula, equation, algorithm, etc. Since IMMSO was originally developed in the Soviet Union and there is insufficient literature available on it in English, we will briefly describe the method and the conditions of its application.

IMMSO was suggested and developed by Ivakhnenko *et al.* in 80s [2, 4, 9]. This method does not require any a priori information on distribution of parameters of the objects under consideration. The method is very efficient in Data Mining problems under these circumstances. However, if a priori information is available, then it can be outperformed by other methods, such as some Pattern Recognition methods, which can give significantly better results.

The method has one restriction: it is not applicable to a continuous class of models, because it is based on the competition of models. This is why this method is called *inductive*. Its main principle of model selection is the principle of stability: the models describing different subsets of a given data set must be similar.

Here is how the IMMSO works:

1. An expert defines a sequence of models, ordered from the simplest to more complex ones.
2. Experimental data are randomly divided into two data sets: training data and control data.
3. For a given kind of model, the best parameters are determined using first the training data and then the control data set. For this, an internal criterion of concordance between the model and the data can be used, e.g., the least squares criterion.
4. Both models are compared on the basis of an external criterion, such as the criterion of regularity, criterion of unbiasedness, etc. If this external criterion, or a combination of such

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criteria, achieves a stable optimum, the process is finished. Otherwise, more complex model is considered and the process is repeated from the step 3.

The method uses the notions of *internal criteria* and *external criteria*. The difference is that internal criteria use the same data both to determine the model parameters and to evaluate its quality, while external ones use different data for this purpose. In practice, usually external criteria are constructed as non-negative functions with zero being the optimum.

The external criterion can reach an optimum when the data contain:

- a regular component defined by the chosen model structure, and
- a random component: noise.

Indeed, a too simple model ignores the noise, but at the same time it does not recognize the regularities in the data. On the other hand, a too complex model can reflect very well such regularities but also will be too sensitive to noise. In both cases, the differences between two models will be significant and the values of the penalty function—external criterion, large. The principle of model *self-organization* consists in that an external criterion passes through its optimum as the model complexity is gradually increased. It has been proved that for noisy data and a small data sample, the minimum of mathematical expectation of an external criterion is unique [4].

However, if a chosen class of models does not reflect the regularities in the data or reflects them only partially then the “random” component in the data is not exactly noise but rather an undefined, or unaccounted for, component. The contribution of such a component to experimental data can be comparative with the contribution of the model component. This distorts the results and the minimum found by the algorithm may be located far from the real best point. This is why it is very important for the user to correctly choose the model class. This choice should be based on the knowledge of the nature of the objects described by the data. If there is no information about this nature, some general class of models that is guaranteed to include the real model class is chosen.

IMMSO was successfully used in numerous applications in science and engineering for identification of physical laws, approximation of multidimensional processes, and short-term and long-term forecasting of processes and events [2, 9]. For example, it was used in geology [3], applied mathematics [6, 13], and text processing [1, 10, 12]. A

review of various variants of IMMSO can be found in [5]. Further development of inductive modeling is considered in [7, 8].

1.2. Limitations of the method

The IMMSO can give unsatisfactory results if:

1. The data volume is very limited
2. The chosen model class essentially differs from the real model underlying the data’s nature
3. The level of noise is too high

In more detail, we can comment these conditions as follows:

1. *Data volume.* Obviously, the limited data set does not allow revealing highly complex models. If model complexity is associated with the number of its parameters then the number of data points can not be less than the number of model parameters. In practice, this condition is much more rigid; see Section 2.1.
2. *Model class.* If the selected model class does not reflect the real structure of the objects being modeled, then the results are unpredictable. Usually, in this case oversimplified models are suggested by the method, due to the principle of stability—a consequence of the criterion of regularity; see Section 2.3.
3. *Noise level.* If the level of external noise is too high then the results depend on a specific measurement. In this case, again, usually oversimplified models are suggested by the method due to the principle of independence from data—a consequence of the criterion of unbiasedness; see Section 2.3.

1.3. Research questions

Noise immunity of inductive modeling was considered in detail in [2, 4], and other works related with Neural Network training [8]. They concern mainly the influence of white noise. In this work, we study inductive modeling in presence of unaccounted components in addition to the noise.

As case study, we consider polynomial functions. Obviously, the results may vary if another class of functions is considered, for example, Fourier series, etc., or with another type of models, such as differential models. However, we believe that our qualitative conclusions will still hold.

The paper is structured as follows: Section 2 describes our experimental settings and presents the definitions of the external criterion. Section 3 presents the obtained results. Section 4 gives the conclusions.

2. Experimental settings

2.1. Model class

Let us first define the notions of *model*, *components of model*, *model complexity*, and *noise*. The notion of a model in natural and technical sciences implies the following:

- Mathematical model represented by an equation or a system of equations
- Algorithmic model represented by a list of rules for data transformation

In this paper, mathematical models are considered in the following form:

$$\psi(x) = F(x) + \varphi(x) + \omega(x), \quad (1)$$

where x is an independent variable defining the data observations points; $F(x)$ is a numerical function of a given class reflecting principal components of the model; $\varphi(x)$ is a numerical function reflecting unaccounted for—additional—components of the model, and $\omega(x)$ is a random component—noise.

As a case study, we assume the following classes of functions:

- Principal components correspond to the members of a polynomial: $F(x) = a_0 + a_1x + a_2x^2 + \dots$, where a_0, a_1, a_2 , etc. are unknown coefficients.
- Unaccounted for component is a periodical function $\varphi(x) = b \cos(kx)$, where b is a parameter defining its contribution to the observation data; k defines variability of the additional component, which can be close or far from the variability of the principal components.
- Noise $\omega(x)$ is a random function with zero mean value and with variance s^2 .

The models (1) are used in time series analysis related with seasonal changes of temperature. These models have been considered in [2].

Note that given the coefficient $k \geq 5\pi$, the period of harmonic function is $T \leq 2\pi/5\pi = 0.4$. It is easy to see that such a period allows the function to be defined on the interval $[-1,1]$ at least 5 times.

When applying the IMMSO, we suppose that:

- Additional component $\varphi(x)$ is unknown
- Variance of noise is unknown

The IMMSO implies determining the best polynomial in the form

$$F(x) = a_0 + a_1x + a_2x^2 + \dots \quad (2)$$

It is well-known that the method of least squares, which is usually used in IMMSO, decreases the effect of noise $\sqrt{n}$ times when the number of data points exceeds the number of parameters of the model n times. Therefore, in order to decrease the effect of noise 2 or 3 times, the number of observations should exceed the number of parameters 5–10 times. This is what is usually recommended in practice by the IMMSO experts and developers [2].

With this, to study the effect of noise with a small number of data points, one needs to make several measurements of the noise component and then generalize the results.

The term *model complexity* usually denotes the number of components of the model and the strength of relations between them. In case of polynomial function, the model complexity can be defined as the highest power of the polynomial.

2.2. Observation data

The observation data we used for our experiments are the values of function $G(x) = x^2 + b \cos(5\pi x)$ calculated at N points from the interval $[-1,1]$, where x^2 is the principal component of the model and $b \cos(5\pi x)$ as the additional component. In our experiments, we consider the following values: $b \in \{0.1, 0.2, 0.5\}$, i.e., 10%, 20%, and 50% of the maximum value for the principal component x^2 on the interval $[-1,1]$, respectively.

We added to the data some noise: normally distributed random values with zero mean and root-mean-square value $s \in \{0.1, 0.2, 0.5\}$. It also gives 10%, 20%, and 50% of the maximum value for x^2 on the interval $[-1,1]$.

The number of points is $N \in \{50, 1000\}$. Therefore the training set and the control set contain 25 and 500 points, respectively. Consider a complete polynomial (2) of the fourth order. It contains 5 members. In this case, each polynomial parameter is “covered” by 5 and 100 points, respectively. This decreases the error of noise $\sqrt{5} \approx 2$ and $\sqrt{100} = 10$ times, respectively. In case of small number of points ($N = 50$) and high level of noise ($s = 50\%$), we implemented 5 realizations of the noise component and found the solution for each realization.

Table 1. Energetic ratios between model components

Ratio	b/s	0.1/0.1	0.1/0.2	0.1/0.5	0.2/0.1	0.2/0.2	0.2/0.5	0.5/0.1	0.5/0.2	0.5/0.5
W_ϕ/W_F		0.025	0.025	0.025	0.10	0.10	0.10	0.625	0.625	0.625
s^2/W_F		0.05	0.20	1.25	0.05	0.20	1.25	0.05	0.20	1.25

2.3. Evaluation of results and energetic ratios

For application of the IMMSO, the following complete polynomial models are considered:

$$\begin{aligned} F_0(x) &= a_0 \\ F_1(x) &= a_0 + a_1x \\ F_2(x) &= a_0 + a_1x + a_2x^2 \\ F_3(x) &= a_0 + a_1x + a_2x^2 + a_3x^3, \text{ etc.} \end{aligned}$$

We evaluate correctness of a solution as coincidence of orders of polynomial models. That is, if the second order polynomial is revealed (independently of the number of its members) then the method is considered to give a correct result. Note that the specific values of polynomial coefficients do not matter: we are only interested in the optimal complexity of the model, not in the specific optimal values of its parameters.

If the coefficient of the highest polynomial member is significantly less than the coefficient of the next member—say, by 2 orders of magnitude—then the model complexity should be decreased. For example, the following third order polynomial $F(x) = 0.3 + 0.1x + 7x^2 + 0.05x^3$ on the interval $[-1,1]$ has in effect second order. However, all such cases proposed by the method can be considered to make a decision in practice. In any case, such a solution should be justified, taking into account, for example, the level of noise, etc.

Obviously, our function in the form of a polynomial function, periodical component, and noise is a particular case of empirical models that could be treated with IMMSO. In order to generalize the results, one should describe the models considering *energetic ratios*.

The power and the root-mean-square value of any function $f(x)$ defined on a given interval T is given by

$$W = \frac{1}{T} \left[\int_T f^2(x) dx \right]. \quad (3)$$

For the functions $F(x) = x^2$ and $\phi(x) = b \cos(kx)$ we have, respectively,

$$W_F = 1/5, \quad W_\phi = b^2/2.$$

Table 1 shows the ratios $\eta = W_\phi / W_F$ and $\varepsilon = s^2/W_F$ for the selected values of amplitude b (additional

component) and selected values of root-mean-square s (noise).

It is easy to see that our experimental conditions are rather complex: the ratio of additional to principal components varies from some 3% to some 60%, and the ratio of noise to principal component varies from 5% to 125%.

2.4. Methods and criteria

In our experiments, the IMMSO was to find a polynomial function of a given form (2). The observation data (artificial data) were divided into training and control sets, the experiments were conducted, and the external criteria were calculated. The winner was defined on the basis of minimum of one or several external criteria described below.

The models, which were constructed at each step of the IMMSO algorithm, used the Least Squares Method (LSM) [11]. Recall that we have two separate models: the first one based only on the training data set, and the second one based only on the control data set. In constructing these models, LSM minimizes the variance of error between the model predictions and the observation data:

$$\varepsilon^2 = \|F - D\|, \quad (4)$$

where $F = \{F_i\}$ is the vector of model function values at the points of observations $\{x_i\}$ and $D = \{D_i\}$ is the vector of the observation data.

There are many variants of external criteria. Usually, in applications of the IMMSO the following two criteria are considered:

- criterion of regularity K_r
- criterion of unbiasedness K_u

Both criteria use the training data set and the test data set. The criterion of regularity reflects the difference between the model data and the control data when the model is constructed on the basis of training data set. So, this criterion evaluates the *stability of the model with respect to data variation*. The criterion of unbiasedness reflects the difference between the two models—those constructed on the training set and on the control set, respectively. So, this criterion evaluates *independence of the model from the data*.

Table 2. Values of combined criterion for $N = 50$

Realization of noise	$F_0(x)$	$F_1(x)$	$F_2(x)$	$F_3(x)$	$F_4(x)$	$F_5(x)$
Noise 1	0.6540	0.7989	0.967	0.9694	1.0629	1.0612
Noise 2	0.7040	0.7522	0.7219	0.7312	0.8234	0.7954
Noise 3	1.0774	1.1167	1.0193	1.1010	1.1014	1.1344
Noise 4	0.4885	0.6407	0.5565	0.5655	0.5473	0.5049
Noise 5	0.6419	0.6166	0.5117	0.5659	0.6545	0.9778
Average	0.71316	0.78502	0.75528	0.7866	0.8379	0.89474

Table 3. Values of combined criterion for $N = 1000$

Realization of noise	$F_0(x)$	$F_1(x)$	$F_2(x)$	$F_3(x)$	$F_4(x)$	$F_5(x)$
Noise 1	0.5855	0.5852	0.4833	0.4833	0.4842	0.4845
Noise 2	0.5790	0.5858	0.5026	0.5033	0.5051	0.5056
Noise 3	0.5754	0.5973	0.5346	0.5353	0.5353	0.5397
Noise 4	0.5853	0.5866	0.5238	0.5394	0.5433	0.5444
Noise 5	0.5774	0.5817	0.5416	0.5440	0.5429	0.5433
Average	0.5805	0.5873	0.5172	0.5211	0.5222	0.5235

Different forms of these criteria can be proposed. A specific form depends on a specific given problem. In our case, we use these criteria in the following forms:

$$K_r = \frac{\sqrt{\sum_C (q_i(T) - q_i)^2}}{\sqrt{\sum_C q_i^2}}, \quad K_u = \frac{\sqrt{\sum_{T+C} (q_i(T) - q_i(C))^2}}{\sqrt{\sum_{T+C} q_i^2}}, \quad (5)$$

where T and C are the systems of equations (2) used for training and control, respectively; $q_i(T)$ and $q_i(C)$ are the model data that is the right part of the equations with the parameters determined on the training and control set, respectively; q_i are the experimental (artificial) data, i.e., the left part of the equations; i is the number of the equation.

Sometimes a model can be better than another one according to the first criterion but worse according to the second one. A combined criterion can be used, e.g.:

$$K = \lambda K_r + (1 - \lambda) K_u, \quad (6)$$

where λ is a user-defined coefficient of preference. In our experiments, we used $\lambda = 2/3$, i.e., we consider the criterion of regularity as more important.

3. Results of inductive modeling

3.1. Sensitivity to the volume of data

We used the following data in this experiment:

- amplitude of the additional component $b = 0.1$ (the lowest level)
- level of noise $s = 0.5$ (the highest)

For this case, we have the relations $\eta = 2.5\%$ and $\varepsilon = 125\%$.

We considered 5 realization of noise. Tables 2 and 3 show the values of combined criterion (6). In graphical form this is shown in Fig. 1.

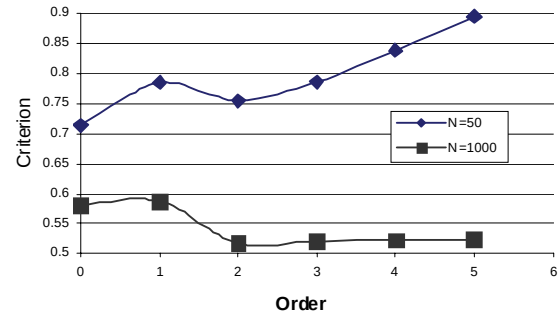


Fig. 1 Dependence on volume of data

One can see that the realizations significantly differ from each other. The average value reaches its minimum for the 0-order model. However, in 2 cases of 5, the winner is the second order model. This means that the result was correct in 40% of our test runs.

In all cases, the minimum is reached for the second order model, due to strong filtration.

3.2. Sensitivity to inexactness of model

We used the following data in this experiment:

- total number of points $N = 50$
- level of noise $s = 0.1$ (the lowest)

Table 4. Average values of combined criterion for $b \in \{0.1, 0.2, 0.5\}$

Additional component	$F_0(x)$	$F_1(x)$	$F_2(x)$	$F_3(x)$	$F_4(x)$	$F_5(x)$
$b = 0.1$ ($\eta=2.5\%$)	0.4727	0.4954	0.2030	0.2162	0.2200	0.2271
$b = 0.2$ ($\eta=10\%$)	0.5061	0.5341	0.2760	0.2877	0.2903	0.2942
$b = 0.5$ ($\eta=62.5\%$)	0.5646	0.5935	0.4236	0.4319	0.4207	0.4241

Table 5. Spectrum of x^2

Harmonics	const	$\cos(\pi x)$	$\cos(2\pi x)$	$\cos(3\pi x)$	$\cos(4\pi x)$	$\cos(5\pi x)$
Value	0.223	0.084	0.005	0	0	0

Table 6. Average values of combined criterion for $s \in \{0.1, 0.2, 0.5\}$

Noise	$F_0(x)$	$F_1(x)$	$F_2(x)$	$F_3(x)$	$F_4(x)$	$F_5(x)$
$s = 0.1$ ($\varepsilon=5\%$)	0.4847	0.5199	0.2027	0.2080	0.2103	0.2131
$s = 0.2$ ($\varepsilon=20\%$)	0.5489	0.5955	0.3387	0.3483	0.3711	0.3827
$s = 0.5$ ($\varepsilon=125\%$)	0.6439	0.7070	0.6134	0.6421	0.6535	0.6640

For this case, we have $\varepsilon = 5\%$. In this experiment we considered five realizations of noise for each value of b . Table 4 contains the average values of combined criterion (6). The results are presented in graphical form in Fig. 2.

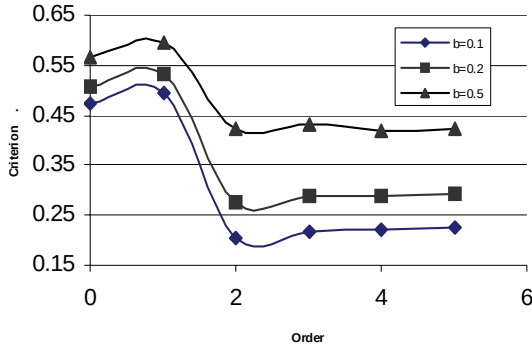


Fig. 2. Dependence on the unaccounted for function

The best model was the one of second order for all values of b . This model was the best for 12 realizations out of 15; in the rest of the cases the winner was the third order model. Therefore, we obtained the correct result in 80% of cases.

It should say that the variability of principal and unaccounted for components, i.e., x^2 and $b\cos(5\pi x)$, is significantly different. Table 5 presents the spectrum of x^2 on the interval $[-1,1]$.

It is easy to see that since second harmonic the spectrum of the principal component is practically zero. The additional component has also fifth harmonic $\cos(5\pi x)$. Therefore, both components are orthogonal. This is why the obtained results are rather good. However, this issue needs especial attention.

3.3. Sensitivity to noise

We used the following data in this experiment:

- total number of points $N = 50$
- amplitude of the additional component $b = 0.1$ (the lowest)

For this case we have $\eta=2.5\%$. We considered three realizations for each level of noise. Table 6 shows the average values of the combined criterion (6). These results are presented in graphical form in Fig. 3.

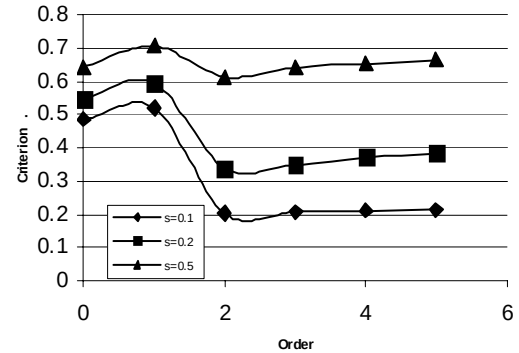


Fig. 3. Dependence on noise

The winner was the second order model for all values of s . With this, for $s = 0.1$ all realizations provided the correct result; for $s = 0.2$, we obtained two incorrect results out of five; for $s=0.5$, two incorrect results out of five as well.

Table 7. Values of all criterion for the case $b = s = 0.1$

Criteria		$F_0(x)$	$F_1(x)$	$F_2(x)$	$F_3(x)$	$F_4(x)$	$F_5(x)$
Regularity	K_r	0.6945	0.7002	0.2299	0.2301	0.2374	0.2374
Unbiasedness	K_u	0.0367	0.1102	0.0760	0.0849	0.1094	0.1158
Combined criterion	K	0.4752	0.5035	0.1786	0.1817	0.1947	0.1969

Table 8. Values of all criterion for the case $b = s = 0.5$

Criteria		$F_0(x)$	$F_1(x)$	$F_2(x)$	$F_3(x)$	$F_4(x)$	$F_5(x)$
Regularity	K_r	0.91954	0.93776	0.79044	0.8038	0.80218	0.79534
Unbiasedness	K_u	0.15672	0.28392	0.31076	0.3478	0.41604	0.44244
Combined criterion	K	0.66526	0.71982	0.63054	0.6518	0.67348	0.67770

3.4. Model self-organization with different criteria

In this experiment, we used the minimum number of points $N = 50$.

1. The best case for modeling:

- amplitude of additional component $b = 0.1$ (the lowest)
- level of noise $s = 0.1$ (the lowest)

For this case we have $\eta = 2.5\%$ and $\varepsilon = 5\%$. The results are presented in Table 7 and in Fig. 4.

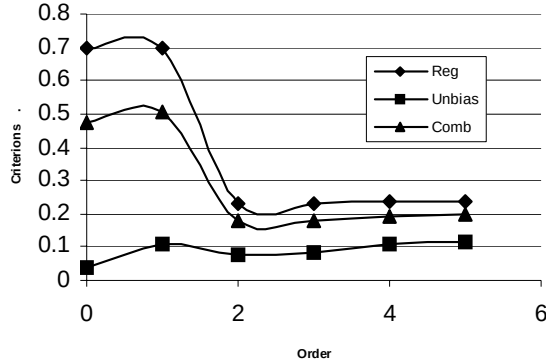


Fig. 4. Criteria for the best case

One can see that the criterion of regularity gives the correct result: the second order model. The criterion of unbiasedness gives the best result for 0-order model, with a local minimum is at the second order model. The combined criterion reaches its minimum on the correct second order model.

2. The worst case for modeling:

- amplitude of the additional component $b = 0.5$ (the highest)
- level of noise $s = 0.5$ (the highest)

For this case we have $\eta = 62.5\%$ and $\varepsilon = 125\%$. The results are presented in Table 8 and in Fig. 5.

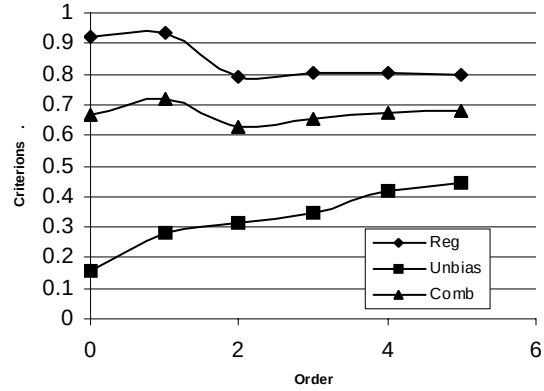


Fig. 5. Criteria for the worst case

One can see that the criterion of regularity gives the correct result: second order model. The criterion of unbiasedness gives the best result for 0-order model. The combined criterion reaches its minimum on the correct second order model, though it is rather smooth and does not distinguish the models well.

4. Conclusions

We studied the performance of the inductive method for the test case of polynomial model with unaccounted for trigonometric function and some noise. In general, the method demonstrated low sensitivity to noise. The results depend on the ratios of the principal component, the additional components, and the noise.

Usually, the works studying noise sensitivity for Ivakhnenko's inductive method consider normal white noise. In the paper, we considered another case, when this distortion is created by unaccounted for model components in addition to the usual noise.

We have presented the results of inductive modeling for different volumes of data, and contributions of unaccounted for model component and the level of noise. Such results can be easily generalized to take into account the ratios of principal component, additional component, and noise.

In the future, we plan to use spectral approach to analyze the noise sensitivity of inductive modeling.

5. References

- [1] Alexandrov, M., Blanco, X., Ponomareva, N., Rosso, P. (2007): Constructing Empirical Models for Automatic Dialog Parameterization. In: *Text, Speech, Dialog*, TSD 2007. Springer, LNCS 4629, 455–462.
- [2] Ivakhnenko, A. (1982): *Inductive Method of Model Self-Organization of Complex Systems*, Kiev, Tekhnika Publ. (in Russian).
- [3] Ivakhnenko, A., Peka, G., Vostrov, N. (1984): *Combined Method for Modeling Petroleum and Water Fields*. Kiev, Naukova Dumka Publ. (in Russian).
- [4] Ivakhnenko, A., Stepashko, V. (1985): *Noise Immunity of Modeling*, Naukova Dumka Publ. (in Russian).
- [5] Ivakhnenko, A., Ivakhnenko, G. (1995): The Review of Problems Solvable by Algorithms of the Group Method of Data Handling (GMDH). *Pattern Recognition and Image Analysis*, Vol. 5, No. 4, 527–535.
- [6] Ivakhnenko, A., Ivakhnenko, G. (1995): Simplified Linear Programming Algorithm as Basic Tool for Open-Loop Control. *System Analysis Modeling Simulation (SAMS)*, vol. 12, No. 1, 11–18.
- [7] Ivakhnenko, A., Ivakhnenko, G. (2000): Problems of Further Development of the Group Method of Data Handling Algorithms. Part I. *Pattern Recognition and Image Analysis*, vol. 10, No. 2, 187–194.
- [8] Ivakhnenko, A., Ivakhnenko, G., Savchenko, E., Wunsch, D. (2002): Problems of Further Development of GMDH Algorithms: Part 2. *Pattern Recognition and Image Analysis*, Vol. 12, No. 1, 6–18.
- [9] Madala, H., Ivakhnenko A. (1994): *Inductive Learning Algorithms for Complex System Modeling*. CRC Press.
- [10] Makagonov, P., Alexandrov, M. (2002): Constructing empirical formulas for testing word similarity by the inductive method of model self-organization. In: *Advances in Natural Language Processing*. Springer, LNAI 2389, 239–247.
- [11] Shoup, T. (1982): *Solution of engineering problems with computers*.
- [12] Sidorov, G. (2005): Linguist’s Intelligent Workplace for Spanish Language. In: *Advances in artificial intelligence and computer science*, J. *Research on computing science*, vol. 14, pp. 173–178.
- [13] Triseev, Yu. P. (1987): Approaches to the Solution of Mathematical Programming Problems on the Basis of Heuristic Self-Organization. *Automation and Information Sciences*, vol. 20, No. 2, 30–37 (in Russian).

Pattern Recognition

Classification of Data Sequences by Similarity Analysis of Recurrence Plot Patterns

E. Bautista-Thompson

R. Brito-Guevara

Centro de Tecnologías de la Información

DES-DACI, Universidad Autónoma del Carmen

Ciudad del Carmen, Campeche, México

ebautista@pampano.unacar.mx

rbritoguevara@hotmail.com

Abstract

Quantification of the similarity between data sequences is important in different database and data mining tasks such as indexing, retrieving, clustering and classification, many similarity metrics (Euclidean, DTW, among others) operate directly in the raw representation of the sequences, but this implies when the sequences are compared, not take into account information about the collective behavior of the data that forms a sequence and the hidden relations between such data, such information can be important for classification of sequences based on their structures and their relations with the dynamics that such structures can represent (e.g. stationary, random, complex). We propose a computational technique for similarity analysis and classification of recurrence plot patterns: RecurrenceVs. The results show that the proposed technique is able to classify data sequences by similarity families based on the recurrence plot patterns, which preserve the information about the structure and dynamics represented by the data sequences.

1. Introduction

The extraction of common structural features in sets of time series and sequences of data is an important task in the identification of patterns of interest for example: in the analysis of time series dynamics [1], the identification of motifs in genomic sequences [2], the construction of queries for sequence extraction in databases [3]. Similarity in data sequences can be measure with different metrics such as Lp Norms [4], Dynamic Time Warping [5], Similarity Histograms [6], etc.; such measures operate directly over the data sequences. For other side, analysis of the dynamics of sequences of data, understood as the relationships

between the different data inside a sequence, can be done with different techniques for example: Singular Spectral Analysis [7], Empirical Mode Decomposition [8], and Recurrence Quantification Analysis [9]. In particular, the recurrence quantification analysis is based on the generation of a data representation named recurrence plots, these plots shows hidden relationships between data such as non-stationary behavior, periodicities, and randomness; the patterns generated with these plots can be used to compare sequences of data, at a new level of information. There are not quantitative comparison tools for recurrence plots, software such as VRA, RQA and CRP lacks this functionality [10, 11, 12], this is the main motivation for the development and proposal in this work of a computational technique for quantitative comparison and analysis of similarity between recurrence plots. Section two, explains the theoretical basis of the recurrence plots. Section three, describes the algorithmic aspects of RecurrenceVs, the software tool that implements the technique. In Section fourth, the experimental results of the evaluation of the software tool are presented. Finally in section five, the conclusions of this work are presented.

2. Recurrence plots

Recurrence plots are graphical representations of sequences of data, which allows the detection of hidden dynamical patterns and nonlinearities inside the data. These plots allow the visualization of recurrent patterns, non-stationary patterns, and structural changes. The recurrence plots are part of a technique known as Recurrence Analysis that was developed by Eckman et. al. in 1987 [9]. The patterns generated by the recurrence plots multiplies the information of a sequence of data (time series, genomic sequence, etc.), a fundamental idea behind a sequence of data that

represents a measure of an observable from a particular system is that this set of data contains multidimensional information about such system [13, 14].

A recurrence plot is generated by comparison of each datum in a sequence with itself and with the rest of data.

$$x(i) = \{x(1), x(2), x(3), \dots, x(n)\} \quad (1)$$

This comparison is made with a metric such as Euclidean distance; this allows to build a correlation matrix of spatial and temporal nature (as is the case of time series).

$$d_{ij} = \|x(i) - x(j)\| \quad (2)$$

$$D = \begin{bmatrix} d_{11} & d_{12} & \dots & d_{1n} \\ d_{21} & d_{22} & \dots & d_{2n} \\ \vdots & \vdots & \ddots & \vdots \\ d_{n1} & d_{n2} & \dots & d_{nn} \end{bmatrix} \quad (3)$$

Each element d_{ij} in matrix D is associated with the Euclidean distance between a datum in position i and a datum in position j inside a sequence, if a datum or subsets of data are recurrent this behavior will be detected by means of sets of equal distances. Also, each distance d_{ij} is associated with a value from a discretized gray tone scale, in this way an image of the recurrence plot pattern is generated. The resultant pattern is symmetric due to the redundancy in the calculation of the distances $d_{ij} = d_{ji}$.

Comparing the patterns generated by the recurrence plots can be useful in order to identify sequences of data with similar behavior or dynamics, this similarity must be quantified in order to have an objective comparison of the recurrence plots and be able to classify them. In the next section the algorithms for the comparison method are described.

3. RecurrenceVs: algorithms

The design of the algorithms was divided in three stages: first the load and preparation of the data, second the construction and comparison of the recurrence plots and third the visualization and storage of the results. In the first stage, an important aspect is the normalization of the data sequences to be compared, the data sequences can have different length, and data range (the data types used are: integer and real), in order to generate comparable recurrence plots they are normalized to a default length of 1000 data by sampling each sequence, but this parameter of length can be adjusted or not used if shorter time series

are used in the similarity analysis. The normalization in range is done by scaling the values in order to have a range of values between 0 to 1; the Figure 1 shows a diagram of this stage.

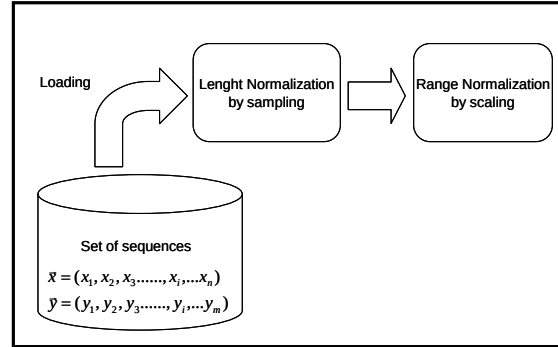


Figure 1. Stage one, normalization of the sequences of data to be compared.

The construction of the recurrence plots is made as was described in section two. The comparison between two recurrence plots is made by comparing the corresponding vector elements (i, j, d_{ij}) of the matrix of distances D for each one of the recurrence plots, for each position the difference between the distances d_{ij} is calculated.

$$diff_{d_{ij}} = \|(d_{ij})_{rp1} - (d_{ij})_{rp2}\| \quad (4)$$

A distance d_{ij} is considered as a match between both recurrence plots if the distance difference (see equation 4), has a value below a similarity threshold S_t .

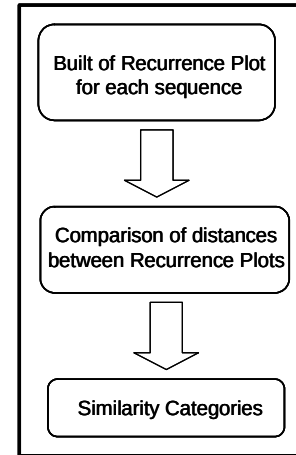


Figure 2. Stage two, recurrence plots construction and comparison between them.

The input parameter S_t is setup by the user before the beginning of the analysis, the threshold allows to establish different degrees of similarity between the recurrence plots, the values of the differences between the distances d_{ij} are discretized as 1 for those below the

threshold and 0 above the threshold and are storage in a binary distance vector, a counting of the number of minimum differences (matches) is done, in order to determine the percentage of minimum distances between the compared recurrence plots, a set of five similarity percentages are established by the user, and depending on the corresponding percentage reached by the counting process a similarity category between the recurrence plots is determined. The Figure 2 shows the steps of this second stage.

Finally, in the third stage the visualization of the recurrence plots and their results are showed, the recurrence plots are visualized using a gray scale that corresponds with the distance between their points, the white color represents the minimum distance and the black color corresponds to the maximum distance present in the plot, each graphic of a recurrence plots is generated with a sampling of their corresponding distance set in order to facilitate its visualization, this is due to the enormous quantity of data generated, for example for a sequence of 1000 data a matrix of 10^6 points is generated. In this stage, the option to save the recurrence plots and the results of their comparison is activated, multiple comparison experiments can be done in one run, for example: comparing a specific recurrence plot against a set of different recurrence plots can be saved by incremental storage of each new experiment, in the Figure 3 is showed the steps of this last stage.

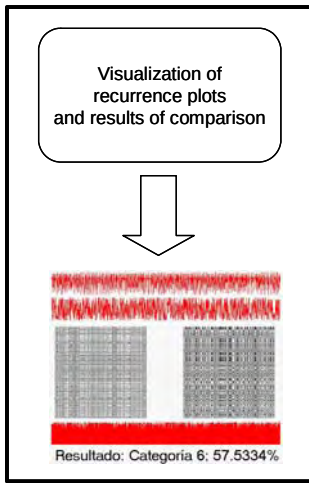


Figure 3. Stage 3, visualization of results.

4. Experimental results

The technique was evaluated with a set of synthetic data sequences previously classified by their similarity degree by means of the direct comparison of the

sequences using the technique of Derivative Dynamic Time Warping (DDTW) [5, 15]. Examples of different comparisons are showed in the screenshots of Figure 4 and Figure 5; in these examples a recurrence plot from a sequence tagged DS11 is compared with the corresponding recurrence plots for the sequences DS12 and DS42; in the previous classification the similarity of these sequences groups DS11 and DS12 in the same class and the sequence DS42 belongs to another class. The results with RecurrenceVs show a correspondence with the aforementioned results.

The experimental set contains a total of 55 synthetic data sequences that belongs to 6 different classes that were previously assigned in [5, 15], in order to validate the proposed technique, randomly from each class a sequence was selected and compared with all the set, in this way a new classification tree was generated and compared with the classification tree reported in [5, 15], so we identified what differences result from the analysis of similarity between the sequences in a recurrence plot representation with respect to their direct or raw data representation. In Figure 6 is show, the behavior of the similarity percentage for the similarity analysis made with the RecurrenceVs tool, for the sequence DS50 that belongs to the class 1 in the DDTW classification. In Figure 7 is show, the behavior of the similarity percentage for the sequence DS12 that belongs to the class 5 in the DDTW classification.

In the six cases developed, the similarity analysis with recurrence plots groups the sequences in their corresponding classes (two examples are the Figure 6 and Figure 7) as in the DDTW classification, so the change in representation preserve the information about the general similarities in structure between sequences from the same family or class, but because we are comparing in a new representation as recurrence plots, the similarity at the level of complete families is different, this was observed when the classification tree was built with data of a matrix of average similarity percentages for each family with respect to the representative sequence used as reference (DS8, DS12, DS29, DS37, DS50, DS53), the Figure 8 shows the classification tree derived from the analysis with RecurrenceVs and the Figure 9 shows the corresponding tree derived from the DDTW classification in references [5, 15]. These differences in the similarity between families correspond with the new information about the data sequences structure extracted with the recurrence plot patterns.

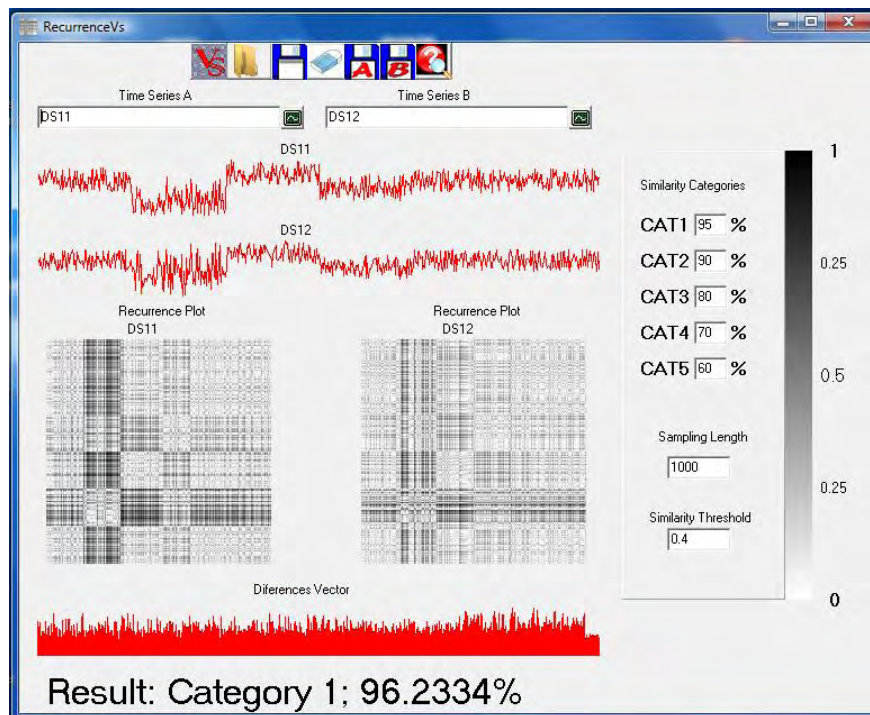


Figure 4. Screenshot of the similarity analysis for the recurrence plots of two synthetic data sequences DS11 and DS12, their similarity corresponds to category 1 (95% of similarity).



Figure 5. Screenshot of the similarity analysis for the recurrence plots of two synthetic data sequences DS11 and DS42, their similarity corresponds to category 3 (84.6% of similarity).

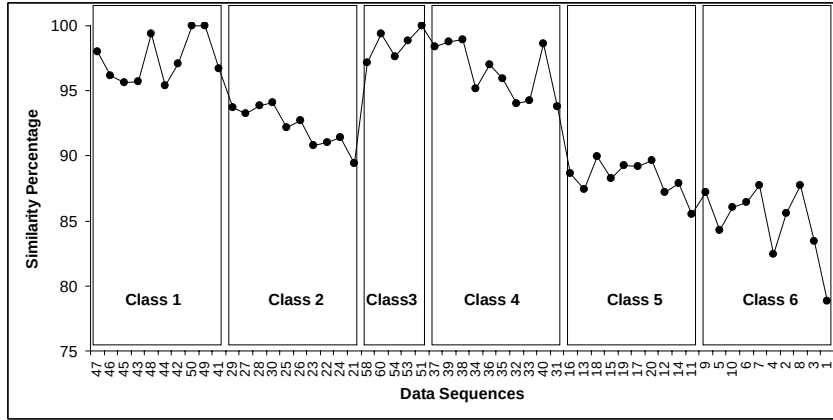


Figure 6. Behavior of the similarity percentage between the recurrence plots for the data sequence DS50 and the rest of the sequences from the experimental set.

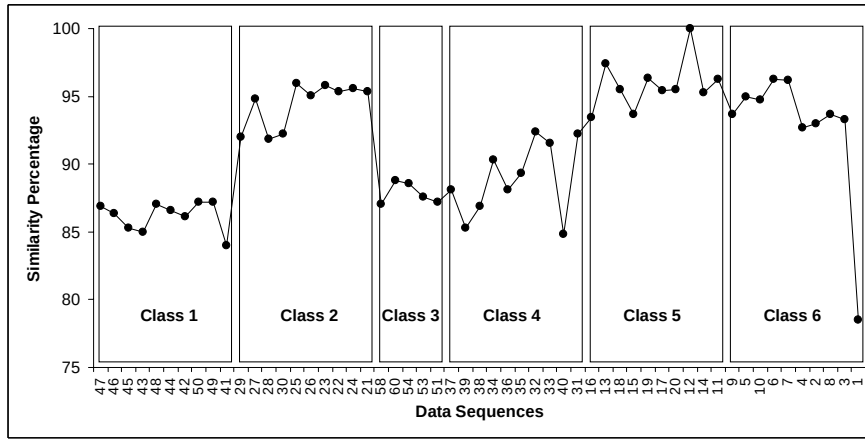


Figure 7. Behavior of the similarity percentage between the recurrence plots for the data sequence DS12 and the rest of the sequences from the experimental set.

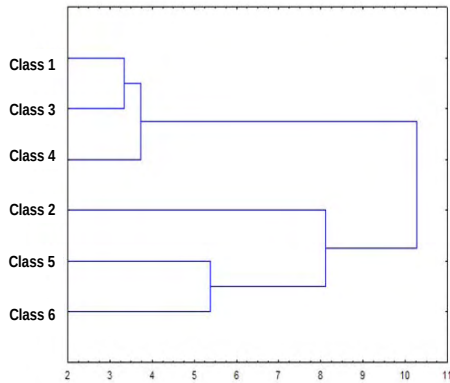


Figure 8. Classification tree derived from the similarity analysis of recurrence plots of data sequences made with RecurrenceVs.

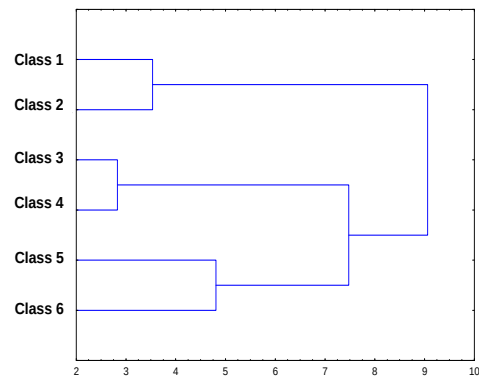


Figure 9. Classification tree derived from the similarity analysis of data sequences made with the DDTW in references [5, 15].

5. Conclusions

A computational technique for the analysis of similarity of data sequences using a new representation with recurrence plot patterns was presented, in this representation hidden patterns that represent the dynamics of a sequence can be visualized, analyzed, and compared; so new classifications based on these patterns that represent relations between dynamics and structure of the sequences can be built. The experiments of validation show expected changes in the way that the families or classes of sequences are grouped by their similarity due to the new representation with these patterns, but at the level of similarity between sequences of the same class the similarity is preserved when is compared with the classification method selected as our reference: DDTW. A current work is the application of RecurrenceVs for the comparison of genomic sequences; also we are working on the study of time series and their dynamics. Finally, new improvements of the functionality of the tool are on the way, and it will be posted in our website for free distribution under the GNU General Public License.

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6. References

- [1] Bautista Thompson, E.F., *Medición de la Predictibilidad de Series de Tiempo: Un Estudio Experimental*, Ph. D. Thesis, Instituto Politécnico Nacional, México DF, 2005.
- [2] S.B. Needleman and C.D. Wunsch, "A General Method Applicable to the Search for Similarities in the Amino Acid Sequences of Two Proteins", *J. Mol. Biol.*, Vol. 48, 1970, pp. 443-453.
- [3] G. Das, D. Gunopulos, and H. Mannila, "Finding Similar Time Series", *Proceedings PKDD'97 LNCS*, Vol. 1263, 1997, pp. 88-100.
- [4] B.K. Yi and C. Faloutsos, "Fast Time Indexing for Arbitrary Lp Norms", *Proceedings of the 26th International Conference on Very Large Databases*, 2000, pp. 385-394.
- [5] E. Keogh and M. Pazzani, "Scaling Up Dynamic Time Warping for Data Mining Applications", *Proceedings of the 6th ACM SIGKDD International Discovery & Data Mining*, 2000, pp. 285-289.
- [6] E. Bautista-Thompson and S. Santos-De la Cruz, "Shape Similarity Index for Time Series Based on Features of Euclidean Distances Histograms", *Proceedings of the 15th International Conference on Computing CIC'06*, 2006, pp. 60-64.
- [7] Golyandina, N., Nekrutkin, V., and Zhiglavsky, A., *Analysis of Time Series Structure SSA and Related Techniques*, Chapman & Hall / CRC, 2001.
- [8] N.E. Huang, Z. Shen, S.R. Long, M.C. Wu, H.H. Shih, Q. Zheng, N.C. Yen, C.C. Tung, and H.H. Liu, "The Empirical Mode Decomposition and the Hilbert Spectrum for Nonlinear and Non-Stationary Time Series Analysis", *Proc. R. Soc. Lond. A*, Vol. 454, 1998, pp. 903-995.
- [9] J.P. Eckmann, S.O. Kamphorst, and D. Ruelle, "Recurrence Plots of Dynamical Systems", *Europhys. Lett.*, Vol. 4. Issue 9, 1987, pp. 973-977.
- [10] VRA Software (by Eugene Kononov), <http://www.myjavaserver.com/~nonlinear/vra/download.html>
- [11] J.P. Zbilut and C.L. Weber, "Embeddings and Delays as Derived from Quantification of Recurrence Plots", *Phys. Lett. A*, Vol. 171, 1992, pp. 199-203.
- [12] Marwan, N., *Encounters with Neighbors: Current Developments of Concepts Based on Recurrence Plots and Their Applications*, Ph. D. Thesis, Postdam University, Postdam, 2003.
- [13] H. Whitney, "Differentiable Manifolds", *Annals of Mathematics*, Vol. 37, 1934, pp. 645-680.
- [14] F. Takens, "Detecting Strange Attractors in Turbulence", *Lecture Notes in Mathematics*, Vol. 898, 1981, pp. 366-381.
- [15] S. Hettich and S. D. Bay, The UCI KDD Archive, <http://kdd.ics.uci.edu>.

Boosting for image interpretation by using natural features

J. Gabriel Avina-Cervantes

M. de J. Estudillo-Ayala
Mario A. Ibarra-Manzano

Sergio Ledesma-Orozco

University of Guanajuato.

Carr. Salamanca-Valle de Santiago km. 3.5 + 1.8km.

Com. de Palo Blanco. Salamanca, Gto., México.

{avina,estudillo,selo,ibarram}@salamanca.ugto.mx

Abstract

In this paper a research in classification of natural images by using ADABOOST (Adapting Boosting) method is presented. This technique is used to identify the nature of the main regions in the image, that is, to identify if they are roads, trees, shades, sky, bushes or others interesting regions; image is previously segmented and each of its regions are represented by a $\mathbf{R}^{12}$ data vector (including features as color, texture and context), in at least 5 classes. The proposed methodology is presented for a multi-class classification problem and for validating our results, performances ratios between ADABOOST and the Support Vector Machines are discussed. This methodology is intent to be applied in medical imagery and in visual based navigation on natural environments; in robot navigation, very good results are obtained even in poorly color saturated images. Finally, the results are described and presented showing that ADABOOST is a reliable classification technique giving slightly better performances than SVM for regions in natural images.

1. Introduction

A great interest on classification of natural images have been manifested by some machine vision researchers in all over the world, because there are so many potential applications in others study fields [4]. For example, in civil engineering and in architecture is very important to classify aerial images in order to identify the best regions on earth where it is possible to developed strong buildings. In medicine, images from ultrasonic devices, electrocardiograms, mastographs, *etc.*, are analyzed to detect some kind of disease or to improve medical diagnosis (cancer detection, cholesterol plaques or heart arrhythmias [10], *etc.*). In music, to classify some songs by musical genre, in the problem of human face detection [13] are some interest-

ing fields where image classification techniques plays a crucial role by helping to improve human visual identification.

In the context of robotics applications, visual based navigation needs an accurate scene categorization or image classification to identify the correct region or zone where the robot could plan a navigable trajectory [8]. In natural images where unstructured data is often the mainly available information; features as color, texture, shape and context are the most useful attributes to identify typical objects or regions in the scene [1]. Thus, image regions as trees, rocks, roads, shadows could be identified in natural images. In mobile robotics, one objective consists in developing an artificial vision systems able to guide the robot into the right direction. The first step to achieve this goal relays on natural image classifications, however it is needed to cope with problems related to illumination and color constancy [9].

Nowadays, fast software and hardware architectures are available to perform image processing algorithms (low-level tasks, *e.g.*, segmentation) accurately and in real time. These embedded architectures make possible to apply useful tools as color texture classification in to the problem of object detection in natural scenes [8] by applying more sophisticated classification techniques as ADABOOST. In this paper, image is segmented by color features with an hybrid thresholding and region growing approach. For region classification, three color features are extracted from segmented regions, these are complemented with seven texture features and two additional contextual features, which are also computed over the same region. Thus, a vector in $\mathbf{R}^{12}$ is used to characterize the regions, which are intended to recognize [8]. To classify object some classical methods have been tested by our team, *e.g.*, K-NN (K- Nearest Neighbors), SVM [7, 11] (Support vector Machines) and Fisher classifiers. In this work, *Adaboost* classifier is efficiently evaluated and compared with SVM in the context of natural image classification in a multi-class approach. Thus, *Adaboost* has found

many potential applications, for example, hand writing recognition [3] and in the context of human face detection [12]. In this last application, deep studies have been developed obtaining recognition ratios between 85% and 90%, processing images (384×288 pixels) in 0.67 fps. Exploiting ADABOOST, this last approach is at least ten times faster than the boost methods reported in [5].

This paper is organized as follows: in Section 1 a short introduction has been given, in the Section 2 a review of ADABOOST classifier is presented, in Section 3 is presented the methodology applied to natural image classification, some results are presented in Section 4 and finally, conclusions are presented in Section 5.

2. ADABOOST classification method

The objective of the ADABOOST algorithm consists in finding at least a function into the hypothesis space that gives us a small error against the distribution D , issued from the labeled training samples. ADABOOST has two differences against other algorithms: (1) ADABOOST adjusts or adapts the hypothesis errors, given by the weak learning algorithms, (2) the limit of the ADABOOST actuation depends only of the actuation of the weak learning model, generated by the distribution function on the learning process.

Adaboost works as follows: in the iteration t , *boosting* algorithm estimates a distribution D_t , for the weak learning model, based on the training samples M . As a result, the weak learning model computes an hypothesis $h_t : \mathbf{X} \rightarrow \mathbf{Y}$ that correctly classifies a subset of the training samples. Moreover, classification error is measured taking into account the distribution D_t . Procedure continues for T iterations, and finally the *boost* method combines the hypothesis $h_1, h_2, \dots, h_T$ in an unique hypothesis H_{final} . Initial distribution D_1 is considered uniform over the training samples M , this is represented as follows:

$$D_1(i) = \frac{1}{m} \quad \forall i \quad (1)$$

where i is an element from the m training samples.

Given the distribution D_t and the learning hypothesis h_t on the iteration t , the next distribution D_{n+1} is computed affecting the weight of the sample i by some factor $Bn \in [0, 1]$ if h_t classifies correctly the input vector $\mathbf{x}_i$; by the way, the weight is maintained unaltered. The weights are again normalized by using the normalizing constant Z_t . In fact, *easy* samples are firstly introduced into the training procedure, these samples are found in the set of samples M correctly classified by many of the *weak* hypothesis which are slightly weightened. It is considered that *hard* samples¹ are

¹ or difficult samples, they could belong to more than one class

often wrongly classified and they are highly weightened. Thus, *Adaboost* algorithm focusses the majority of weights to those hard samples that seems more difficult to classify. *Adaboost* produces an hypothesis with errors $\epsilon_1, \epsilon_2, \dots, \epsilon_T$, where ϵ_t error on the iteration t in the *Adaboost* algorithm is defined as follows,

$$\epsilon_t = Pr_{D_t}[h_t(x_i) \neq y_i]$$

Adaboost algorithm is shown in the next diagram:

1. Compute D_t :

$$D_t = \frac{1}{m}$$

2. for each $t = 1, \dots, T$:

- (a) Given D_t and h_t , compute the density distribution, where z_t is a normalization factor

$$D_{t+1}(i) = \frac{D_t(i)}{Z_t} \cdot \exp(-\alpha_t y_i h_t(x_i)).$$

- (b) Compute the factors

$$\alpha_t = \frac{1}{2} \ln \left(\frac{1 - \epsilon_t}{\epsilon_t} \right) > 0$$

- (c) Find learning hypothesis ("rules")

$$h_t : X \rightarrow \{-1, +1\}$$

- (d) Obtaining the smaller error of ϵ_t in D_t :

$$\epsilon_t = Pr_{D_t}[h_t(x_i) \neq y_i]$$

3. Final hypothesis is:

$$H_{final}(x) = \text{sign} \left(\sum_t \alpha_t h_t(x) \right)$$

3. Object recognition in natural scenes

AdaBoost classifier has a supervised learning process, *i.e.*, user needs to provide spatial data organized in some predetermined classes. The number or nature of classes are defined by the application, in our experiments, natural classes are used, *e.g.*, *Tree*, *Grass*, *Sky*, *Roads*, *Soil* or agricultural terrains. Data vector are obtained from training images, exploiting color, texture and context in each region on the image [8]. For training, images are firstly chromatically corrected [6]. This step is essential, performance of color segmentation and color characterization methods depends on it. Images are segmented by a color segmentation algorithm, exploiting $I_1 I_2 I_3$ color space. Feature $\mathbf{R}^{12}$ data vector is composed by 3 color components (averages color, $I_1 I_2 I_3$), 7 texture components derived from the

Class	Tree	Sky	Grass	Road
Tree	899	2	13	31
Sky	2	406	-	10
Grass	19	-	1017	25
Road	29	4	19	908

Table 1. Data normalized by attribute.

Class	Tree	Sky	Grass	Road
Tree	943	1	-	1
Sky	6	412	-	-
Grass	-	-	1044	17
Road	-	2	25	933

Table 2. Data normalized by class.

“histograms of sums and differences” [2] (extracting the most powerful and uncorrelated texture features) and 2 contextual features. Contextual features take into account spatial distribution of the regions on the image, *i.e.*, class *Sky* is highly probably to be found in the upper side in the image and by the contrary, class *Road* is highly probably to be found in the lower center part of the image. Data vector in $\mathbf{R}^{12}$ must be normalized, before training or recognition steps. This normalization can be done by class or by attributes. Obviously, normalization by attributes is globally most stable and efficient. The normalization procedure is done obtaining the maximum absolute value of each feature on each class. Each element composing the class is then divided by the maximal data vector. Normalized data is then exploited by the ADABOOST algorithm to obtain de recognition rules, a further recognition step with unknown data vectors is used to identify the nature of each region. In the next section some experimental results are discussed.

4. Experimental results

In this work a set of 50 images were selected in the learning stage and 150 images were used for the evaluation stage. To validate learning procedure, a self consistency test was developed, because some times noise in database is presented in the supervised class definition, affecting strongly recognition ratio. For self consistence purpose in database, the following tables are presented. Table 1 shows that from a set of 945 vectors representing the class *Tree*, 2 are classified as *Sky*, 13 as *Grass* and 31 as *Road*; besides from the 418 vectors representing class *Sky*, 2 are classified as *Tree* and 10 as *Road*; and from the 1061 vectors belonging to *Grass*, 19 are wrongly classified as *Tree* and 25 as *Road*; from the 960 vectors representing *Road*, 29 are classified as *Tree*, 4 as *sky*, 19 as *Grass*, computing with the statistics a self cross validation ratio of 95.6%.

Table 2 shows that from the 945 vectors represent-

DB	DB1		DB2		DB3	
Class	C	A	C	A	C	A
Tree	100	91	92	100	99	95
Sky	100	90	-	-	99	97
Grass	97	91	100	76	98	96
Road	100	96	100	94	97	95
Score	99.2	92.5	97.2	90	98.4	95.6

Table 3. Percentage of normalized data.

ing *Tree*, only 2 vectors are wrongly classified, 1 as *Sky* and the other vector as *Road*; from the 418 vectors representing *Sky*, 6 are wrongly classified as *Tree*; from the 1061 vectors corresponding to *Grass* only 17 are wrongly classified as *Road* and finally the 960 vectors representing *Road*, 2 are wrongly classified as *Sky*, 25 as *Grass*, obtaining a self cross validation ratio of 98.5%. This test only shows the self-consistence of the database but it is not relevant for detecting unknown regions which is highly dependant of the discriminant used features.

In the Table 3 are compared the results obtained normalizing data by attribute (A) and by class (C). Note that in self validation, the best results are obtained normalizing data by the class, the worst results are given due to a very few amount of data available for the class, *e.g.*, *Soil*. Classification ratio is higher with DB1 normalized by class with 99.2% and the lower value was for DB2 with 97.2%. Note that average classification is above of 90%, and results obtained from normalized data by class are better than results normalizing by attribute. However, for recognition stage with the evaluation images, *i.e.*, images non included in the learning process, data normalization by attribute is more stable and less sensitives to noise in database. To show the reliability of this approach, 150 natural images were tested in the evaluation phase. Each result takes into account a set of 4 color images. In (a) original image, in (b) segmented images, output images (c) and (d) are obtained by classification steps the former with SVM and the later with *AdaBoost*.

SVM was trained with classes *Tree*, *Sky*, *Grass*, *Road*, *Soil*, *Fuzzy* and *Rock*. The class *Fuzzy* refers to the region in the image representing a mixture between the class *Sky* and *Tree*, *i.e.*, tree with sky. Classes with very few elements some times is not good practice to take them into account for classification procedures, *e.g.*, *Fuzzy* and *Rock*. In Figure 1 are shown the mainly difference in classification between SVM and *AdaBoost*, see regions 4 and 6 which are classified with SVM as *Soil* and with *AdaBoost* as *Tree*. Note in this case, that some *bushes* could be classified as *Tree* or *Soil*. Another difference is represented in the regions 11 and 12 which are detected as *Fuzzy* with SVM and as *Tree* with *AdaBoost*; indeed these differences in this case are insignificant because tree and tree-sky are almost

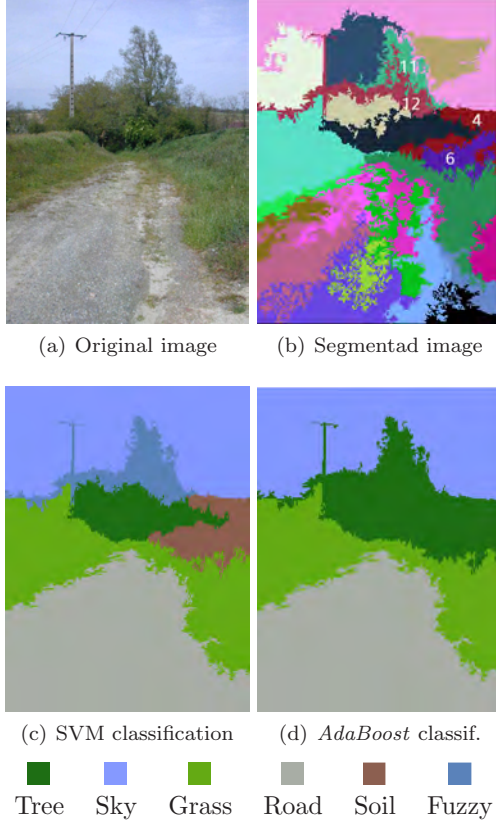


Figure 1. Image with a non marked road

the same. *AdaBoost* was not trained with the class *Fuzzy* because the number of samples for this class is not reliable and the method identified more elements as *Tree* than *Sky* in these situations. In Figure 2 are observed 5 differences in classification. These differences are found in regions 8, 11, 13, 14, and 16 which are marked on the image (b), with SVM regions 8, 11, 13 and 16 are classified as *Fuzzy* and with *AdaBoost* they are classified as *Tree*, for the region 14 SVM could not find a class while *AdaBoost* determines the nature of this class as *Tree*. Figure 3 shows only one difference in the region 12 where SVM determines its nature as *Tree* and *AdaBoost* as *Road*. From the original image 3(a) some parts of the road are merged with the grass which confuses the recognition methods.

In figure 4, 9 regions are shown where both classifiers presented some inconsistencies. Regions 23, 39 and 52 are not detected by SVM, however *AdaBoost* classified them respectively as *Grass*, *Road* and *Grass*; the regions 3 and 17 are classified as *Soil* (agricultural terrain) by SVM, and by *AdaBoost* these same regions are respectively recognized as *Grass* and *Tree*. Moreover, regions 32 and 37 are respectively recognized as the class *Fuzzy* by SVM and the same regions are detected as *Trees* by *AdaBoost*; region 34 with SVM is

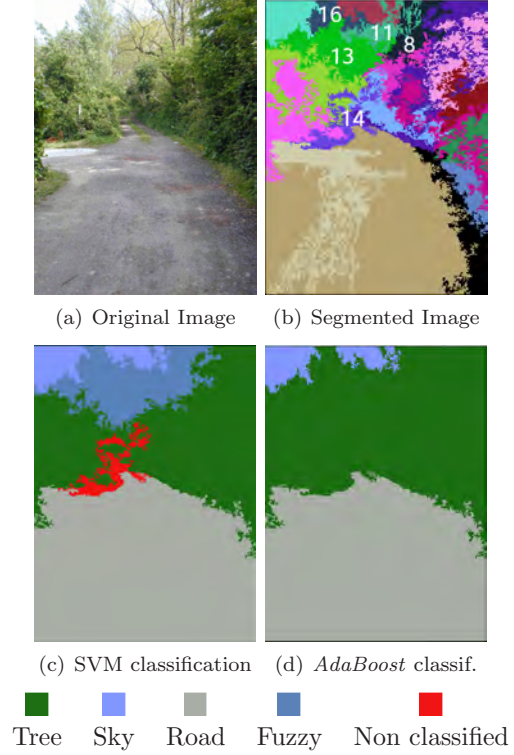


Figure 2. Image with a bifurcated road.

<i>Image/Classifier</i>	<i>SVM</i>	<i>AdaBoost</i>
Fig. 1 Road and lightpole	87.5%	91.6%
Fig. 2 Bifurcated road	85.0%	90.0%
Fig. 3 Gravel road	94.1%	94.1%
Fig. 4 Pine trees line	83.3%	92.5%
Fig. 5 Canal road.	83.3%	83.3%
Average	86.6%	90.4%

Table 4. *AdaBoost* vs. SVM

detected as *Grass* by SVM and by *AdaBoost* is classified as *Tree* and finally, region 53 is classified as *Rock* by SVM and by *AdaBoost* this region is identified as *Tree*. In figure 5, some differences are shown mainly between the class *Fuzzy* (by SVM) identified as *Tree* by *AdaBoost*. Note that in image 5(d) all trees are correctly identified. Another important differences are obtained when canal and road are very close each other and color tonalities of reflected trees on water are confusing for classification methods; SVM classifies them as *Grass* and *Adaboost* as *Trees*.

In Table 4 are summarized and compared the overall results between SVM and *Adaboost*, where are observed two similar recognition rates and results between SVM and *Adaboost*, in the other situations *AdaBoost* produces better recognition results in natural images. Finally, classification results are presented by the confu-

Class	Tree	Sky	Grass	Road	Soil	Rock	Fuzzy	NC	EO
Tree	67	-	-	-	-	-	2	-	2
Sky	-	14	-	-	-	-	-	-	0
Grass	2	-	27	6	-	-	-	-	8
Road	-	-	-	45	-	-	-	-	0
Soil	1	-	-	-	-	-	-	-	1
Rock	-	-	-	-	-	-	-	-	0
Fuzzy	7	-	-	-	-	-	-	-	7
NC	-	-	-	-	-	-	-	-	0
EC	10	0	0	6	0	0	2	0	

Table 5. Classification with *AdaBoost*

Class	Tree	Sky	Grass	Road	Soil	Rock	Fuzzy	NC	EO
Tree	46	-	2	-	2	4	7	6	21
Sky	-	14	-	-	-	-	-	-	0
Grass	1	-	28	1	2	-	-	3	7
Road	-	1	1	42	-	-	-	1	3
Soil	1	-	-	-	1	-	-	-	1
Rock	-	-	-	-	-	-	-	-	0
Fuzzy	-	-	-	-	-	-	8	-	0
NC	-	-	-	-	-	-	-	-	0
EC	2	1	3	1	4	4	7	10	

Table 6. Classification with SVM

sion matrices shown in Tables 5 and 6 for SVM and *Adaboost*. Overall classification ratios give some advantages to *AdaBoost*, which has an efficacy of 89.4% versus 81.2% for SVM in this application. These tables shown some representative examples where some most important inconsistencies between both studied methods are detected and analyzed. These results shown advantages by using ADABOOST for natural image classification; presenting only a natural and obvious limitations, ADABOOST is strongly affected by noise in databases and classes with a few amount of elements, for this reason class *Fuzzy* is not recommended for boosting applications.

5. Conclusions

The classification method *AdaBoost* show a better performance than SVM in application in natural scenes. Recognition ratio of the 99.2% obtained on classification of data in the same database (self consistency) and almost a recognition ratio of 90% for images in the evaluation database. On learning data, better results are obtained when data is normalized by class instead of attributes. However, for the robust applications, as in special evaluation database in natural images, the best results are obtained normalizing data by attributes. Besides, recognition is more stable, reaching a recognition rate of almost 90% with ADABOOST. The mainly differences in performance between SVM and *Adaboost* are mainly derived from

ambiguous regions which are even complex to classify for the human, *e.g.*, far or tree-sky regions (Fuzzy region for SVM, *Tree* for *Adaboost*). Overall image recognition with ADABOOST is done in less than 0.90 s for images of 400x300 pixels.

References

- [1] I. Autio and T. Elomaa. Flexible view recognition for indoor navigation based on gabor filters and support vector machines. *Pattern recognition*, 36(12):2769–2799, December 2003.
- [2] G. Aviña-Cervantes and M. Devy. Dtection et suivi de chemins pour la navigation d’un robot en milieu naturel. In *Journes Francophones des Jeunes Chercheurs en Vision par Ordinateur (ORASIS’2003)*, volume 1, pages 403–410, Gérardmer (France), 19-23 Mai 2003.
- [3] D. Bargerion, P. Viola, and P. Simard. Boosting-based transductive learning for text detection. In *Proceedings of the 2005 Eight International Conference on Document Analysis And Recognition*, volume 2, pages 1166–1171, 2005.
- [4] M. Celenk. Analysis of color images of natural scenes. *Journal of Electronic Imaging (JEI)*, 4(4):382–396, October 1995.
- [5] Y. Freund and R. E. Shapire. Experiments with a new boosting algorithm. In *International Conference on Machine Learning*, pages 148–156, 1996.
- [6] F. Gasparini and R. Schettini. Color balancing of digital photos using simple image statistics. *Pattern Recognition*, 37(6):1201–1217, June 2004.

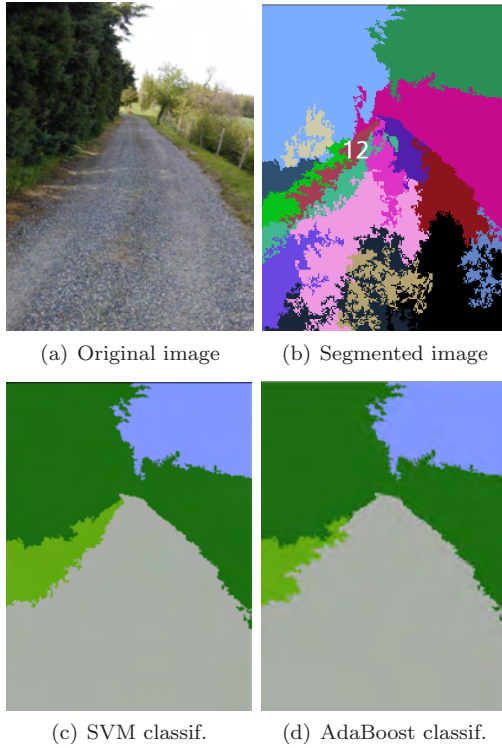


Figure 3. Gravel road image

- [7] S. Li, J. T. Kwok, H. Zhu, and Y. Wang. Texture classification using the support vector machines. *Pattern recognition*, 36(12):2883–2893, December 2003.
- [8] D. Mateus, J. G. Avina-Cervantes, and M. Devy. Robot visual navigation in semi-structured outdoor environments. In *IEEE International Conference on Robotics and Automation (ICRA)*, Barcelona, Spain, April 18-22 2005.
- [9] P. Paysan. Stereo vision based vehicle classification using support vector machines. Master's thesis, University of Applied Sciences Fachhochschule Esslingen, Massachusetts, USA, febrero 2004.
- [10] O. Pujol, M. Rosales¹, P. Radeval, and E. Nofrerias-Fernandez. Intravascular ultrasound images vessel characterization using adaboost. *Lecture Notes in Computer Science, Springer Berlin/Heidelberg*, 2003.
- [11] Y. Qi, D. Doermann, and D. DeMenthon. Hybrid independent component analysis and support vector machine learning scheme for face detection. In *IEEE International Conference on Acoustics, Speech and Signal Processing, ICASSP*, volume 3, pages 1481–1484, Salt Lake City, Utah, May 2001.
- [12] P. Viola and M. Jones. Rapid object detection using a boosted cascade of simple features. In *IEEE Conference on Computer Vision and Pattern Recognition*, volume 1, pages 511–518, 2001.
- [13] P. Viola and M. Jones. Fast and robust classification using asymmetric adaboost and a detector cascade. In *Neural Information Processing System 14*, MIT Press, Cambridge, 2002.

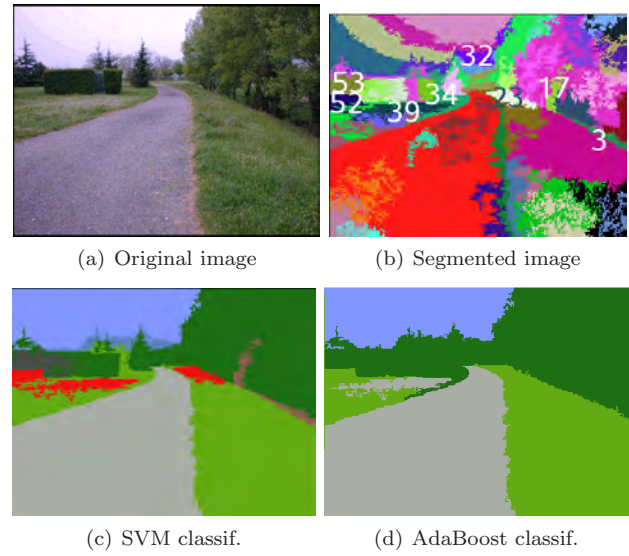


Figure 4. Pines tree road

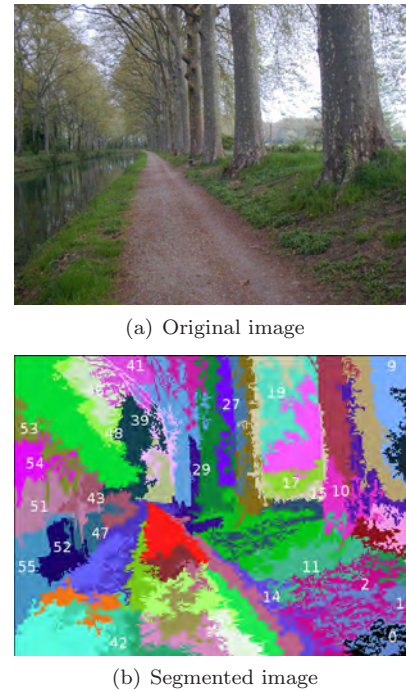


Figure 5. Canal-road image

Face Recognition Based on the Phase Spectrum of Local Normalized Image

Jesus Olivares-Mercado^{*†} Kazuhiro Hotta[†] Haruhisa Takahashi[†] Hector Perez-Meana^{*}
Gabriel Sanchez-Perez^{*}

^{*}ESIME Culhuacan, Instituto Politecnico Nacional, Av. Santa Ana No. 1000,
Col. San Francisco Culhuacan, 04430 Mexico D.F. Mexico

[†]The University of Electro-Communications, 1-5-1 Chofugaoka, Chofu,
Tokyo 182-8585, Japan

Abstract

This paper proposes a robust faces recognition method based on the Phase Spectrum Features of the local normalized image. The Principal Components Analysis (PCA) and the Support Vector Machine (SVM) are used in the classification stage. We evaluate how the proposed method is robust to illumination, occlusion and expressions using "AR Face Database", which includes the face images of 109 subjects (60 males and 49 females) under illumination changes, expression changes and partial occlusion. The proposed method provides results with a correct recognition rate more than 95.5%.

1. Introduction

The development of security systems based on biometric features has been a topic of active research during the last three decades, because the recognition of the people identity to access control is a fundamental aspects in these days. The terrorist attacks happened during the last decade have demonstrated that it is indispensable to have reliable security systems in offices, banks, airports, etc.; increasing in such way the necessity to develop more reliable methods to people recognition. The biometrics systems consist of a group of automated methods for recognition or verification of people identity using physical characteristics or personal behavior of the person under analysis [1]. In particular the face recognition has been a topic of active research because the face is the most direct way to recognize the people. In addition, the data acquisition of this method consists in taking a picture, doing it one of the biometric methods with larger acceptance among the users. Various kinds of face recognition methods have been proposed [2, 3]. In recently years, faces are recognized with high accuracy recognition.

However, partial occlusion, illumination variations and expression decrease the accuracy drastically.

The recognition is a very complex activity of the human brain. For example, we can recognize hundred of faces learned throughout our life and to identify familiar faces at the first sight, even after several years of separation, with relative easy. However it is not a simple task for a computer. For instance, recently proposed face recognition systems, achieve a recognition rate of about 90% when the the face in the image is not rotated or the rotation is relatively low [4]. Although this recognition rate is good enough for several practical applications, it may be not large enough for applications where the security should be extreme; such that we cannot tolerate a high erroneous recognition rate. In particular, illumination, occlusion, facial expressions are big obstacles in the practical environment. This paper proposes a face recognition algorithm that is able of achieving an erroneous recognition rate below 5% with this characteristics.

In recent years, the robust face recognition method using phase spectrum has been proposed [5]. However, they extracted phase information from global image. Global features are influenced easily by illumination, occlusion and expression [6, 7]. Therefore, we use the device for emphasizing the local features of a face image before extracting phase spectrum. Concretely, the norm of local region is normalized in advance. This process is effective for face recognition. When the part of a face is occluded by something such as sunglasses or scarf, global similarity between images is much influenced. In local normalized image, the similarities of some local regions are influenced by occlusion but the similarities of almost local regions are not influenced. Therefore, the local normalized image is robust to partial variations. After normalizing the norm of local region, we extract the phase information in the frequency domain representation. This method is very useful because the

phase spectrum of an image retains the most of the intelligibility of this. Furthermore, we can increase the recognition accuracy using the PCA to obtain the main characteristics of the training faces. The features obtained by PCA are fed into one-vs-all SVM.

The proposed method is evaluated using “AR Face Database” [8]. The data base includes the face images with partial occlusion, illumination variations and expression. The phase spectrum of local normalizes image outperforms conventional eigenphase method [5]. In particular the robustness to partial occlusion is also improved. This shows the effectiveness of the proposed approach.

In section 2, the details of proposed system are explained. Experimental results are shown in section 3. Finally, section 4 describes conclusions.

2 Proposed System

This section provides a detailed description of the proposed face verification algorithm which consists in two moduls of four stages each one. Figure 1 shows the block diagram of proposed algorithm.

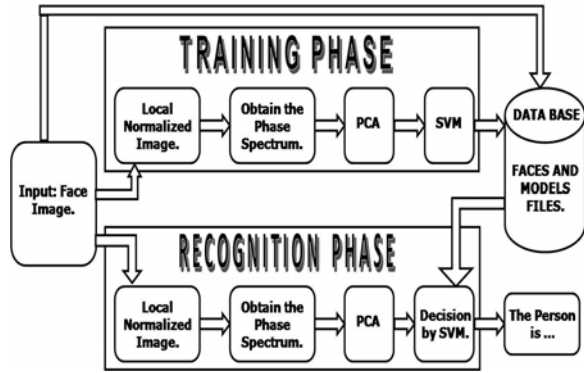


Figure 1. Proposed face recognition algorithm.

Firstly we normalize the norm of the local region of the image. The phase spectrum is extracted from the local normalized image, after that, the Principal Components Analysis (PCA) [9, 10] is applied to the phase spectrum to obtain a dominant feature of the faces. Next, the features in principal components space are fed into classifier based on Support Vector Machine (SVM) [11, 12]. In the final step of training, one-vs-all SVM is used to classify multi-classes. In the final step of test, PCA features of phase spectrum of a test image are fed into all SVMs and the test image is classified to the class given maximum likelihood.

2.1 Feature Extraction Stage.

The feature extraction plays a very important role in the any pattern recognition system. To this end, the proposed algorithm uses the Phase Spectrum of Local Normalized Image. To obtain local normalized image, the norm of local regions of $M \times M$ pixels where $M = 3, M = 6$ and $M = all$ are normalized to 1. Figure 2 shows this process. By normalizing the norm of local regions, it becomes robust to partial variation such as illumination changes or occlusion. We note local region as $I(x)$ where x is the center position of local region, the local norm is normalized as:

$$I(x)' = \frac{I(x)}{\|I(x)\|} \quad (1)$$

where $\|I(x)\|$ is the norm of the local region. All the training images must be normalized before training. In the test, we need to apply the same process before extracting the phase spectrum. In this paper we consider the local normalized image to improve the accurate classification and robustness to partial variations.

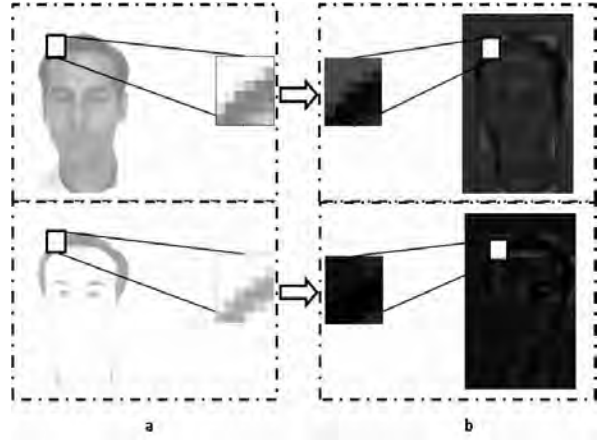


Figure 2. a) Original image. b) Image after apply the normalization.

After normalizing the norm of local region, we extract the phase spectrum. This can be computed through of a Fourier Transform which are given by:

$$F(u) = |F(u) \exp^{j\phi(u)}| \quad (2)$$

where

$$|F(u)| = [R^2(u) + I^2(u)]^{\frac{1}{2}} \quad (3)$$

is the magnitude, and

$$\phi(u) = \arctan \left[\frac{I(u)}{R(u)} \right] \quad (4)$$

is the phase, is needed extract the phase spectrum of every images of both training and testing. Oppenheim et. al [13, 14] have shown that phase information of an image retains the most of the intelligibility of an image. This is also demonstrated by Oppenheim's experiment shown in Figure 3.

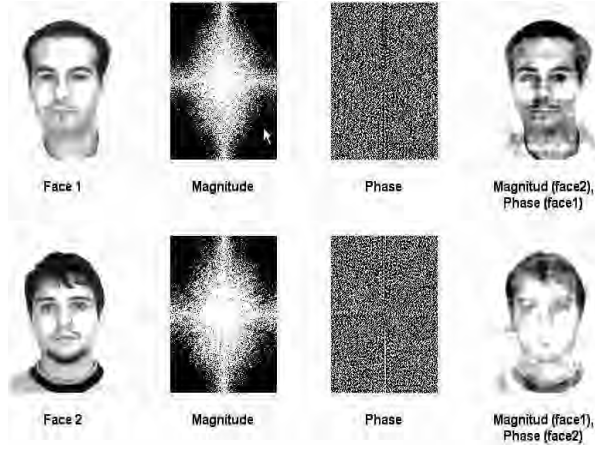


Figure 3. Oppenheim's experiment.

Those two face images are obtained from the “AR Face Database”[10]. We apply the fourier transform to these two images and obtain magnitude and the phase. We combine the phase of the image 1 with the magnitude of the image 2 and the phase of the image 2 with the magnitude of the image 1. Thus we see that the component that provides more information about the image has been the phase. This shows that phase information is more effective than magnitude.

2.2 Classification stage.

To perform the face classification task, a PCA is used to obtain the main characteristics of the faces training, Figure 4 shows the process:

Image 1, Image 2...Image N in Figure 4 are the phase spectrum of the training faces. In training phase, basis vectors are obtained by PCA. In the testing phase, the basis vectors obtained in training phase are used to extract the features for SVM.

After extracting the features by PCA, SVM is used to classify a test face image. Figure 5 shows classification phase. S is the feature vector of the person to recognize.

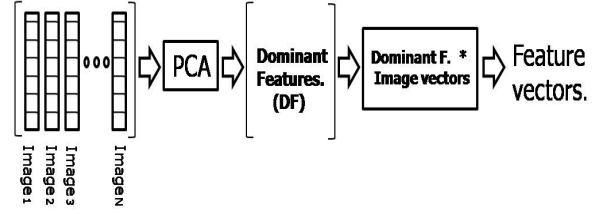


Figure 4. Scheme of feature extraction by PCA.

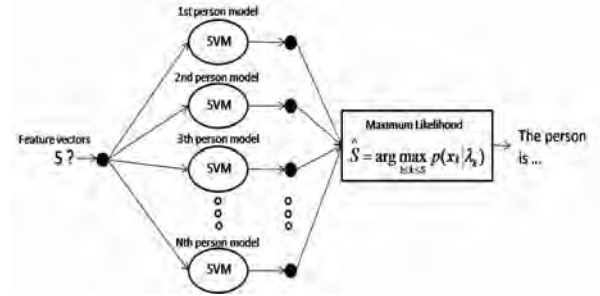


Figure 5. Classification phase by SVM.

S is applied to all one-vs-all SVMs. The class given the Maximum Likelihood is used as the person's identity, the equation to obtain the Maximum Likelihood is as follows:

$$\hat{S} = \arg \max_{1 \leq k \leq S} P(\lambda_k | x) \quad (5)$$

where $\hat{S}$ is the winner and thus reveals the person's identity to whom this picture was assigned, x is the column vector of the image to analyze and λ_k is the SVM model of the person k .

3 Evaluation Results

The evaluation of proposed system was carried out by computer simulations using “AR Face Database”, created by Purdue University, USA. The database includes the face images of 109 subjects (60 males and 49 females) under illumination changes, expression changes and partial occlusion. The images size is 288×384 pixels. In this paper, we resize the image of a 48×36 pixels for processing. Each subject has 39 images, in the followings experiments, 6 images per subject are used in training and the remaining 33 images per subject are used in test.

The algorithm assumes that the gray level of picture background is constant. First the algorithm estimates the normalized image and extract the phase spectrum. Once the normalized image and phase spectrum have been estimated, the image is converted in a column vector and so on with all

Table 1. Accuracy on face recognition

	M=3	M=6	M=All	Normal
Training images 1	95.8	94.6	93.2	93.2
Training images 2	79.5	78.3	75	75

the training images to implement the process to obtain the features vectors as shown in Fig. 4.

Next, the feature vectors of training images are applied to a SVM to obtain the model of the each class, these models are used in the classification stage, where the input of each one is the feature vector of the face to classify and the output is a probability, and the class to which it belongs is obtained by selecting maximum likelihood as shown in Fig. 5.

In this work we take 3 different variants in the window size to make the local normalized image, and comparing these with the method that only uses the phase spectrum of standard image. We try $M = 3$, $M = 6$ and $M = all$. $M = all$ means that the window is equal to the size of the image. The local norm is normalized with non-overlap manner. We take another option which is “Normal” as shown in Table 1, “Normal” means that the image will be processed without being normalized. Namely, this is nearly same as eigenphase approach [5].

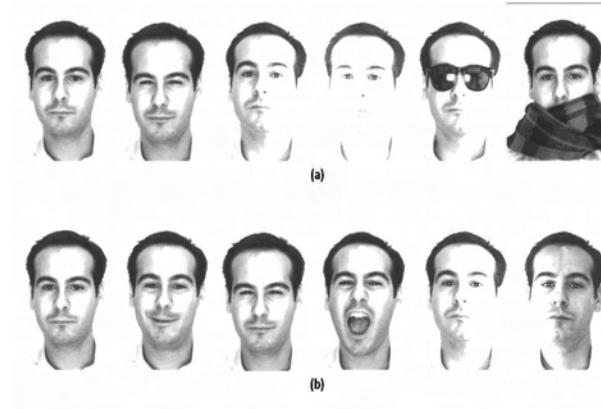


Figure 6. a) Training images 1. b) Training images 2.

The table 1 shows the results with 2 sets of images of training, in the first set of images (training images 1) takes 6 images in which illumination variations, expressions, sunglasses and scarf are included. The Figure 6(a) shows the example of training images 1. The recognition accuracy of $M = 3$ achieves 95.8%, our method outperforms the eigenphase method. The improved rate is 2.6%.

The second set of images (training images 2) takes 6 images which only illumination variations and expressions are

included. Therefore the accuracy to the second set shows the robustness to partial occlusion. Figure 6(b) shows the example of the second training set. Table 1 and figure 8 shows accuracy to training image 2. The recognition accuracy of $M = 3$ achieves 79.5%. The improved rate is 4.5%. The difference from eigenphase method becomes larger than that in the training images 1. This shows that the local normalized image has the robustness to partial variations.

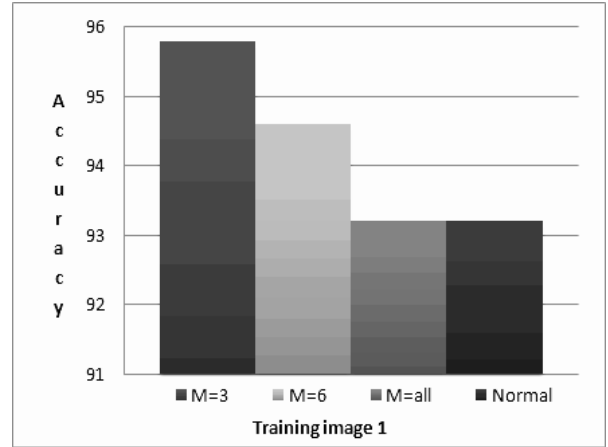


Figure 7. Accuracy with the Training images 1.

Simulation results show that proposed algorithm performs fairly well in comparison with other previously proposed methods [4, 5], even with faces that present different illumination, expression and partial occlusion. Although the part of a face is not visible because of the accessories such as sunglasses and scarf, the proposed method achieves high accuracy.

4 Conclusions

This paper proposed a face recognition algorithm based on the *Phase Spectrum Features of the local normalized image*. PCA is used for dominant feature extraction and the SVM is used to perform the recognition task. Evaluation results shows that the proposed system achieves a accuracy between 93.2% in the worst case and 95.8% at best when we use the training image 1. We obtain a accuracy between 75% in the worst case and 79.5% at best when we use the training image 2 in which face images with partial occlusion are not included.

Hence, we can see that the proposed system improved about 2% to 4% by taking the normalization of local part of an image. Although the database includes the face images of 109 subjects with various variations such as illumination

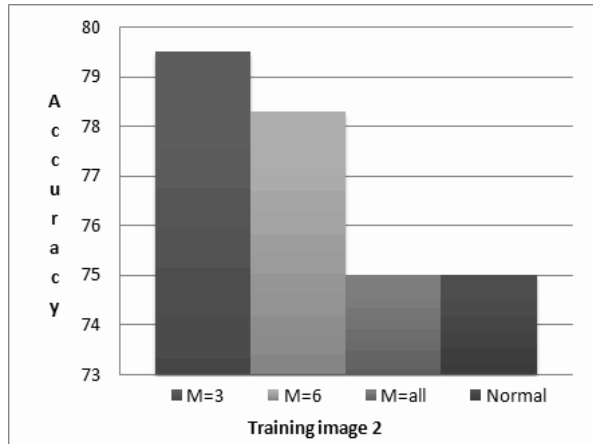


Figure 8. Accuracy with the Training images 2.

changes, facial expressions and partial occlusion, high accuracy is obtained. This is the main contribution in this work.

Acknowledgments.

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References

[1] P. Reid, BIOMETRICS for Networks Security, New Jersey: Prentice Hall, 2004, pp 3-7.

[2] R.Chellappa, C.L.Wilson, and S.Sirohey, "Human and machine recognition of faces: A survey," Proceedings of the IEEE, vol. 83, no. 5, pp. 705740, 1995.

[3] W.Zhao, R.Chellappa, P.J.Phillips, and A.Rosenfeld, "Face recognition: A literature survey," ACM Computing Surveys, vol. 35, no. 4, pp. 399458, 2003.

[4] G. Aguilar-Torres, G. Sanchez-Perez, M. Nakano-Miyatake and H. Prez-Meana, "Face Recognition Algorithm Using the Discrete Gabor Transform," coniele-comp, p. 35, 2007

[5] Marios Savvides, B.V.K. Vijaya Kumar and P.K. Khosla, "Eigenphases vs. Eigenfaces," Proceedings of the 17th International Conference on Pattern Recognition (ICPR04), vol-3, August 2004.

[6] K. Hotta, "Robust face recognition under partial occlusion based on support vector machine with local Gaussian summation kernel." Image an Vision Computing, 2008 (in press).

[7] A.M. Martinez, "Recognizing imprecisely localized partially occluded, an expression variant fece from a single sample per class." IEEE Trans. on Pattern Analysis and Machine Intelligence, Vol. 24. No. 6. 748 - 763, 2002.

[8] A.M.Martinez and R. Benavente, "The AR face database," CVC Technical Report #24, June 1998.

[9] Lindsay I Smith. "A tutorial on Principal Components Analysis." February 26, 2002

[10] C.M. Bishop, "Pattern recognition and machine learning.", springer, 2006.

[11] V.N.Vapnik, "Statistical Learning Theory." John Eiley & sons. 1998.

[12] Chih-Chung Chang and Chih-Jen Lin, LIBSVM : a library for support vector machines, 2001. Software available at <http://www.csie.ntu.edu.tw/~cjlin/libsvm>

[13] M. H. Hayes, J. S. Lim and A. V. Oppenheim. "Signal Reconstruction from Phase or Magnitude." IEEE Trans. Acoust., Signal Processing, vol. ASSP-28, pp. 672-680, Dec. 1980.

[14] A. V. Oppenheim and J. S. Lim. "The importance of phase in signals." Proc. IEEE, vol 69, No5, pp. 529-541, May 1981.

Neural Networks

Analysis of Time Series with Artificial Neural Networks

Raymundo A. González-Grimaldo

Juan C. Cuevas-Tello

Engineering Faculty, Autonomous University of San Luis Potosí, México

tragg85@gmail.com, cuevas@uaslp.mx

Abstract

This paper presents the study of time series in gravitational lensing to solve the time delay problem in astrophysics. The time series are irregularly sampled and noisy. There are several methods to estimate the time delay between this kind of time series, and this paper proposes a new method based on artificial neural networks, in particular, General Regression Neural Networks (GRNN), which is based on Radial Basis Function (RBF) networks. We also compare other typical artificial neural network architectures, where the learning time of GRNN is better. We analyze artificial data used in the literature to compare the performance of the new method against state-of-the-art methods. Some statistics are presented to study the significance of results.

Keywords: Neural Networks, Radial Basis Functions, General Regression Neural Networks, Time Series

1. Introduction

A time series is a set of observations that are ordered in time, usually sampled with regular intervals. When the intervals are of different size (unevenly), one obtains an irregularly sampled time series; this time series is related to the problem of missing information [1]. The time series analysis allows to obtain information about the source and the underlying problem related to the observations. In other words, one can perform extrapolation and predict the behavior of the time series, and also the interpolation by knowing the information where there are not observations at that time. Time series are present in many areas including physics, biology, finance, medicine and climate.

The time delay estimation problem in gravitational lensing is a problem that involves irregularly sampled time series. Gravitational lensing is a phenomenon that occurs when the light coming from a distant light source is deflected by a massive object (e.g., a galaxy). This phenomenon provokes that the observed images present distortions,

appearing as multiple images from a single source. The gravitational lensing was predicted by the General Relativity Theory of Einstein [9, 15]. The data gathered from a gravitational lens are irregularly sampled due to climate, availability of equipment, among other factors. Moreover, the observed data are noisy.

Our specific problem is to find time delay between pairs of time series, with this one can measure the dark matter and answer many cosmological questions. Therefore, it is a concern of current research work in astrophysics.

To solve this problem, one starts manipulating time series with the above features, and trying to find a model that describes the time series. This is a difficult task since it is impossible to find the exact model that describes an irregularly sampled time series with noise. Sometimes it is possible to obtain a model that describes the behavior of certain time series through laws from physics. But, it is not possible or it does not exist a model to describe a real phenomenon from observed data [7].

This paper presents a study of the above time series through Artificial Neural Networks (ANN), according to the best of our knowledge, this is the first approximation using artificial neural networks. Methods to deal with the time delay problem are developed in the astrophysics literature [10, 12, 13, 8]. Two main methods have been developed in computer science [3, 5], and they employ Kernel Methods and Bayesian Estimation, respectively.

Nevertheless, there are several ANN architectures to analyze time series, where the most used is backpropagation [4]. These architectures are used mainly to model the tendency in the future. We use ANN to match time series, as a regression problem. One of the main problems with backpropagation is the learning time, including setting it up (hidden layers, number of neurons). Therefore, here we also study Radial Basis Functions (RBF) [6] and General Regression Neural Networks (GRNN) [16], where the learning time is faster than backpropagation.

The main contribution of this paper is that we propose a new method to estimate time delays based on GRNN, this method employs cross validation to estimate the parameter

spread. We compare the results on artificial data [3], where the true time delay is known and state-of-the-art methods have been tested. In the comparison, some statistics are used to measure the significance of results.

The remainder of this article is as follows: §2 presents the description of a RBF and a GRNN and the a study of a single time series with different ANN architectures. In §3, we define the application area, gravitational lensing, and the time delay problem and the data that were used in the experiment setup. §4 introduces our method. Finally at the end comes results and conclusions.

2. Artificial Neural Networks (ANN)

This section describes mainly two ANN: RBF and GRNN. We also analyze a single time series, which is irregularly sampled. We compare backpropagation, RBF and GRNN to perform a regression task.

2.1. Radial Basis Functions (RBF)

A neural network RBF has basically three layers (see Fig. 1a) [6, 7], however, some authors refer to only two layers because the first layer, inputs, is not counted. The second layer, hidden layer, is known as radial basis layer or kernel layer [14]; it applies a non linear transformation from input layer to the hidden layer. Finally, the output layer is a linear transformation from the hidden layer to the output layer.

The major contribution to RBF networks is thanks to Moddy, Darken, Renals, Poggio and Girossi [7]. The main objective was to achieve a neural network with fast training to work in real time systems.

Different to backpropagation (also known as Multi Layer Perceptron, MLP), RBF networks have activation models which are different to the hidden layer, compared with the output layer. In other words, the hidden layer has radial functions, which are non linear, and the output layer has linear functions [6, 7]. Another difference with backpropagation is that between the input and hidden layer there are not weights.

A RBF is defined as a network with d inputs, m neurons in the hidden layer and c outputs, where the activation of the output neurons for an input pattern n , $\vec{X}(n) = (x(n)_1, x(n)_2, \dots, x(n)_d)$, is given by [7]

$$y_k(n) = \sum_{i=1}^m w_{ik} \phi_i(n) + w_k \phi_0 \quad k = 1, 2, \dots, c \quad (1)$$

where w_{ik} is the weight between the hidden neuron i and the output neuron k , and $\phi_i(n)$ are the activations for the input

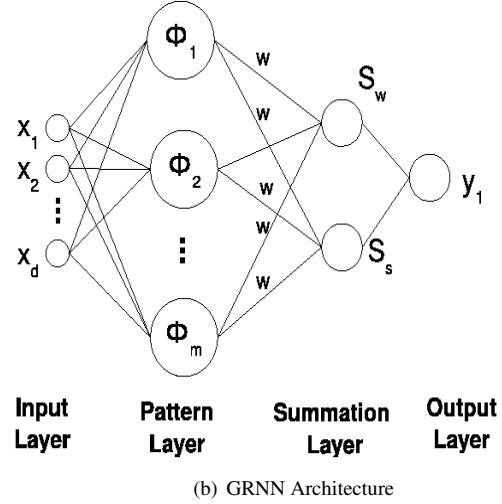
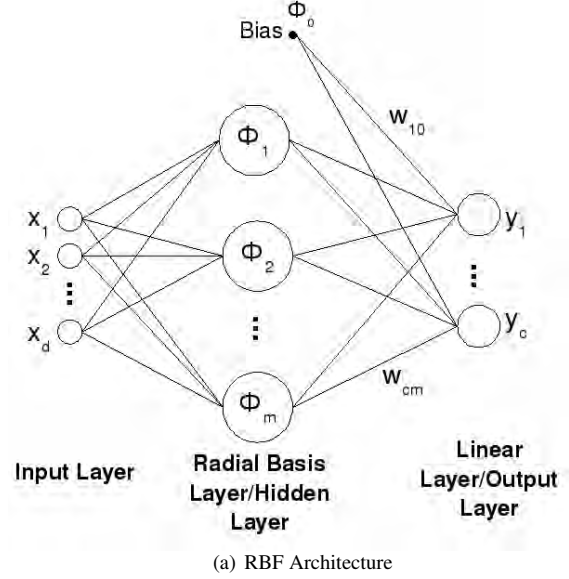


Figure 1. Artificial Neural Network Architectures

pattern $\vec{X}(n)$. The functions ϕ_i represent the radial functions, which give the name to RBF networks. These functions may have different forms including Gaussian, multi-quadratic and inverse multicquadratic. However, Gaussian functions are the most used in the literature, known as Parzen-Rosenblatt density estimator or Parzen window [6]. Consequently $\phi_i(n)$ is defined as follows

$$\phi_i(n) = \exp \left(\frac{- \|\vec{X}(n) - \mu_i\|^2}{2\sigma_i^2} \right) \quad (2)$$

where $\|\vec{X}(n) - \mu_i\|$ is the Euclidean distance between the input $\vec{X}(n)$ and the center μ_i . The standard deviation, spread, of the Gaussian function is σ_i (Parzen window size).

The parameters are the weights (w_{ik} and $w_k\phi_0$), centers (μ_i), and widths (σ_i). Typically, the learning is hybrid because involves supervised and unsupervised learning. Unsupervised learning is used for centers and widths and supervised learning for weights [6].

2.2. General Regression Neural Networks (GRNN)

These neural networks are a variant of RBF networks, and they were developed to approximate any function between a set of input and output vectors. It obtains the function from the training data. This model was proposed by Specht (1991) [16].

Typically a GRNN is defined by four layers: input, hidden (Parzen estimator), sum and output. The first two layers are similar to RBF networks. The third layer has two nodes with a sum each: $S_s = \sum_{i=1} \phi_i$ and $S_w = \sum_{i=1} w_i\phi_i$, and the output layer is $y = S_w/S_s$ (see Fig. 1b) [16].

The GRNN model is supported by the theory of non-linear regression [16]:

$$E[y|\vec{X}] = \frac{\int_{-\infty}^{\infty} y f(\vec{X}, y) dy}{\int_{-\infty}^{\infty} f(\vec{X}, y) dy} \quad (3)$$

where $\vec{X}$ is the input vector $(x_1, x_2, \dots, x_d)$ and y represents the output. $E[y|\vec{X}]$ is the expected value of the output given the input $\vec{X}$, and $f(\vec{X}, y)$ is the probability density function. By using a Parzen estimator to obtain $f(\vec{X}, y)$ from the training data, one comes up with the output as follows [16]

$$y(\vec{X}) = E[y|\vec{X}] = \frac{\sum_{i=1}^m w_i \phi_i}{\sum_{i=1}^m \phi_i} \quad (4)$$

If one employs Gaussian functions as basis, the previous equation is expressed as [16]

$$y(\vec{X}) = \frac{\sum_{i=1}^m w_i \exp\left(\frac{-D_i^2}{2\sigma^2}\right)}{\sum_{i=1}^m \exp\left(\frac{-D_i^2}{2\sigma^2}\right)} \quad (5)$$

where D_i is the Euclidean distance, similar to Eq. 2. Comparing Eq. 1 with Eq. 5, the only difference is that Eq. 5 is

normalized and without the term $w_k\phi_0$, the bias term. Still, Eq. 5 is similar to Eqs. 5.134 and 5.138 in [6], which correspond to Nadaraya-Watson regression estimator and normalized RBF network, respectively.

In practice, the advantage of GRNN over RBF is that the only parameter to estimate is σ (known as spread).

2.3. Time Series and Artificial Neural Networks

This section presents a case of study, a time series $\{x(t)_{t \in \mathbb{R}^+}\}$ where t represents the time and $x(t)$ the value at that time¹. Our task is to find a neural network that learns a time series, which is irregularly sampled (see Fig. 2).

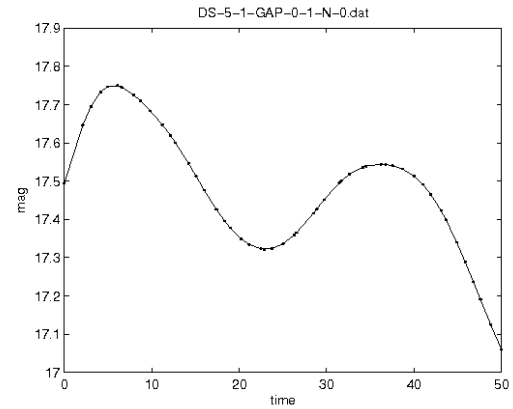


Figure 2. An irregularly sampled time series.

The first idea to think of is to use backpropagation with a single input t and one output $x(t)$ (see Fig. 3a). Another, it is to use a time window [7], the vision (see Fig. 3b).

The results of applying the architectures in Fig. 3 to the time series in Fig. 2 are in the Figures 4 and 5. In both cases, the number of hidden layers is fixed to two. The number of neuron in each layer is changed heuristically to obtain the best results.

The description of plots within Figures 4, 5 and 6 is as follows: the plot at the top-left position depicts the time series to learn. At the top-right position is the function learned. At the bottom-left position, the plot shows the function learned at times that were not considered during the learning, i.e. time of unobserved data. The plot at the bottom-right position shows all previous plots together.

At this point, the best results are for the backpropagation architecture in Fig. 3b, because one gets less oscillations in Fig. 5 than in Fig. 4. Note that is difficult to learn where there are not observations during training.

¹ $x(t)$ will be also referred to as **mag**, because is the measurement unit for real optical data in gravitational lensing, see §3

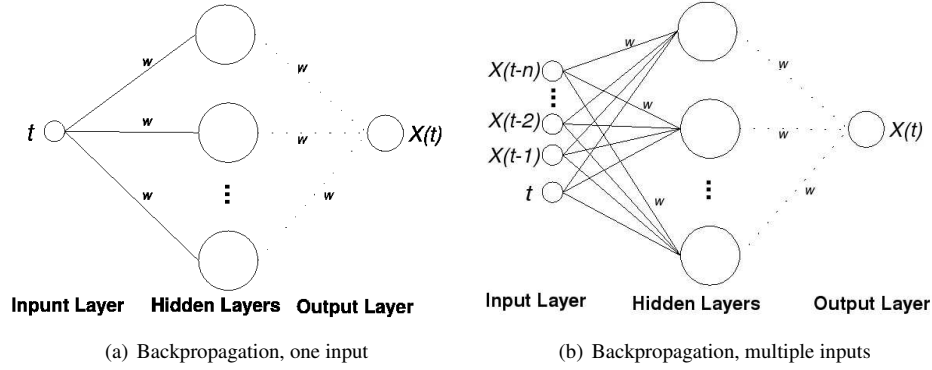


Figure 3. Backpropagation ANN

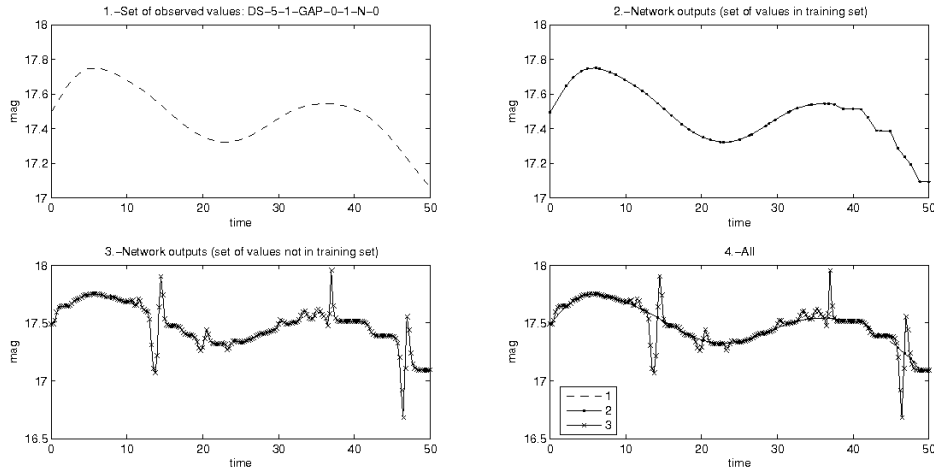


Figure 4. Results of backpropagation (one input). The training time is 59.5 seconds. Read text at §2.3 for details of each plot.

Now, we depict the results of the GRNN architecture with one input and one output in Fig. 6. The spread was fixed to the unit.

As one can observe, Fig. 6 gives the best results not only in accuracy (oscillations where there are not observations, gaps), also in learning time. It is much more faster. This experimental setup was used with different data sets (see §3.1), with an without noise and the results are similar. Therefore, in the following sections, we only use GRNN to model the time series introduced in the following section.

3. Gravitational Lensing and Time Delay Problem

It is a phenomenon of study in astrophysics. Gravitational lensing occurs when the light, coming from a far away quasar or supernova, is deflected for a massive object such as a galaxy. Given this deflection, an observer on the Earth gets two or more images from the single source. The study of this phenomenon is very important for the humanity because the degree of deflection is proportional to the mass between the quasar and the observer (astronomer). Typically this mass is dark, so this is one technique to measure the dark matter in the universe and consequently other cosmological parameters such as Hubble constant and the expansion of the universe [9, 15].

The data obtained from gravitational lensing are

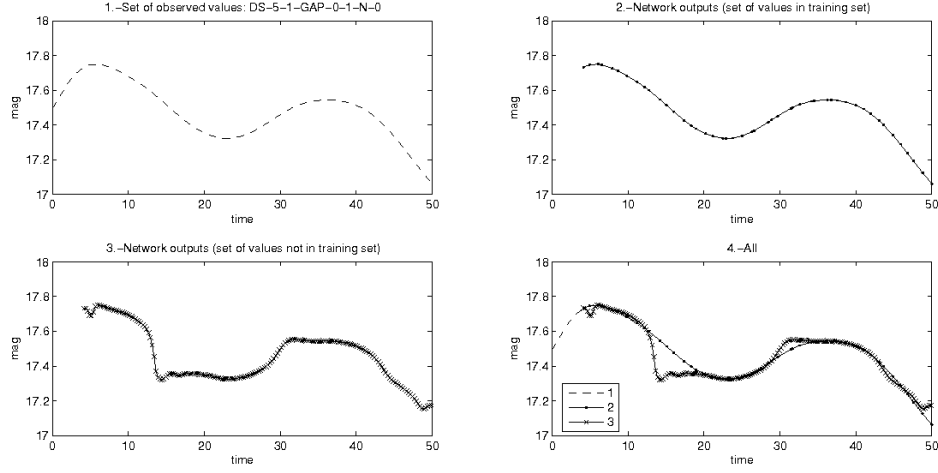


Figure 5. Results of backpropagation (time window). The training time is 97.1 seconds. Read text at §2.3 for details of each plot.

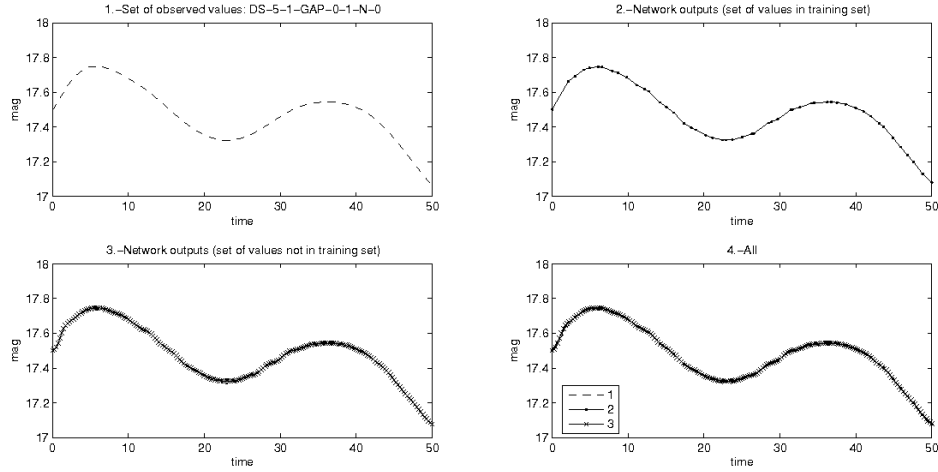


Figure 6. Results of GRNN. The training time is 2.3 seconds. Read text at §2.3 for details of each plot.

recorded in time series, see Fig. 7. The angle of deflection is proportional to the time delay between the curves A and B. Therefore, the time delay estimation is a direct method to measure the angle of deflection and the mass itself.

3.1. Artificial Data

Because the importance of the study of time delay, many efforts have been done to estimate the time delay with real and artificial data [10, 12, 13, 8, 3, 5]. The problem of real data is that the definite time delay estimation for

most known gravitational lens remains uncertain [3]. Even though, in this paper, we study the GRNN on artificial data.

DS-5 The dataset DS-5 has a true time delay of 5 days, and the true offset in brightness between image A and image B is $M = 0.1$ mag. These data simulate real optical observations. It has five underlying functions (shapes). These data sets are irregularly sampled with three levels of noise and gaps of different size. It has 50 realizations per level of noise only. Consequently, this yields 38,505 datasets, with 50 samples each [3].

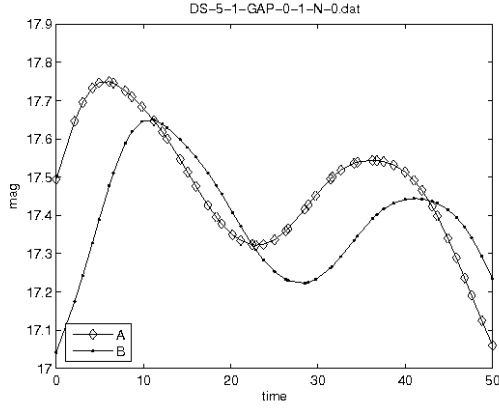


Figure 7. Example of a time series. Note, that there is a delay in time between the curve A and B.

4. Time Delay and GRNN

We propose a novel method based on GRNN to estimate the time delay between pairs of time series.

First, we combine the pair of time series A and B within a single one, say C. This is the time series to learn as in the example in §2.3. Then, we need to find in Eq. 5 the proper σ , spread, for the GRNN. Since we have two different time series, one can find different spreads. We use five-fold cross-validation to obtain the spread in each time series, then we obtain the average. This is the spread that we use. Another method is to obtain the spread from the combined curve, however, this approximation has basically two difficulties: *i*) the combined curve is large and the computational cost on cross-validation is very expensive. *ii*) Because we do not the true time delay when combining the curves (we use trial time delays instead), one can mislead the estimated spread.

To combine the curves, we have two vectors $\vec{A}$ and $\vec{B}$ and its corresponding vector $\vec{T}$, the time; all of the same size. Then, we generate a new vector called $\vec{T}_B$. We apply a trial time delay Δ_p to $\vec{T}_B$, where $\vec{T}_B$ is the time associated to $\vec{B}$. Now, we generate two new vectors $\vec{C}$ and $\vec{T}_C$, that represent the combined curve. We add $\vec{T}$ and $\vec{T}_B$ to $\vec{T}_C$ in order with its respective value $\vec{A}$ and $\vec{B}$ in $\vec{C}$. If there is a duplicated time in $\vec{T}$ and $\vec{T}_B$, we get the corresponding element in $\vec{C}$ by calculating the mean between the corresponding $\vec{A}$ and $\vec{B}$. This combination of light curves is also used by other methods to estimate time delays [8, 10].

Once the method for combining curves has been described, the next steps are straightforward:

1. Use five-fold cross-validation on $\vec{A}$ and $\vec{B}$ to obtain the average spread.

2. Define a set of trial time delays $\Delta_p = [\Delta_{min}, \Delta_{max}]$.
3. Combine the curves given the trial time delay.
4. Train the GRNN on the combined curve $\vec{C}$ by using the spread, estimated as above.
5. Obtain the mean squared error, MSE, at training points between $\vec{C}$ and the curve generated by GRNN.
6. Register the pair (Δ_p, MSE) .
7. Repeat from Step 2, for each Δ_p .

After repeating the above steps, one comes up with a curve in a two dimensional space: MSE correspond to y -axis and Δ_p to x -axis. The best time delay is when the MSE reaches its minimum.

5. Results

We tested the above method to estimate time delays on the dataset DS-5, introduced in §3.1. We compared the results with several methods: Linear Interpolation (LI) [2], two versions of Dispersion spectra method D_1^2 and $D_{4,2}^2$ [10], PRH method [12, 13], K-V a Kernel-based method [3] and EA-M-CV method (evolutionary algorithm) [2]. For all methods were used trial time delays in the range of 0 to 10 with increments of 0.1, since one knows a priori the true delay is 5 days. When real data are used, this range must be estimated.

Table 1 contains the results from all methods including GRNN, the statistics MSE (Mean Squared Error), AE (Absolute Error, average), $\hat{\mu}$ (mean) and $\hat{\sigma}$ (standard deviation) are obtained over the 38,505 datasets described in §3.1. In bold fonts are highlighted the best results. The best results are for GRNN taking in account the statistics MSE, AE and $\hat{\sigma}$. The best results are for D_1^2 considering the statistic $\hat{\mu}$.

MSE and AE measure the accuracy of estimates, that is the error between the estimated delay $\{\hat{\Delta}_i; i = 1, 2, \dots, 38,505\}$ and the true delay $\Delta_0 = 5$. The $\hat{\mu}$ measures the bias and $\hat{\sigma}$ the dispersion or variance.

6. Conclusions

We have introduced a novel method to estimate time delays using GRNN. From Table 1, the results are promising because they are competitive with state-of-the-art methods. We have compared different neural network architectures on irregularly sampled time series, and we conclude that GRNN is more accurate and faster than RBF networks and backpropagation, see §2.3. This method can be used to problems different to gravitational lensing, basically when the sampling is not regular [11] or there are missing data [1].

Table 1. Comparison among different methods, including GRNN

Statistic	LI	D_1^2	$D_{4,2}^2$	PRH	K-V	EA-M-CV	GRNN
MSE	0.49	0.74	0.99	13.46	0.47	0.63	0.30
AE	0.39	0.52	0.59	3.01	0.39	0.41	0.31
$\hat{\mu}$	5.068	5.013	5.589	2.704	4.946	5.015	5.034
$\hat{\sigma}$	0.70	0.86	0.80	2.86	0.68	0.79	0.54

As part of the future work, we want to test the GRNN method on real optical and radio data for the problem of gravitational lensing. We also plan to test this method on other public artificial datasets to compare the performance of this method. A formal analysis of the complexity of algorithms in Table 1 is desirable, with this, one can compare the performance in time.

References

- [1] R. Cook and D. McLeish, editors. *Workshop on Missing Data Problems*. Fields Institute, Toronto, Canada, 2004. <http://www.fields.utoronto.ca/programs/scientific/04-05/missing-data/>.
- [2] J. Cuevas-Tello. *Estimating Time Delays between Irregularly Sampled Time Series*. PhD thesis, School of Computer Science, University of Birmingham, 2007. <http://etheses.bham.ac.uk/88/>.
- [3] J. Cuevas-Tello, P. Tiño, and S. Raychaudhury. A kernel-based approach to estimating phase shifts between irregularly sampled time series: an application to gravitational lenses. In *Machine Learning: ECML 2006, Lecture Notes in Artificial Intelligence (LNAI 4212)*, pages 614–621. Springer-Verlag, Sept. 2006.
- [4] G. Dorffner. Neural networks for time series processing. *Neural Network World*, 6(4):447–468, 1996.
- [5] M. Harva and S. Raychaudhury. Bayesian estimation of time delays between unevenly sampled signals. In *IEEE International Workshop on Machine Learning for Signal Processing*, pages 111–122. IEEE, 2006.
- [6] S. Haykin. *Neural Networks: a Comprehensive Foundation*. Prentice Hall, 1999.
- [7] P. Isasi-Viñuela and I. M. Galván-León. *Redes de Neuronas Artificiales Un Enfoque Práctico*. Pearson Prentice Hall, 2004.
- [8] J. Lehar, J. Hewitt, D. Roberts, and B. Burke. The radio time delay in the double quasar 0957+561. *Astrophysical Journal*, 384:453–466, 1992.
- [9] P. Newbury and R. Spiteri. Inverting gravitational lenses. *SIAM Review*, 44(1):111–130, 2002. Society for Industrial and Applied Mathematics.
- [10] J. Pelt, J. Hjorth, S. Refsdal, R. Schild, and R. Stabell. Estimation of multiple time delays in complex gravitational lens systems. *Astronomy and Astrophysics*, 337(3):681–684, 1998.
- [11] D. Percival. Spectral representation of irregularly sampled radar image sequences. In *Proceedings of International Conference on Image Processing*, volume 3, pages 911–914. IEEE, 1996.
- [12] W. Press, G. Rybicki, and J. Hewitt. The time delay of gravitational lens 0957+561, I. Methodology and analysis of optical photometric data. *Astrophysical Journal*, 385(1):404–415, 1992.
- [13] W. Press, S. Teukolsky, W. Vetterling, and B. Flannery. *Numerical Recipes in C++: The Art of Scientific Computing*. Cambridge University Press, second edition, 2002.
- [14] L. C. J. Robert J. Howlett, editor. *Radial Basis Function Networks 2 new advances in design*. Physica-Verlag, 2001.
- [15] P. Saha. Gravitational Lensing. *Encyclopedia of Astronomy and Astrophysics*, 2000.
- [16] S. Torra-Porras. *La siniestridad en seguros de consumo anual de las entidades de preveión social. Perspectiva probabilística y econométrica. Propuesta de un modelo econométrico neuronal para Cataluña*. PhD thesis, Universidad de Barcelona, 2003.

Using Complex Neural Networks on Noise Cancelling

Donato Hernández

Facultad de Ingeniería Mecánica, Eléctrica y Electrónica

Universidad de Guanajuato

Carretera Salamanca-Valle de Santiago km 3.5+1.8, Comunidad de Palo Blanco, Gto., México.

donato, selo, rcelorio, avina and canedo@salamanca.ugto.mx

Sergio Ledesma

Rene Martínez

Gabriel Aviña

Gerardo Canedo

Abstract

We present a method to reduce noise on signals applying complex auto-associative neural networks. Experimental results using the sine, triangular and sawtooth signals are performed to validate our results. This method is based on the use of complex neural networks learning and is capable of eliminating or reducing noise on learned signals. First, a training set consisting of signal values without noise at different phases is used for learning. Second, The signal is processed first using the Fast Fourier Transform (FFT) to obtain the frequency components. Third, three different complex neural networks are trained using the transformed clean signal until a minimum error criterion is satisfied. Finally, noisy transformed signals are applied to the complex neural network for simulation purposes. The neural networks' output signal is inverse transformed and compared with the clean signal for each case to verify the difference between them.

1. Introduction

Noise is a real problem in system and transmission signals, some methods are used commonly to reduce it, like analog filters [11, 10] or digital filters [3]. However, if we know the signal's pattern, that is, we have a mathematical expression for the signal's frequency and waveform, we can use an Artificial Complex Neural Network (ACNN) to reduce noise in the most of the cases, signals with waveforms like sine, triangular and sawtooth can be used to train a neural network, and then use a noisy signal as input to obtain a noise reduced or cancelled signal.

In this work, we used the Neural-Lab program, an independent neural network project for the implementation of algorithms and handling developed in the Visual C++ lan-

guage. We used as first part, the creation of some training sets with the signal's frequency and waveform without noise to be used in the ACNN. Then in the second part, we trained complex neural networks to meet an error criterion, using the Fast Fourier Transform of the signal and 0, 1 or 2 hidden layers to obtain a specific Mean Square Error (MSE) by means of Simulated Annealing (SA) [4, 1] to achieve the goal. Later, an input noisy signal is applied to the ACNN to obtain the output filtered signal.

This paper is structured as follows. Section 2 indicates the algorithm implementation is explained. Section 3 reviews some resulting work and discusses some points about the application of the algorithm to noisy signals. Finally, section 4 provides the contributions of the paper and shows direction for future work.

2 Noise Reduction.

Noise reduction is the operation of eliminating undesired frequency components of a signal, if the signal's waveform is known, then a clear signal can be obtained using a trained ACNN by means of an adequate training set with the signal values.

2.1 Training set

The first step to implement a neural network is to create the training set, representing the values for one period (2π) of the sine waveform for example; a program written in Visual C++ for that purpose was written. The complete training set for the neural network included the signal at different phases. It can be observed in Figure 1 some sine waveforms included in the training set. For clarity, only 10 training cases are shown in this figure. The total training set is composed of 360 sine waveforms at different starting points, in a neural network with 32 inputs and 32 outputs.

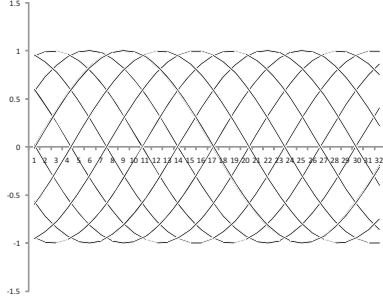


Figure 1. The training set for the sine waveform.

That is, an increment delta of $1/360$ was used for the sine waveform signal. This can be represented by the following equation:

$$f(t) = \sin(\omega t + \theta),$$

where θ is the phase angle with values from 0° to 360° (2π), one signal period.

In the same way, the training sets for the triangular and the sawtooth signals were built, varying the phase for all possible cases. To build the triangular training set equation 2 was used, it can be observed from Figure 2 for the triangular and sawtooth signals.

Let define

$$\Lambda(t) = mt, \quad 0 < t < T_0, \quad (1)$$

where m is the line slope and T_0 is the signal's period.

Then, we can define the sawtooth signal as

$$f(t) = \sum_{k=-\infty}^{\infty} \Lambda(t + \lambda - kT_0) \quad (2)$$

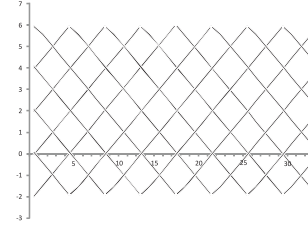
where λ is the phase with uniformly spaced values from 0 to 8 seconds in this example.

The total training set is composed of 256 triangular or sawtooth waveforms, while the used neural network has 32 inputs and 32 outputs. That is, a delta increment of $1/64$ of the period of the triangular or sawtooth signals was used. In this cases, Figure 2 shows only eight training cases for the triangular 2(a) and sawtooth 2(b) signals.

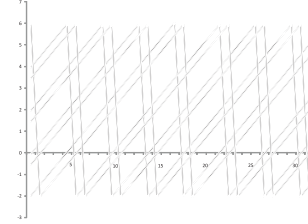
Now that we have the training set of values for each signal, we proceed to obtain the FFT, for training the ACNN with the values corresponding to the signal without noise.

2.2 Training the neural network

The second step is to train the respective neural network to meet a specified mse, a program written in Visual C++



(a) Triangular.



(b) Sawtooth.

Figure 2. Training sets.

(Neural Lab) for that purpose was used; two methods to perform the optimization of the neural network i) conjugated gradient [9, 5], and ii) Levenberg Marquardt [9, 7] were used. For the layer initialization the following methods were used:

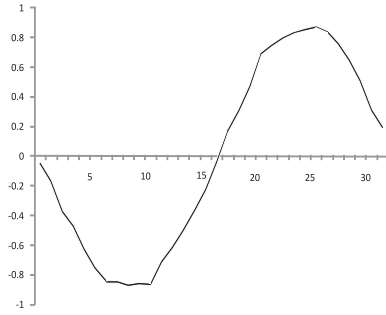
- Genetic algorithm [9, 7, 2, 6].
- Simulated annealing [9, 7, 4, 1].
- Annealing + regression [9, 8, 7].
- Regression only algorithms [9, 7, 6].

Simulation results for each of the signals is reviewed and discussed in the next section.

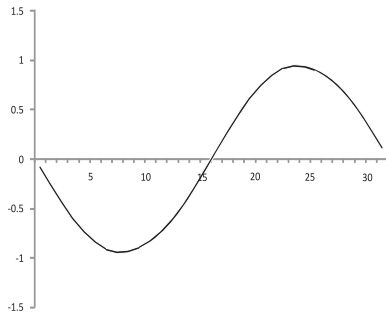
3 Experimental Results

This section evaluates the performance of our proposed algorithm for noise reduction. A clear signal to test the trained neural network and later a noisy signal with 5%, 10% and 20% of additive noise level for each case was used, it can be observed from figures 3, 4 and 5, that the ACNN output signal has less noise than that input signal, and it has the same waveform of the signal used for training.

Also, the triangular and sawtooth signals with the same noise levels (5%, 10% and 20%) were tested, and the obtained results can be observed from figures 6 and 7. As it



(a) Sine with 5% noise.



(b) Output sine signal.

Figure 3. The sine signal with 5% noise level.

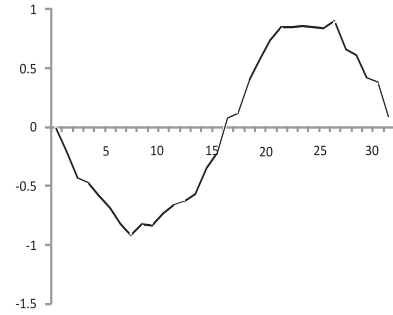
can be seen the error for the sawtooth signal is very significant, a 10% of amplitude in the worst of the cases, this is because this kind of signal, has an abrupt transition from a high amplitude to low amplitude, and this is difficult to learn for complex neural networks due to the high frequency content of this kind of signals.

4 Conclusions and future work

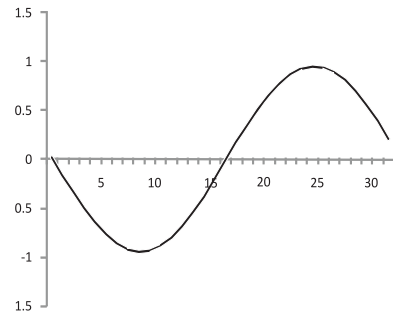
The factors that determine the signal quality are related to the noise level present in the input signal. The main contributions of this paper have been:

- A method to use neural networks to reduce noise in systems signals.
- The implementation of an algorithm to train a neural network with signals' waveforms.
- The study of some common signals' waveforms like the sine, triangular and sawtooth with a noise level.

This work has not been finished yet, some ideas to be applied in the training sets are part of current and future work.

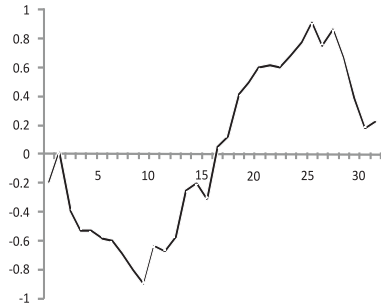


(a) Sine with 10% noise.

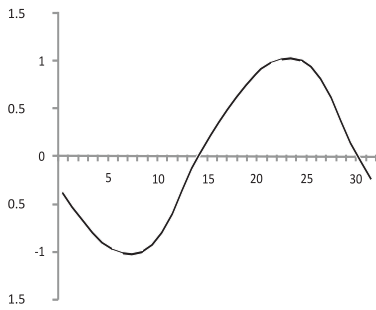


(b) Output sine signal.

Figure 4. The sine signal with 10% noise level.

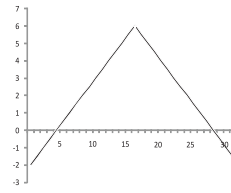


(a) Sine with 20% noise.

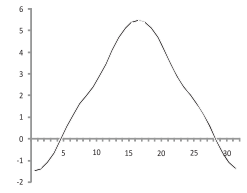


(b) Output sine signal.

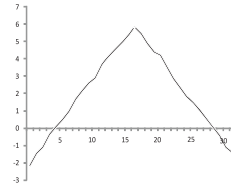
Figure 5. The sine signal with 20% noise level.



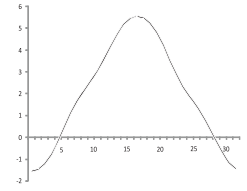
(a) Triangular.



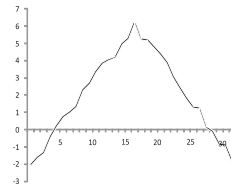
(b) Output signal.



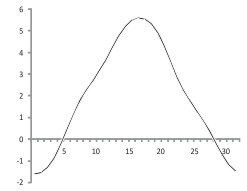
(c) Triangular with 5% noise.



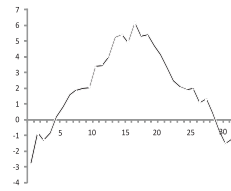
(d) Output signal.



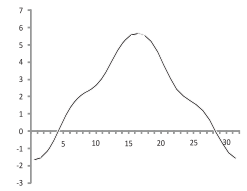
(e) Triangular with 10% noise.



(f) Output signal.



(g) Triangular with 20% noise.



(h) Output signal.

Figure 6. The triangular signal.

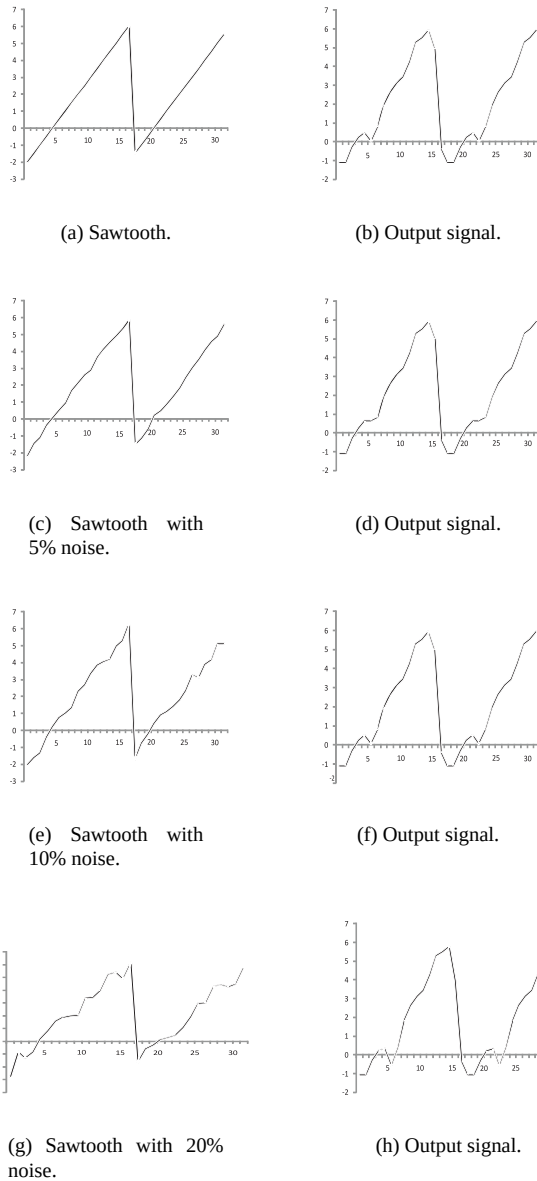


Figure 7. The sawtooth signal.

We have special interest in the development and application of new techniques using ACNN, to reduce noise, with less levels of hidden layers to test the performance. It may be interesting to decrease the number of neurons to meet a required mse for training a validation. Additionally, the hardware implementation of neural networks using a minimum of elements should be explored.

References

- [1] E. Aarts. *Simulated Annealing, Theory and Practice*. John Wiley & Sons Inc., 1987.
- [2] L. Davis. *Handbook of Genetic Algorithms*. Van Nostrand Reinhold., 1991.
- [3] R. E. Kalman. A new approach to linear filtering and prediction problems. In *Journal of Basic Engineering.*, pages 35–46. Journal of Basic Engineering., 1960.
- [4] S. Kirkpatrick, C. D. Gelatt, and M. P. Vecchi. Optimization by simulated annealing. In *Science, Vol 220, Number 4598.*, pages 671–680. Science., 1983.
- [5] T. Masters. *Practical Neural Network Recipes in C++*. Academic Press, Inc., 1993.
- [6] T. Masters. *Signal and Image Processing With Neural Networks*. John Wiley & Sons Inc., 1994.
- [7] T. Masters. *Advanced Algorithms for Neural Networks*. John Wiley & Sons Inc., 1995.
- [8] T. Masters. *Neural, Novel and Hybrid Algorithms for Time Series Prediction*. John Wiley & Sons Inc., 1995.
- [9] W. H. Press, S. A. Teukolsky, W. T. Vetterling, and B. P. Flannery. *Numerical Recipes in C++*. Cambridge University Press., 2002.
- [10] K. S. Shanmugam. *Digital and Analog Communications Systems*. John Wiley & Sons Inc., 1979.
- [11] D. Wobschall. *Circuits Design for Electronic Instrumentation*. John Wiley & Sons Inc., 1979.

A neural network may show the best way to learn how to count for students in elementary Math courses

Antonio Neme, Vicente Carrión
Universidad Autónoma de la Ciudad de México
Grupo de Dinámica no Lineal y Sistemas Complejos
México, D.F., MX
neme@nolineal.org.mx

Abril Muñoz, Pedro Miramontes
Universidad Nacional Autónoma de México
Facultad de Ciencias

Alejandra Cervera
Comisión Nacional Para el Uso y Conocimiento de la Biodiversidad, México

Abstract

Learning how to count in different bases has been seen as a trivial task in almost all introductory mathematics courses. However, the low performance shown by many students, while performing this task, is appalling. This situation has motivated serious research in this matter. In order to study a model of count learning, we analyze the performance of a multilayer perceptron that learns to count in several bases (5, 10, 13, 20, 60). We give evidence that it is not equivalent for the model to learn to count in all base as the errors are not equivalent, biased toward a low error when the task is to learn to count in base 20. When the task is to learn to count in all bases following a given sequence, the model shows non-equivalent errors for some bases. This may shed some light on education planning that can result in better introductory courses.

1 Introduction

The learning capability of a system is based on a modification in a set of parameters that allows the system to improve its performance, to find alternative routes or to propose new insights into the problem [1]. In biological organisms with neural system, learning occurs as the organism interacts with its environment which leads to a modification in synaptic strengths, or to develop new synaptic connections [2].

Learning in humans has been extensively studied from

a vast set of points of view. From the computer sciences perspectives, learning has been analyzed mainly from two different angles, conflicting at times with each other. The first point of view is that of learning hard computing, mainly symbolic processing. The second one, which is the foundation of the present work, tries to model the human learning process as the result of interacting components, named neurons. This perspective receives the name of connectionism. Learning by connectionism systems has been formalized and there are several interesting results [3, 4].

The multilayer perceptron (MLP) has been widely applied as a model of human learning [2]. Although it may be a very critic issue to assume that learning in MLP is equivalent to learning in living organisms, one of the hypothesis we assert in this work is in fact that learning in both systems, *in vivo* or *in silico*, are equivalent and what can be inferred in one of them is valid in the other, at least in some general way. That is, learning dynamics and constraints are independent of the substrate.

The application of neural networks in the mathematical education field has not been widely studied. An exception is the work reported in [5], in which the use of multilayer perceptrons as a tool to model the learning dynamics of multiplication tables has identified a sequence of input vectors that allows the students to perform better in this task.

In particular, learning mathematical concepts is a very specific situation and is reported to be a very difficult task for many students all over the world [6]. Different theories about those apparent difficulties in learning mathematical concepts have been proposed [8, 9], such as those that states there is an intrinsic difficulty in learning some issues

and others that sustains there is a lack of capabilities in the advisor.

One particular mathematical concept that is among the first ones to be learned is that of counting [6]. Once the subject learns to count, it has been implied that he or she has achieved the skill of addition, as well as acquired the concept of positional relevance. Learning to count in base-10 is completed at an early stage. Learning to count in bases other than 10 is taught some years afterwards, and, at least in some countries, it has been reported as a very problematic task since many students fail to show a proper understanding of the concept. Bases other than 10 have been applied in different civilizations, for example, base 60 was used in Babylon and base 20 system was widely employed in Mesoamerica, mainly by the Mayan civilization [7]).

At least in some countries, mainly in Latin America, when students are evaluated on their mathematical skills, the results are not encouraging [12]. In those countries, one of the main problems seems to be the conversion between bases. This was the point of departure for our research.

The question we have raised is that the order in which we learn to count in different bases may not be the most adequate for the students to acquire the skills needed to properly understand the concept. That is, learning to count in base 10, then in base 20 and then in, say, base 60 may not be the best sequence if the final goal is that the student learns the concept of counting independently from the base system. Some exercises have been proposed in different experimental schools that seek to help the students to understand better the basic mathematical concepts, among them, that of number and counting [13].

Here, we propose the use of a neural network that learns to count in different bases and that identifies the sequence of bases that, at the end, leads to a lower error. Also, we try to identify if there is an inherent difficulty, at least for neural networks, in learning to count in some bases.

In our models, several multilayer perceptrons (MLP) are trained with the same data set, but presented in different order. One of them, say, MLP i , learns to count first in base 10, then in base 20, then base 5 and finally in base 60. Other MLP, j , learns to count in base 20, then in base 5, then in base 60 and finally in base 10. The idea is to compare the performance of MLP i and j so the sequence that leads to lower errors may be put into practice in elementary Mathematics courses.

Here, we base our research in basic math concepts learning modeled in the MLP.

2 Multilayer perceptrons and learning

The basic unit of neural computation is the neuron. Several models have been proposed, but the one that was applied in this work is the perceptron. A perceptron is a unit

that receives stimulus from the environment or from other perceptrons. Each stimulus is weighted and the total stimulus the perceptron receives, is the sum of the ponderated stimulus. The discrete state perceptron may be in one of two states: active or inactive. If the total stimulus that the perceptron receives is higher than a given threshold, then the perceptron is active, otherwise it is inactive. In the continuous state perceptron its state (also called output) is a continuous function on the total stimulus. Because of its mathematical properties such as a smooth derivative [14], the sigmoidal function is often applied: $y = \frac{1}{1+exp(-\alpha x)}$, where x is the total stimulus an unit receives and α is a parameter that determines smooth or abrupt steps.

Although the computational power of a perceptron is limited to classify only linearly separable inputs, a network of perceptrons shows a more interesting set of computational capabilities. Among those features are the possibility of classifying non-linearly separable inputs and that of universal approximation [15]. There are several topologies of neural networks, but the one we applied in this work is a feedforward network, in which units are located in different layers and units in the same layer do not interact with each other. The basic topology for feedforward networks consists in three layers: the input layer, the hidden layer and the output layer.

Those neurons in the hidden layer receive the influence of those units in the input layer, whose output or state is defined almost always by a linear function over the input to the network (see fig 1). The units in the output layer receive the influence from the units in the hidden layer. Between each pair of units located in subsequent layers, there is a synaptic strength, or weight. The modification of the set of weights in a network is known as learning. There are as many free parameters that may be tuned as there are weights connecting units. The output of the last layer is known as the network output and is compared with the desired output that is associated to each input vector. This kind of network is known as a multilayer perceptron, or MLP (see fig 1). In the MLP, the processing is done by layers. Only after the neurons in the hidden layer process the input, their outputs are sent to the output layer, which also processes that information.

The MLP seeks a function that relates the input data with the output or desired state. It can be trained by a number of learning algorithms, but the statistical learning theory and the non free lunch theorem states that on average over all learning spaces, all algorithms show the same performance [16]. So, if there is a functional relation between the input data and the desired output, the MLP will find an approximation with an arbitrarily small error. At a low level analysis, the MLP behavior may be seen as that of point approximation by means of hyperplanes [15].

The learning in MLP, that is, the weight modification

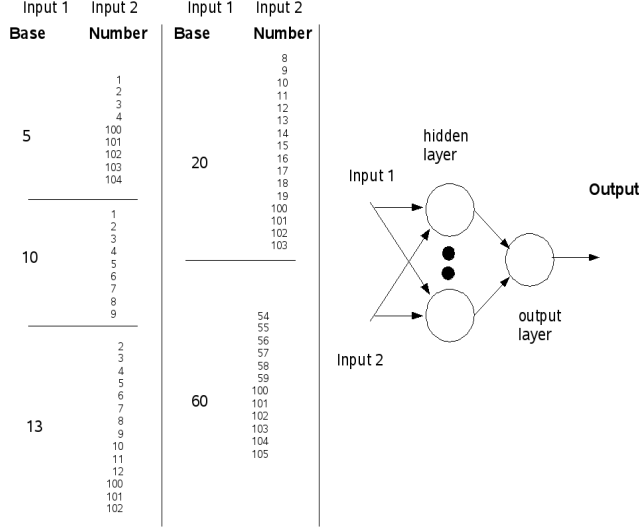


Figure 1. The MLP learns to count in several bases. It approximates the function successor.

scheme, is achieved by many different algorithms. Among them, the genetic algorithm and backpropagation are the most frequently used. In this work, the MLP are trained by a genetic algorithm [10, 11].

3 Experiments and results

Two learning tasks were studied in this work. In the first one, each MLP learns to count in only one base. In the second task, a given MLP learns to count in all the bases, which is a learning of simultaneous tasks. Learning of simultaneous task is not trivial when tasks are not correlated [17]. Here, there is not a clear correlation between the different learning sequences, as the symbols in each case represent different quantities because of the positional properties.

The learning dataset consists of two inputs and one output. The first input is a number between 0 and 999 while the desired output is the successive number. The second input identifies the base in which the counting is realized (see fig. 1). Bases 5, 10, 13, 20 and 60 were included. The reason for selecting bases 10, 20 and 60 is historical, as numerical systems of those bases have been widely documented in several civilizations [7]. Base 5 and 13 were selected in order to analyze learning in prime bases.

In the first learning task, a MLP is presented with 800 input vectors that represent a number between 0 and 999 in the specified base. For each input, the desired output, that is, the successive number of the input, is presented. After the training stage, the MLP is tested with a set of 200 inputs, that represents numbers between 0 and 999 in the same base

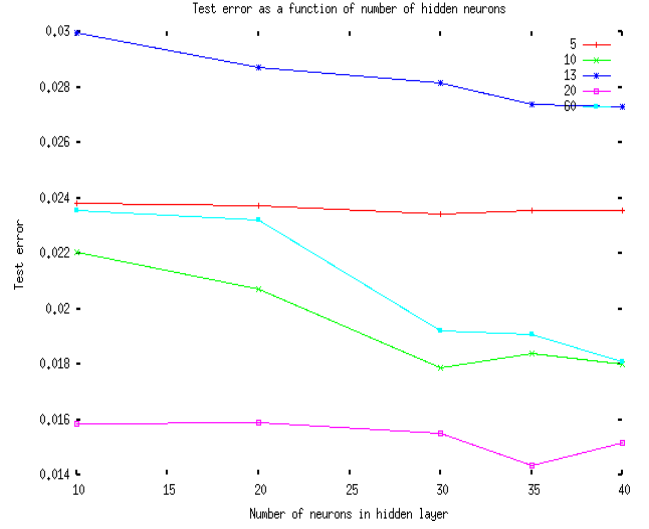


Figure 2. Test error for the studied bases as a function of the number of neurons in the hidden layer.

as that in which it was trained, but that have not been previously learned. Fig. 2 shows the average test error for 100 simulations achieved by MLP with the specified number of neurons in the hidden layer, for each base. The number of epochs is 5000 for all cases and $\alpha = 2.0$. The error that was measured is the mean squared error (MSE), which is defined as $Er = \frac{1}{N} \sum_{i=1}^N (y_i - o_i)^2$, where y_i is the network output for input i and o_i is the desired output for the same input. N is the number of input vectors. The training set and test sets include the same numbers for all bases, in the proper base representation.

From fig. 2 it is clear that MLPs learn to count with a lower error in base 20. Base 13 is the one that presents a higher test error while learning to count in base 10 and base 60 is almost equivalent (in sense of error) for all MLPs, specially for large ones (30 or more neurons in the hidden layer). Learning to count in base 5 does not improve with more neurons in the hidden layer.

In the second task, MLPs learn to count in the five bases. Each MLP is taught the same data set, but each base system is presented in different order, that is, the MLP learns to count first in a given base and then it continues to learn to count in the following bases. This is done for all the base systems studied.

There are 25 data sets, as there are $5! = 24$ different orderings of the studied base systems plus a control data set ordering. The first data set is that in which the MLP learns to count in base 5 first, then base 10 followed by base 13,

Table 1. Test error for different base learning sequences for each base. The MLP has 35 neurons in the hidden layer. It is observed that the test error is higher when the MLP is evaluated in bases 13 and 20, independently from the learning sequence.

Base sequence	5	10	13	20	60
5 10 13 20 60	0.024	0.026	0.035	0.036	0.029
60 20 13 10 5	0.024	0.027	0.036	0.036	0.029
13 5 60 20 10	0.024	0.028	0.036	0.035	0.028
5 60 13 20 10	0.023	0.027	0.036	0.036	0.029
10 20 5 60 13	0.023	0.027	0.035	0.036	0.029
random	0.024	0.027	0.036	0.035	0.03

base 20, and finally base 60. In the last data set, there is not an order in base learning in which the inputs are presented to the MLP, that is, it may occur that the number 10 in base 5 is taught followed by the number 24 in base 13 and so on.

Tables 1-2 show the MSE when the task was to learn to count in the five base systems, with 35 neurons in the hidden layer and with 6 hidden neurons. The left-most column shows the base learning sequence. Only four of the 24 sequences are presented, as well as that of random base learning. For a MLP that learns to count in a given base sequence, the test error is shown when the MLP is evaluated only in the specified base. It is observed that the test error for bases 20 and 13 are always higher than the test error for the remaining bases. It was expected for all the bases to show similar test errors, as there is not a bias in the composition of training and test data set.

This difference in test errors is not caused by the relative position in which bases 13 and 20 are taught, as can be seen in the table, as in some sequences, those bases are presented at the beginning while in other cases are presented at the end.

This may seem as a contradiction with the results shown in fig. 2, where the test error for MLP, whose unique task was to learn to count in base 20, was lower than that in those MLP that were asked to learn in other bases.

This apparent contradiction may be interpreted as follows. Even though a MLP may properly learn to count in base 20, when it is asked to learn in other bases as well, its performance is degraded. Although we are in a stage in which we can not give a full sustained answer to this problem, we believe that the positional information that is implicit in the input vectors is hard for the MLP to manage. This behavior is consistent with empirical observations [18], in which a confusion is reported when children study different bases. The training error for all base sequences presents the same distribution, that is, bases 13 and 20 are the harder to learn.

The difference previously addressed between test errors for bases 13 and 20 and that for the remaining bases is not caused by a biased distribution of weight vectors. In order to discuss it, let's first define the weight space.

Table 2. Test error for different base learning sequences for each base. The MLP has 6 neurons in the hidden layer. It is observed that the test error is higher when the MLP is evaluated in bases 13 and 20, independently from the learning sequence.

Base sequence	5	10	13	20	60
5 10 13 20 60	0.025	0.029	0.036	0.035	0.033
60 20 13 10 5	0.027	0.026	0.033	0.039	0.031
13 5 60 20 10	0.026	0.029	0.035	0.037	0.036
5 60 13 20 10	0.030	0.027	0.035	0.036	0.035
10 20 5 60 13	0.026	0.030	0.034	0.041	0.030
random	0.026	0.028	0.037	0.035	0.036

Each MLP has a total of $M = 3 \times n$ weights, where n is the number of hidden neurons. The genetic algorithm and backpropagation that train the MLP are thus a search process of weight vector in the M-dimension space. How different are the MLP that learn to count in, say base 10 than those that learn to count in base 60 is a matter of comparison of weight vectors in the M-dimensional space. Through a self-organizing map (SOM), we have compared the differences of MLP accordingly to the learned base. The SOM is a non-linear mapping from a high-dimensional space to a low dimensional space (two dimensional in order to be analyzed in the computer screen) that presents very high quality of clustering [19].

In the low-dimensional map space, multidimensional objects are mapped in such a way that similar objects tend to be distributed nearby while different objects tend to be mapped in distant areas. Fig. 3 shows the map obtained for several MLP with 35 neurons in the hidden layer, which define a weight space of dimension 105. The numbers 1 to 5 refers to the MLP that were taught one of the five possible base learning sequences shown in table 1 and the number 6 refers to random sequence: MLP that were taught base 5, then base 10, base 13, base 20 and base 60 are identified with number 1. MLPs that were taught bases in order 60, 20, 13, 10 and 5 are identified with number 2 and so on.

It is observed that for a given sequence (from 1 to 6), MLPs that learned the bases in a specific sequence do not form a cluster. In other words, the weights for the MLPs that learned a given sequence are not located in a specific region, which would mean that there is a preferred region in the multidimensional weight space. So, the anomalous high test errors for the bases 13 and 20 can not be explained in

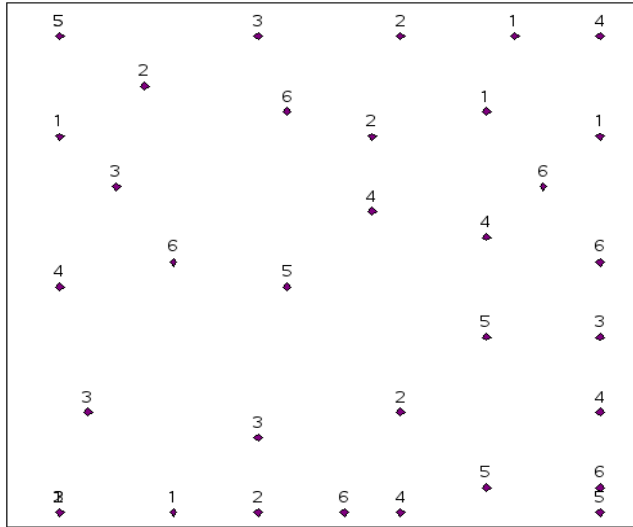


Figure 3. Map obtained by SOM. The numbers indicate the base sequence (see table 1).

terms of a bias in the weights towards regions that overfit other bases at the expense of under fitting them.

4 Conclusions and discussion

Learning to count in several bases is a task that has been reported to be difficult in young students. We applied MLP to study if there is an intrinsic difficulty (measured in terms of error) in learning to count in several bases.

We found that when a MLP is asked to learn to count in only one base, the base that was learned the best is 20, whereas base 13 is the one with the highest test errors, which means that it is difficult for the MLP to properly generalize the concept of counting in that base. This is in the same line with empirical results obtained when studying learning at elementary school.

When the MLPs are trained to learn to count in five bases, there is a difference in test errors. Bases 20 and 13 present higher errors than the remaining bases, independently from the network size. This may suggest an implicit difficulty in counting in those bases. This may be taken into account by education planners, as, at least for the five bases we included in our work, there is not a base sequence in which learning base 20 or 13 present errors as low as those from the remaining bases.

Although a direct extrapolation from models to real subjects may be dangerous, at least we have given evidence that there are inherent difficulties in learning simultaneously to count in in several bases.

References

- [1] Engel, A., Van den Broeck, C. Statistical mechanics of learning Cambridge University Press. (2001).
- [2] O'Reilly, R., Munakata, Y. Computational explorations in cognitive science. MIT Press (2000).
- [3] Judd, S. Neural network design and the complexity of learning. MIT Press (1990).
- [4] Vapnik, V. Statistical learning. Springer. (1998).
- [5] Tatuzov, A. Neural network models for teaching multiplication table in primary school. International Joint Conference on Neural Networks 5212-5217. (2007).
- [6] Krebs, G., Squire, S., Bruant, S. Children's understanding of the additive composition of number and of the decimal structure: what is the relationship? International Journal of Educational Research. Vol. 39, No. 7. 677-694. (2004).
- [7] Bell, E. The development of mathematics. McGraw-Hill. (1949).
- [8] Filloy, E., Puig, L., Rojano, T. Educational algebra: a theoretical and empirical approach. Springer. (2008).
- [9] Fuson, K. Childrens Counting and Concept of Number . New York: Springer Verlag. (1998).
- [10] Mitchel, M. An introduction to genetic algorithms. MIT Press (1996).
- [11] Fogel, D., Chellapilla. Evolving an expert checkers playing program without using human expertise. IEEE Trans. on Ev. Computation 422-428. (2001).
- [12] 2003 Report from the Programme for International Student Assessment (PISA). available on: www.pisa.oecd.org
- [13] Thompson, I. The role of counting in the idiosyncratic mental calculation algorithms of young children. European Early Childhood Education Research Journal, 3, 1, 5-16. (1995).
- [14] Haykin, S. Neural networks a comprehensive foundation, 2nd. edition. Prentice-Hall. (1999)
- [15] Rojas, R. Neural networks: a systemic introduction. Springer (1996).
- [16] Wolpert, D. Macready, W. No free lunch theorems for search. Technical report. SFI (1995).

- [17] Liu, Y., Yao, X. Simultaneous training of negatively correlated neural networks in an ensemble. IEEE Trans. on Systems, Man and Cybernetics. Vol. 29 No. 6 (1999). 716-725.
- [18] Nunes, T., Bryant, P. Las matemáticas y su aplicación: la perspectiva del niño. Siglo XXI. (1997).
- [19] Kohonen, T. Self-organizing maps. Second edition Springer. (2001).

Fault Prediction Using Artificial Neural Network and Fuzzy Logic

Shafqat M. Virk
Department of CSE.,
U. E. T., Lahore,
Pakistan.
shafqat.virk@uet.edu.pk

Aslam Muhammad
Department of CSE.,
U. E. T., Lahore,
Pakistan
maslam@uet.edu.pk

A. M. Martinez-Enriquez
Department of CS,
CINVESTAV-IPN,
Mexico
ammartin@cinvestav.mx

Abstract

This paper studies different vehicle fault prediction techniques, using artificial neural network and fuzzy logic based model. With increasing demands for efficiency and product quality as well as progressing integration of automatic control systems in high-cost mechatronics and safety-critical processes, monitoring is necessary to detect and diagnose faults using symptoms and related data. However, beyond protective maintenance services, it is viable to integrate fault prediction services. Thus, we studied different parameters to model a fault prediction service. This service not only helps to predict faults but is also useful to take precautionary measures to avoid tangible and intangible losses.

Keywords: Faults, Artificial Neural Network, Fuzzy Logic, Neuro-Fuzzy, Neuro-Neuro, Recurrent Neural Network, Back-propagation

1. Introduction

A conventional maintenance strategy can be corrective and preventive. In corrective maintenance, components are maintained on a needed basis, usually after a major break down, while in preventive maintenance, they are replaced based on a conservative schedule to prevent commonly occurring failures. Because of corrective maintenance strategies, the system becomes unavailable for a much greater time, while preventive maintenance strategies are quite costly because of frequent replacement of components [1]. Hence, a new predictive maintenance strategy is required, in order to know the fault in any component beforehand. This process of predicting future failures of different components of a vehicle is known as Vehicle Fault Prediction Analysis. We investigate possibilities of failure of different vehicle components, and implement a suitable architectural model in order to predict the health/remaining life of a vehicle part.

We study and explain the Neuro-Fuzzy architecture and we discuss some shortfalls in it (see Section 2) then Neuro-Neuro architecture (see Section 3) is explained, Both of these models are compared (see Section 4) to find out the pros and cons of each one. Section 5 contains a short description of related work. Conclusions and directions for future work are given in Section 6.

2. Neuro-Fuzzy architecture

The Neural Network paradigm is inspired by biological nervous system such as brain information processing [2][3]. Moreover, it is configured for a specific application, such as pattern recognition or data classification through a learning process [4]. Apart from self learning capabilities, neural networks can be self-tuned, and are applicable to model various systems.

Neural networks also have some drawbacks which limit its capability to be used alone for various problems. Among these drawbacks, the most important one is its inability to handle linguistic and vague information. Conversely, a fuzzy rule base consists of readable if-then statements that are almost in natural language but it cannot learn the rules by itself [7].

Neuro Fuzzy Systems (NFS) combine the advantages of fuzzy logic with the learning capabilities of neural networks [5]. Moreover, it is a popular framework for solving complex problems. While learning capability is an advantage from the viewpoint of Fuzzy Inference System (FIS), the formation of linguistic rule base will be an advantage from the viewpoint of Artificial Neural Network (ANN).

NFS can be categorized as cooperative and concurrent model. The former is considered as a preprocessor (see Figure.1) wherein ANN learning mechanisms determines the fuzzy inference membership functions or fuzzy rules from training data. Once FIS parameters are determined, ANN goes to background. The rule base is usually determined

using fuzzy clustering algorithms [11]. Membership functions are usually approximated by NN from the training data.

In NFS concurrent model, continuously ANN assists FIS to determine the required parameters especially when the input variables of the controller cannot be measured directly [6].

2.1. Neuro-Fuzzy model

In this model, the series of values of each parameter is passed to its respective neural network as shown in Figure 1. Each neural network will predict the value of its corresponding parameter. The predicted values will then go to the fuzzy logic and remaining life of component will be calculated by the fuzzy inference system.

2.1.1 Neural network. Each NN is trained by Back-propagation algorithm [4] in order to predict the next value. Back-propagation is a gradient following method, and it is used to train feed forward neural networks. The weights are updated continuously at the end of each training cycle. The final output results in a predicted value of the particular parameter. Each NN has to be trained once for one particular set of training data. If there is any change in training dataset of a particular parameter, then NN has to be re-trained for the updated training data. The updated weights and network's final output are stored at the end of training phase. The training phase of any particular NN ends when the mean squared error reaches to a desired cut off level. For instance the cut off level of 0.01 is a good option.

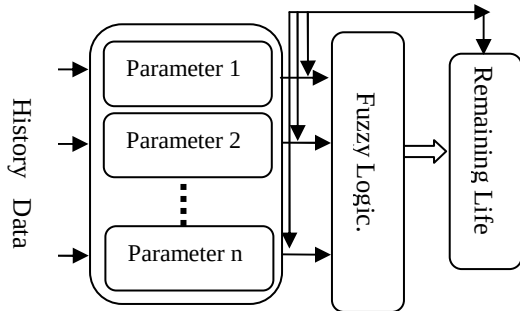


Figure 1: Neuro-Fuzzy architecture

ANN is tested for any given selected pattern of the same training dataset to verify the produced predicted output, at the end of its training phase.

But, whether this single value is enough dependable to predict the fault of a particular component. Certainly not, because at any single given instance of time, this single value represents the behavior of NN at that instant. As sensors send value after a specific time

interval and NN needs to take into account more than one value at different time to predict the fault of a particular component. Hence, we introduced the Recurrent Neural Network (RNN) prediction concept, so that final ANN predicted output spans over a period of time.

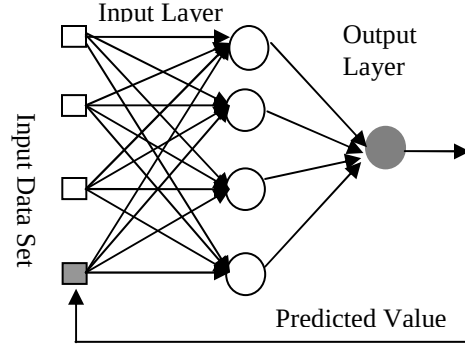


Figure 2: Recurrent NN architecture

The predicted output of NN replaces the oldest input to the neural network, forming a new input training dataset (see Figure 2). This input data set is then multiplied by the updated weights to form a new output. In this way, a series of values is predicted at the end of testing phase. Thus, the final output of the network is calculated by taking the mean of these predicted values. This final output of each network is given to the fuzzy inference system.

2.1.2 Fuzzy logic. Accuracy of different fault predictions in NF model depends on if-then-action rules of fuzzy inference system. In order to accurately predict, it is necessary to find the exact combinations of membership functions of different parameters. Membership functions of fuzzy systems are the maximum and minimum range of the individual parameters. Each predicted parameter provided by NN is the input to the fuzzy inference engine. This kind of model also has the option of user control in addition of the predicted output of the NN, i.e. user has the possibility to enter the values of different parameters for a particular fuzzy predicted model, so the fuzzy inference system can predict the fault required for a particular component.

2.1.3 Neuro Fuzzy model for battery parameter. Battery is one of the most critical components of a vehicle. The life time of batteries can be estimated on the basis of six parameters: ambient temperature, liquid level, quality of plates, vibration, state of health and state of charge.

Prediction model

Based on ambient temperature, liquid level, quality of battery plates, and vibration parameters, there exist eighty-one possible combinations. The assigned values of ambient temperature are: (cold, warm, hot), while those of liquid level, quality of battery plates and vibration are: (low, medium, high). Thus, the output variable has nine membership functions from zero to eight.

The life of battery is greatly influenced by the ambient temperature and the quality of battery plates. The maximum remaining life of battery is predicted when ambient temperature is warm.

The syntax of the if-then-action rules of Fuzzy Inference System used for determining the health of battery is.

```
IF condition_1 AND
   condition_2
   .....
   ..... AND
   condition_n
THEN Action
```

The condition part contains one of the possible values of input parameter and the action part contains the resulting value of output parameter.

Examples of rules used for the prediction of health of battery are given below:

Brakes_Rule1: IF Ambient Temperature is warm AND Vibration is low AND Liquid Level is high AND Quality is high THEN Reliability is eight

Brakes_Rule2: IF Ambient Temperature is cold AND Vibration is high AND Liquid Level is low AND Quality is low THEN Reliability is zero

Brakes_Rule3: IF State-of-Charge is high AND State-of-Health is high THEN Output is six

Brakes_Rule4: IF State-of-Charge is low AND State-of-Health is low THEN Output is two

3. Neuro-Neuro architecture

We designed three layers of a Neuro-Neuro architecture: data, prediction, and health calculation layer. Data layer extracts the sensor values of parameter from log files. This information is the input data for the NN for the prediction layer. This layer consists of individual neural networks for each parameter involved in health prediction. The health calculation layer calculates the remaining number of days of component's health, depending on the predicted values.

3.1. Neuro-Neuro model

Neuro-Neuro model consists of three layers (see Figure 3). In data layer, the required sensor data related to prediction is acquired. This information is saved into a file which is used for training the NN in the prediction layer. In the prediction layer there exists a NN for each parameter. This NN is trained using history data acquired in the data layer, and it is trained by Back-propagation algorithm. The data layer generates the input-output patterns. There exists a disadvantage of overfitting with Back-propagation algorithm [4]. To overcome this problem, we divided training data into different training patterns. Therefore, NN is trained on half number of patterns. It might affect the prediction capability of neural network. Hence, it is recommended to use maximum 15 numbers of patterns for training. Usually, a NN spends more time during training. Hence we introduced momentum factor [4] which improves the convergence of the Back-propagation algorithm. Thus, the weight modification is not only dependent on the present input, but also on the modification of the previous weight.

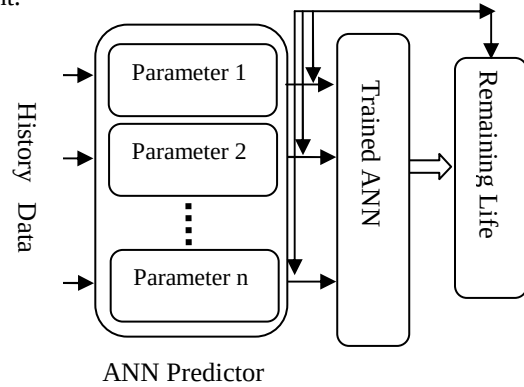


Figure 3: Neuro-Neuro architecture

Once the NN is trained with the previous input-output patterns, it is ready to predict next values in the series with similar interval of input data series. The user controls the prediction of any number of values in the series. It is recommended that the number of values to be predicted should be equal to the number of values in the input pattern. The next values in the series will be similar to previous predicted values.

The calculation layer uses a NN for predicting the health of a component (number of days). In Neuro-Fuzzy architecture fuzzy logic is implemented to predict the health of a component. Several inference rules and membership functions was implemented to predict the health.

Similarly to the prediction layer neural network, the health prediction layer is trained using Back-

propagation algorithm. The training data of this NN come from previous NN which is already available. Each value in the input pattern designates the values of a single parameter used for the prediction. The output value represents the number of days. Once training data is created in the form of input-output pattern, the NN is ready for the group training. When a NN is trained, predicted values are read from stored files and the remaining health of a component is calculated.

3.2. Implementation of NN architecture

The Neuro-Neuro model uses Neural Networks both for prediction of future values and for calculating the remaining life of component. This model has layered architecture consisting of three layers (see Figure 4).

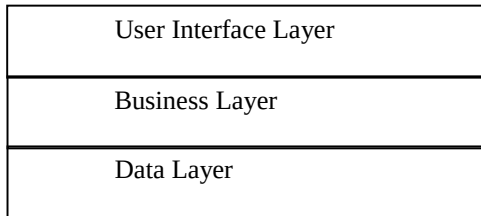


Figure 4: Layered architecture of NN model

The user interface layer consists of parameters i.e. training visualization, training data, predicted values, which are visible to user. During training of a NN, user can observe the progress of training through individual pattern error, group pattern error, expected output and actual output of output NN. User can edit training data through accessing training data files. The weights of trained NN and predicted values are visible to user.

In the business layer, we have implemented Back-propagation algorithm to train each NN. For prediction of series of number of values as per user requirement we have used recurrent neural network. It is important to maintain the accuracy of the system. Hence, NN for each parameter has only one output. Thus, after group training using history input-output patterns each NN can predict only one value. It improves the prediction accuracy of each NN.

In the data layer, all the data related to training i.e. input-output patterns, scaled input-output patterns, weights, parameters, and testing i.e. predicted values are managed. Each data is written in text format so that user can easily understand and update these data. The training patterns and predicted values are rescaled before saving.

3.3. Health prediction using NN architecture

We have developed this model to provide dynamic behavior to the prediction tool. This prediction tool eliminates drawbacks from Neuro-Fuzzy model. Because of its dynamic behavior it can adapt itself for health prediction of any component. In the following section we discuss battery health prediction.

3.3.1 Battery health prediction. After studying working of battery and their faults, we found that following parameters are important to predict health of battery.

- **Temperature:** It is the internal temperature of the battery. If the internal temperature is very high due to some reason it will dry out the liquid inside battery very quickly resulting in failure of battery. So it is an important parameter for battery health prediction.
- **Liquid level:** The level of liquid inside battery has a strong effect on remaining life of the battery so, it was important to include this parameter for battery health prediction.
- **State of Health (SoH):** The percentage of maximum possible charge that is present inside the battery [10], i.e. (available capacity) / (maximum attainable capacity). If the SoH is not enough, the battery will die soon. This is why it is an important parameter for battery health prediction.
- **State of Change (SoC):** This measurement reflects the battery general condition and quality to deliver the specified performance compared to a new battery. Moreover, SoC represents the battery performance compared to the past, present and expected future [10]. It is also an important parameter for battery health prediction.

After completion of training, user can save final weights of neural network. These weights are used in testing mode for prediction of values.

In testing mode, future values are predicted using trained neural network. User has to specify number of input neurons and weight file. In this mode user has to select data file containing present sensor values. This file consists of only input patterns. Using current sensor values and weights, future values are predicted. User can specify number of values to be predicted. According to number of values to be predicted, the RNN will predict series of values. These values are saved in a text file and are used in prediction mode.

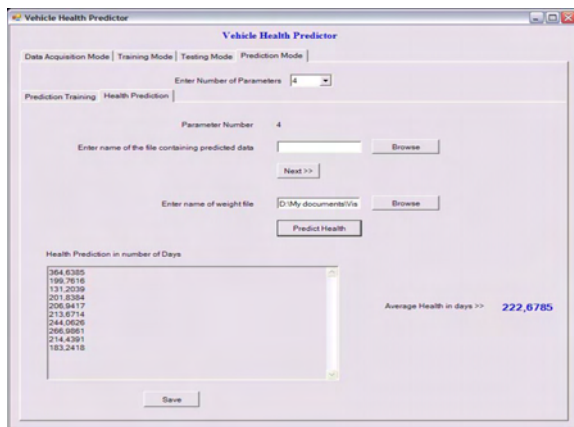


Figure 5: Battery health prediction model

In prediction mode, neural network in health calculation layer is used to predict health of a component in number of days. This mode consists of two modes: training and health prediction. In training mode, user has to connect training file. To scale training patterns from 0 to 1 user has to mention parameter ranges, and error cutoff range to decide about convergence during training.

After completion of training, final weight are displayed and saved into the selected text file. This file is used to predict health of battery. In health prediction mode, user has to attach files containing predicted data for each parameter. The tool will prepare input patterns for prediction. User can connect weight file, saved during training mode. Using these files, health of battery is predicted in number of days for each pattern. It will display average of predicted health in number days along with individual health prediction. User can save these values in text file.

4. NF and NN models: similarities and differences

The health prediction of a component depends on more than one parameter. The second section of both prediction models consists of different techniques: the Neuro-Fuzzy model uses fuzzy logic and the Neuro-Neuro model uses Neural Network.

4.1. Neuro-Fuzzy model

In the prediction layer by neural network the prediction tool is dynamic. Hence, user can select any number of parameters for prediction. The health depending on predicted values is calculated using fuzzy logic. The membership functions depend on parameters used for prediction. The rules are generated

according to the number of involved parameters. Hence, the behavior of this section is static. If we make changes in the number of parameters, the corresponding membership functions and rules should be updated, which is one of the important drawback of the system.

In current scenario, there are some parameters which are not measurable. We have tested this model for available parameter values. The prediction capability of NN depends on data used for training. For some of unavailable parameters we have used self-created data. The NN usually takes more time for training. The fuzzy logic calculates health quickly. If the behavior of data is known and correct rules are generated then fuzzy logic is useful in such prediction tools. Neural networks should be used with fuzzy logic to provide learning ability to the prediction tool. Fuzzy logic provides quick health prediction ability depending on rules and membership functions.

4.2. Neuro-Neuro model

In the prediction layer using neural network, the prediction tool is dynamic. Hence user can select any number of parameters for prediction. NN is used to calculate health depending on predicted values. Hence NN provides dynamic behavior for second part of prediction tool. This is an advantage of Neuro-Neuro model over Neuro-Fuzzy model. The training of each neural network consumes more time in order to start prediction process. This is the major drawback of this model.

5. Related work

Most of the research work is related to the prediction of mean time between failures (MTBF) and/or mean time to failure (MTTF) rather than remaining life time prediction. This paper shows the possibilities for predicting the remaining life time of a component using ANN and Fuzzy Logic by means of different parameters: SoC, SoH, Temperature, Vibration, Liquid Level, Quality of Plates, etc.

Other works such as State-of-Charge and State-of-Health prediction of lead-acid Batteries by B.S Bhangu [8] tries to predict the health of an individual parameter of a component e.g The health of battery depends on State-of-Charge (SoC) and State-of-Health (SoH) of the battery. But some other parameters, for example temperature, vibration, liquid level and quality of plates, are also directly connected to health of battery. Above mentioned work tries to predict the SoC and SoH. Thus, remaining life of battery is not directly predicted, but it helps to predict the individual

parameter that can be used for prediction of remaining life. On the other hand, our model extends this work by providing a general purpose model that uses more than one parameter rather than only one. For example, our model uses five parameters including SoC, SoH, Vibration, Temperature, Liquid Level and Quality of Plates to predict the battery health (remaining life).

In [9], it was tried to predict the failure of hardware components using naïve Bayes classification, a linear classification technique that is based on a generative model. Applying this technique on history data, these data can be classified as good and bad data. If the ratio of bad data is greater than the good data, we can predict that the component may fail in future. Statistical analysis then can be applied to make this prediction more and more reliable. But in this approach all importance is given to classifiers, if the classifier produces incorrect results due to some reasons it will directly effect the prediction. In addition, this research work does not provide a general purpose approach that can be applied to predict the remaining life of any component.

Our approach used ANN to predict the next values in a series. ANN is good enough to handle noisy data. Even though working with noisy data will not directly effect the prediction.

Hence, even though some work exists for prediction of remaining life of component, most of them are related to the prediction of individual parameter and specific failure rather than a general purpose. We have tried to come up with a complete model that can be used to predict the remaining life of a component. Successful practical implementation of this model proves its usefulness.

6. Conclusions and future perspectives

Even though, it is not easy to accurately predict the remaining life of a component by using different available computing techniques, however it is possible to get an idea about the health of a component. This paper presents a different approach to predict the life of a component: users can get idea about the health of a component, and predictive maintenance strategy can be applied in order to know the fault in any component beforehand. After doing research and analysis of different AI techniques, we conclude that ANN is most commonly used and feasible for prediction. Moreover, fuzzy logic is used for decision making and control purposes. That is why; we implemented two architectural models on the basis of these two techniques: Neuro-fuzzy and Neuro-Neuro.

It is possible to predict the number of days of the remaining life of any component, using these models.

In this way, users can take precautionary measure before the component actually breaks down.

Taking into account the results of our two prediction models, it is necessary to explore how to remove the drawbacks of Neuro-Fuzzy model. A faster system will result which could be helpful to design a system for real time environment.

Another important feature, as a future work, is to integrate the prediction system in an electronic control unit, keeping track of current sensor values and predict future series of values in real-time.

7. References

- [1] Luo, J., Namburu, M., Pattipati, K., "Model Based Prognostic Techniques", AUTOTESTCON 2003. IEEE Systems Readiness Technology Conference. Proceedings
- [2] S. Haykin, "Neural Networks A Comprehensive Foundation", Prentice Hall, New Jersey, second edition, 1999
- [3] Pettersson, J., Wahde, M., "Generating balancing behavior using recurrent neural networks and biologically inspired computation methods". IEEE Transactions of Evolutionary Computation, September 2003
- [4] Wahde, M., "An Introduction to Adaptive Algorithms and Intelligent Machines", Göteborg, Sweden, second edition, 2002
- [5] Nürnberger, A., "A Hierarchical Recurrent Neuro fuzzy System", IFSA World Congress and 20th NAFIPS International Conference, 2001. Joint 9th, ISBN: 0-7803-7078-3
- [6] Abraham, A., "Neuro fuzzy Systems State-of-the-art Modeling Techniques", School of Computing and Information Technology, Monash University, Churchill 3482, Australia
- [7] Passino, K.M, Yurkovich, S., "Fuzzy Control", Copyright 1998 Addison-Wesley
- [8] Bhangu, B.S., Bentley, P., Stone, D.A. and Bingham, C.M., "State-of-Charge and State-of-Health prediction of Lead-Acid-Batteries for Hybrid Electric Vehicle using non linear observers", Department of Electronic and Electrical Engineering, University of Sheffield UK.
- [9] Broadwell P., "Component Failure Prediction Using Supervised Naïve Bayes Classifications", <http://citeseerx.ist.psu.edu/viewdoc/summary?doi=10.1.1.3.4.641>
- [10] Daniel C., Cox, Regina P.K., "Battery State of Health Monitoring. Combining Conductance Technology with Other Measurement Parameters for Real time Battery Performance Analysis", copyright 2000 IEEE
- [11] Klawonn F., Höppner F., Kruse R., Runkler T., "Fuzzy Cluster Analysis" Wiley, Chichester, 1999.

Genetic and Hybrid Intelligent Systems

A tool for decision making in manufacturing process flexible

Oscar Báez Senties, Alberto Aguilar Lasserre, Fernando Ortiz Flores, Raúl Torres Osorio and
José Pablo Nuño de la Parra
*Instituto Tecnológico de Orizaba, División de Estudios de Posgrado e Investigación,
Oriente 9 no. 852 Col. E. Zapata Orizaba, Ver. México
{obaiez70, albertoaal, r_exergia, s_lechoux}@yahoo.com.mx*

Abstract

Scheduling of semiconductor wafer fabrication system is identified as a complex problem, involving multiple and conflicting objectives (makespan and minimizing waiting time for instance) to satisfy. In this study, we propose an effective approach based on an artificial neural network technique embedded in a multiobjective optimization loop for multi-decision scheduling problems in a semiconductor wafer fabrication environment.

1. Introduction

Scheduling of semiconductor wafer fabrication system is identified as a difficult task, mainly because of the typical features of the process scheme, such as complex product flows (the so-called wafer fab is indeed a multipurpose plant), high uncertainties in operations, rapidly changing products and technologies [6]. It is thus a significant challenge to develop effective scheduling methods in wafer fabrication. Discrete event simulation (DES) is one of the most widely used methods to study, analyze, design, and improve manufacturing systems. The combined use of a DES and an optimization procedure based on a genetic algorithm was an efficient solution to short-term job-shop scheduling problems and was adopted in our previous works [3].

In spite of its acknowledged benefits, this kind of approach often reaches its limits in the industrial practice because of the highly combinatorial nature of the problem. In addition, the main emphasis of much of the work on scheduling has been on the development of predictive methodologies with a single objective. Actually, production managers have to cope with various objectives, which contributes to scheduling complexity: makespan is an important goal in low-volume and high variety production circumstances within competitive market

environments; another major objective in scheduling of semiconductor wafer fabrication is reducing waiting time for work-in-process (WIP) inventory to improve responsiveness to customers: in addition, the shorter the period that wafers are exposed to contaminants while waiting for process, the smaller the yield loss. Increasing the throughput is also an important stake, since the investment in fabrication equipment is capital intensive.

In this study, we propose an approach based on an artificial neural network (ANN) technique coupled with a multiobjective genetic algorithm (MUGA) for multi-decision scheduling problems in semiconductor wafer fabrication.

1.1. General framework and plant modelling

The main emphasis of much of the work on scheduling has been on static systems with a single objective. As noted above, wafer fabrication is complex, dynamic and highly stochastic. Satisfying the multiple objectives might be more important than only optimally meeting a single objective. In this study, we propose an effective approach based on an artificial neural network technique for multiobjective and multi-decision scheduling problems in semiconductor wafer fabrication. A Data Base (DB) was developed in absence of real data. DB was applied for the learning phase of artificial neural network (ANN). ANN was used for modelling process. A multiobjective genetic algorithm (MUGA) coupled to the ANN, is then used to optimise the decision variables and to obtain the feasible values. Figure 1 shows the role of the proposed methodology.

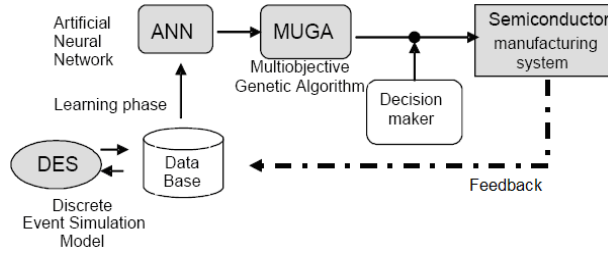


Figure 1. General methodology

2. Process description

Taking into account the diversity of the products manufactured and processes implemented, there are a significant number of stages and the durations of treatment is highly variables. The process consist in 14 different equipment units used (from which 12 are different) in a process involving 24 operating stages. To answer at the request of the market, the workshop has, for the majority of the stages of manufacture, several equipment in parallel able to carry out the same operations. Two shared equipment units are used up to 6 times in the manufacturing sequence, to reproduce the so-called re-entrant flow. Some units are in parallel, which confers more flexibility to the process.

3. Discret – Event Simulator

Simulation modeling has become an indispensable tool in the effort to accurately assess the true capacity of a facility, allowing manufacturers to more confidently commit the full capacity of the fab. Simulation modeling has been widely used in the semiconductor industry for a wide range of strategic objectives such as fab design, equipment selection, capacity and cycle time planning, etc. [8]. Its usage in the tactical area has been relatively limited. The pertaining areas include short-interval scheduling, dispatching of individual lots, equipment events scheduling, short term deployment of operators, etc. To model the wafer fab in a high degree of detail, Discrete Event Simulation (DES) techniques were previously implemented, leading to the development of SIMNET Industrial Software of Simulation Network (marketed by the University of Arkansas). Process planning simulations evaluate assignments of jobs to machines and routings for those jobs through the shop. Scheduling simulations seeks solutions to daily issues including on-time order completion, priority changes, and unexpected changes in resource availability. Typical events taken into account and managed in the simulation core have been widely presented by Bérard [2]. The different runs performed allow identifying the

more sensitive variables on some performance criteria, such as makespan, cycle time, limitation of WIP, etc.

3.1. Inputs – Outputs data of SIMNET

This section describes the data set that will be used in the simulation for then training a typical multilayer perceptron (MLP) neural network. There are multiple criteria that can be used in evaluating the system performance and system status of the semiconductor fabricator. The criteria are mainly based on completion times, inventory level, or machine utilization. Three criteria were selected here, involving the computations of:

Mk: Makespan, representative of the total time of duration of production.

AWT : Average wait time of all products (n) at each processing unit (WT_i refers to the waiting time of product i).

TS: Total storage is about the total number of the all products which one waited the release of the following equipment

Table 1. Decision variables

Decision variable	Number and type of parameter
Maximum number of recipes	5 (equivalent to product families)
Maximum number of recipes treated simultaneously (NR)	ex. 4 among the 5 available recipes
Identification of family sequence (FS)	24 combinations, ex : 4,3,1,2 Batches are always released with this sequence order into production
Total of batches (NP)	3 discrete values (16, 24, 32) ex : 16 [4 4 – 3 3 – 1 1 – 2 2] 8 Products [4 4 – 3 3 – 1 1 – 2 2] 8 Products
Number of campaigns (NC)	3 discrete values (2,3,4) ex : 2 This variable sets the number of batches within a campaign [44 33 11 22] [44 33 11 22] [campaign 1] [campaign 2]
Time Between Batches (TBB) (identical from a batch of two products to the following two, whenever the campaign)	3 discrete values (120,750,1250) ex : [campaign 1] [campaign 2] [4 4 3 3 1 1 2 2] [4 4 3 3 1 1 2 2] ↺ ↺ ↺ ↺ TBB = 120 min
Time between Campaigns (TBC)	5 discrete values (500,1500,1900,2400,2800) ex: [4 4 3 3 1 1 2 2] campaign 1 Separated by TBC = 500 min from [4 4 3 3 1 1 2 2] campaign 2

We consider the production of 5 different families of products with their associated recipe identified by a number (1, 2, 3, 4, 5). This workshop must ensure a production varying 4 of 5 products. For each stage are specified: the zone of membership, the name of the stage, the equipment concerned and processing times of each product. A batch is constituted by 50 wafers and 3 levels of productions are to be considered according to market demand, i.e., 16, 24 and 32 batches. The definition of each decision variable is proposed in Table 1, where an example is developed. Each wafer batch is identified by a number corresponding to its family. A campaign is constituted by the release of several products belonging to various family sequences into the wafer fab. In order to make 5400 simulations a software interactive with SIMNET was developed to the source code in C⁺⁺ language.

4. ANN Modeling

Historically, lengthy development and execution times made simulation models impractical to use in this type of cases. This lack is the primary reason to use simulation for the training of neural network model. We then resort to an artificial neural network technique [11] to model the semiconductor plant, since its efficiency has been successfully demonstrated in semiconductor processing by many researchers [8], [11].

The neural network stores the information in the strength of the neuron interconnection through so-called weights. Because the weights of the link between the neurons can not be pre-determinate for a large scale neural network, the learning ability is necessary for a neural network to adjust the weights during a learning phase in which all the examples are presented to the ANN repeatedly. Tan-sigmoid transfer function (tan-hyperbolic) was used as an activation function for hidden and output layers.

After the learning phase the ANN can be used to obtain the answer to an input pattern. If a neural network is developed successfully after training, good results can be archived with a low processing time and small memory resources. For this purpose, the MATLAB software package dedicated to neural network modelling was used [7]. The neurons of the input layer correspond to the wafer manufacture process parameters (decision variables). The output neurons correspond to the wafer manufacture performance parameters to be evaluated (criteria). The same network structure was adopted for the combination: 2 of 4 criterion. The network also includes one hidden layer, which helps the network in

learning the non-linear mapping between the input and output layers.

The mean square error (MSE) was chosen as performance function for supervised training. As each input is applied to the network, the network output is compared to the target. We want to minimize the average of the sum of these errors. The MSE algorithm adjusts the weights and biases of the linear network so as to minimize this mean square error. In essence, the MLP training procedure aims at obtaining an optimal weight set that minimizes a pre-specified error function. The error is computed as the difference between the target and network output values. The error function is the MSE and error of learning ($e(k)$) are defined as:

$$MSE = \frac{1}{Q} \sum_{k=1}^Q e(k)^2 \quad (1)$$

$$e(k) = y_i(k) - g_i(k) \quad (2)$$

where Q refers to the number of input-output example patterns used in training, i the index of the example pattern (vector) and, $y_i(k)$ and $g_i(k)$ the desired (target) and MLP predicted values of the k^{th} output node, respectively. The “network overfitting” occurs when: the network is training excessively (overtraining) and/or the network architecture has more hidden nodes than necessary (overparametrization). There are different procedures for obtaining optimal network architecture [9].

The experimental results of criteria (Mk and TS) have been used for the learning phase of the neural network. The most popular and successful learning algorithm used to train multi-layers networks is the back propagation (BP) scheme. Like the quasi-Newton methods, the Levenberg-Marquardt algorithm was designed to approach second-order training speed without having to compute the Hessian matrix.

The total combination set of decision variables was generated to build the neural net-work, i.e. 5400 (training and test data, 2:1) at the preliminary stage of the study. For search number of neurons were used the minimum value of root mean square error (RMSE).

$$RMSE = \sqrt{\frac{1}{Q} \sum_{k=1}^Q e(k)^2} \quad (3)$$

The results the RMSE for Mk of Training and Test was estimated 0.0471 and 0.0310, respectively. Besides, for TS the results were estimated: Training 0.0643 and Test 0.0797.

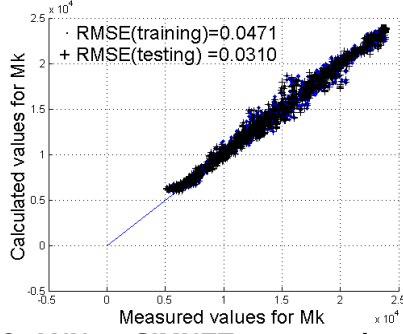


Figure 2. ANN vs SIMNET computed values for Mk criteria.

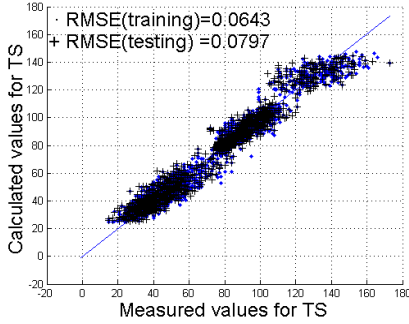


Figure 3. ANN vs SIMNET computed values for TS criteria.

And the correlation coefficients R of individual for Mk output variables was calculated for Training and Test are 0.993 and 0.981, respectively. For TS the values of R are Training 0.993 and Test 0.978. The same procedure has been used for to create the different MLP models, in the combination of the criterion : Mk-AWT and Mk-TS. All these models were based on the multilayer perceptron architecture [4]. Figures 2 and 3 show the comparison between ANN and SIMNET computed values for Mk and TS criterion.

5. Multiobjective Optimization

The neural networks have then been embedded in a multiobjective genetic algorithm (MUGA) to optimize the decision variables and to deal with the set of compromise solutions for the studied criteria, thus giving the optimal Pareto zone solutions [1]. Lately, there has been a large development of different types of multiobjective genetic algorithms, which are reflected in the literature. Following the guidelines proposed by Dietz [5], this approach was adopted in this work. In the MUGA procedure, the search for an optimal solution (decision) vector begins from a randomly initialized population of probable (candidate) solutions. In the ANN/MUGA procedure, the search

for an optimal solution (decision) vector begins from a randomly initialized population of probable (candidate) solutions.

The size of the initial population was 250 individuals, which is a trade-off between an efficient searching process and the avoidance of premature convergence. This initial population was chosen randomly within the assigned input constraints.

Binary codes were used to code the nonbinary parameters. The number of bits in the binary string was set to 6 for each of the six input variables. This is adequate judging from required precision in defining the numerical values of each input variable. The results are the outputs function of MLP. They were compared to give the ranked fitness values (perform). The results were obtained for the entire population. We used the four typical genetic operators to alter the composition of the offspring in the next generation, based on the preset probability values [10]. These included the following.

1. Selection operator. Based on the ranked fitness values, a selection of a new population was made with respect to the probability distribution, i.e. the Roulette-Goldberg method.

2. Crossover operator. Recombination to the individuals in the new population was applied. The probability of crossover was set 0.5.

3. Elitism. This rate complementary of the rate of crossing. The elitism was used to avoid the loss of the best current solution at the time of the passage of one generation to the following. It was set to 1 (the best solution by generation).

4. Mutation operator. Performed on bit by bit basis. The probability of mutation was set to 0.4.

Cyclic repetition of the above steps, the algorithm was stopped after a fixed number of generations (G) specified in the program. In our case G was set to 1000. A procedure of sorting (TRI) was used in all the individuals evaluated. This procedure finds the feasible solutions values of the function objective, but correspondent to individual differents. Finally, all the optimal solutions are obtained. For this strategy as MLP neural network as MUGA were developed to the source code in C++ language. The figure 4 shows procedure for the GA-based optimization of an ANN.

The big advantage of genetic algorithms over other methods, particularly over other stochastic procedures such as Simulated Annealing, is that a GA manipulates a population of individuals. It is therefore tempting to develop a strategy in which the population captures the whole Pareto front in one single optimization run.

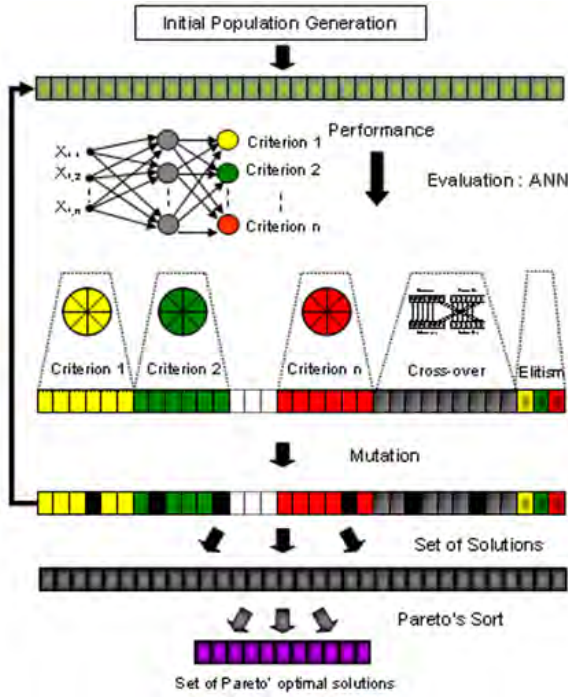


Figure 4. Combination between neural networks models and multiobjective genetic algorithm.

6. Results and discussion

The total time of manufacture is between 5101 and 23900 min. The MLP was first used to model the two criteria Mk and TS. The results obtained show the typical compromise between Mk and inventory in process. Since the conflicting behavior between each pair of criteria (Mk, TS and AWT) was demonstrated, the final multi-criteria Mk-AWT was carried out keeping the two manufacturing criteria independent: this simply means that the same survival rate was considered for each criterion (figure 5). The MUGA was second used to demonstrate that the two criteria considered, represent conflicting goals.

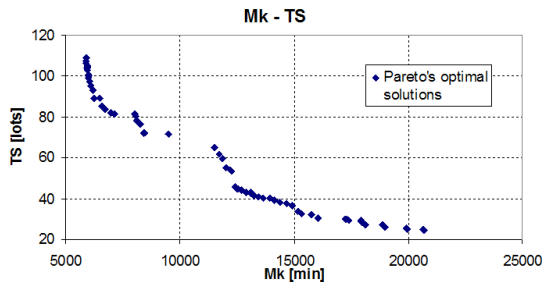


Figure 5. Pareto's optimal solution Mk-TS

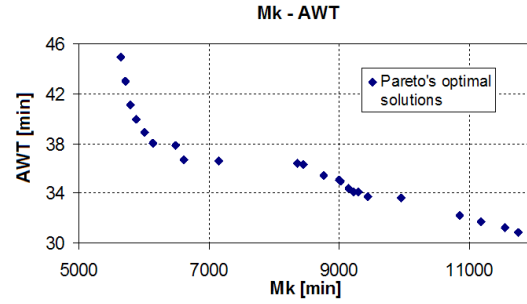


Figure 6. Pareto's optimal solution Mk-AWT

Figure 6 shows the results obtained performed with an identical parameter set to guarantee the stochastic nature of the GA. Let us note that the Pareto zone is constituted of sparse points, since the adaptation function related to the criterion takes discrete values.

The solutions are the reference which the decision maker can use this information, to select the best alternatives, which will succeed in increasing the performance of the real state of the system. Time average necessary to find the solution optimal in the procedure for two criteria was 13.54 min.

The ANN/MUGA hybrid strategy for three criteria is possible to consider simultaneously. We utilized 28 neural networks hidden and 1000 epochs. The possible solutions were calculated in this case. Within this information the decision maker has more elements to decide on his choice without SIMNET. If the decision maker want to obtain approximated value of Mk = 23655.40 min, AWT = 47.58 min and TS = 35 lots, it will be necessary to use a inter arrival time of 1250 min, to manufacture 4 types of products 1, 2, 5, 4, in 4 campaigns by 32 products and finally the order of pass will be 1,2,5,4. Applying the optimized result from table 2 to the SIMNET simulator, we calculated the criteria values. The result is shown in table 2.

This type of results must make it possible to the decision maker to determine the parameters leading to the most adequate decision for the supervision of the manufacture of electronic components. The behavior of the tri – criterion, between Mk-AWT-TS is shown in figure 7.

Table 2. Statistical parameters

Criteria	ANN/ MUGA	SIMNET	ANN/ MUGA	SIMNET
	Mean [min]		Standard deviation [min]	
Mk	13368.01	13737.75	5677.82	5641.27
AWT	62.25	60.25	25.17	27.24
TS	46.18	48.69	11.62	16.77

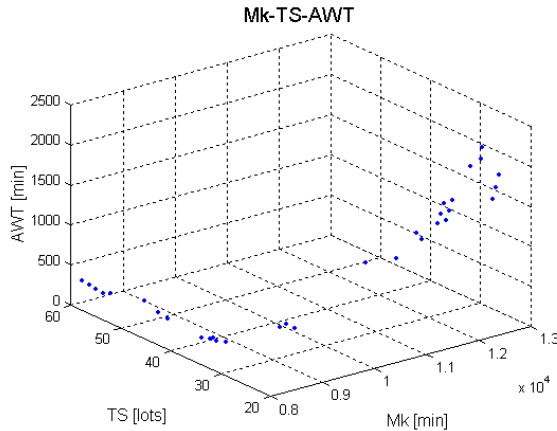


Figure 7. Pareto's optimal solution Mk-AWT-TS

7. Conclusions

Semiconductor wafer fabrication involves an important number of decision problems. The objective of this paper was to propose an optimization strategy in order to assign appropriate decision variables. More precisely, a scheduler for selection of decision variables in order to obtain desired performance measures at the end of a certain production interval was developed. In the proposed methodology, a three-level strategy based on the combined use of a simulation technique, a neural network and a multiobjective genetic algorithm is suggested to optimize functions objectives for the wafer fab. The ANN/MUGA hybrid method presented like a tool of decision making aid in the supervision of the development of electronic components. A simplified example was utilized considering three criteria: makespan, total storage and average of times of storage. The results indicate that this methodology is an effective method considering the complexity of semiconductor wafer fabrication systems, as a time saving way to achieve a prompt response in a dynamically changing environment.

8. References

[1] Baez O., Azzaro-Pantel C., Pibouleau L., Domenech S., Davin A. Aide à la décision pour la supervision d'un atelier discontinu: application à la fabrication de composants

électroniques. Récents Progrès en Génie des Procédés, Numéro 92 – 2005, Toulouse.

[2] Bérard F., Azzaro-Pantel C., Pibouleau L., Domenech S., Navarre D., Pantel M, 1999, Towards an incremental development of discrete-event simulators for batch plants: use of object oriented concepts, Comm. Escape 9, Budapest, (Hongrie) 31 Mai - 2 Juin, 1999.

[3] Charles A.S., Floru I. R., Azzaro-Pantel C., Pibouleau L., Domenech S., Optimisation of preventive maintenance strategies in a semiconductor batch plant, Computers and Chemical Engineering, 27(2003) pp 449 – 467.

[4] Chen S., Billings S.A. Neural networks for non linear dynamic system modeling and identification, International Journal of Control 56 (2) (1992) 319-346.

[5] Dietz A., Azzaro-Pantel C., Pibouleau L., and Domenech S., 2005a, A Framework for Multiproduct Batch Plant Design with Environmental Consideration: Application To Protein Production, Industrial Engineering and Chemistry Research, 44, pp. 2191-2206.

[6] Ellis K.P., Lu, Y. Bish E.K., 2004, Scheduling of Wafers Test Processes in semiconductor Manufacturing, International Journal of Production Research, 2004, Vol. 42 No. 2, 215-242.

[7] Matlab, Neural Network Toolbox, User' Guide, Inc., 2001.

[8] Min H.S., Yih Y. Development of real-time multiobjective scheduler for a semiconductor fabrication system, International Journal of Production Research, 2003, Vol. 41 No. 10, 2345-2364.

[9] Nandi S., Ghosh S., Tambe S.S., Kulkarni B.D. Artificial Neural Network Assisted Stochastic Process Optimization Strategies. AIChE J. 2001, 47, 126.

[10] Pibouleau L., Domenech S., Davin A. and Azzaro-Pantel C. Expérimentations numériques sur les variantes et paramètres de la méthode du recuit simulé. Chemical Engineering Journal, Volume 105, Issue 3, 15 January 2005, Pages 117-130.

[11] Sung C.S., Choung Y.I. A neural approach for batching decisions in wafer fabrication, International Journal of Production Research, 1999, Vol. 37 No. 13, 3101-3114.

Combining Genetic Algorithms and FLDR for Real-Time Voice Command Recognition

Julio César Martínez Romo, Francisco Javier Luna Rosas,
Instituto Tecnológico de Aguascalientes,

jucemaro@yahoo.com,

fjluna@ita.mx,

Miguel Mora-González
Universidad de Guadalajara
(CULagos)
mmora@culagos.udg.mx

Abstract

In this article we propose an effective method for a user-dependant voice command (small vocabulary) recognition system based on the combination of genetic algorithms and the Fisher's Linear Discriminant Ratio (FLDR). A genetic algorithm here is used to search in the frequency domain of the voice for those sub-bands whose energy is discriminant enough so as to distinguish between at least two different classes, and at the same time, being able to appropriately agglomerate the different utterances of one word of the vocabulary in a compact class. Once the sub-bands have been evolutionary selected, its energy is represented in feature vectors; in this way, very few samples of each voice command are required to build each words' model; moreover, this is a convenient method for feature selection. Real time implementation was done in a DSP TMS320LF2407 using elliptic bandpass filters –one per sub-band- with floating point representation. Encouraging results were obtained.

1. Introduction

Automated speech recognition is a complex task to be solved, with many practical applications. The nature of speech signals is rather complex, given that even as few as two utterances of the same word said by the same person do not produce exactly the same waveform, neither in amplitude nor in time; however, certain properties of such waveforms are similar enough so as to be recognized.

In the last decades [1], many efforts have been done in order to find methods for reliable speech recognition; some of such methods were derived from probability theory (i.e., Hidden Markov Models (HMM), Gaussian Mixture Models (GMM), etc.), from artificial intelligence (neural networks), and dynamic programming, such as DTW (dynamic time warping). Many researchers have created interesting hybrids from these fields to improve some aspect of the speech

recognition chain. A survey of the development of speech recognition methods and principles can be found in [2].

Nowadays, real time voice recognition systems are the fashion in many commercial electronic devices such as cellular phones, automated answering machines, and others. In such cases, the specific topic addressed is that of speech recognition with a small vocabulary scenario, in which a few “commands” are used to several purposes, for instance, navigation within a voice-driven menu. For applications like this one, a very robust method for waveform modeling can be constructed to lower the classification error between different words.

Genetic Algorithms (GA) have been previously used in the field of speech recognition as an optimization technique because of its ability to search globally the solution surface of the problem; in general, the idea has been to improve the performance of some recognition paradigm already in use; for instance, in [3] Kwong and other used a genetic algorithm to train a HMM in order to obtain a minimum classification error (MCE). In [4], Lan, Pan, and Lai, used the genetic algorithm to replace the steepest descent method for the training of a BPNN, obtaining a 91% of correct classification in the speaker independent problem. In [5], Kwong used a genetic algorithm to overcome certain limitations of the DTW method, such as the nontrivial computation of the K-best paths, or the stringent rule on slope weighting; he used two derivatives of the original GA: the hybrid GA and the parallel GA. In all the cases, the introduction of the GA improved the accuracy of the recognition, with the side effect of increasing the computational effort to reach the solution. A variation in the application of GA as previously described consists not in improving the training itself, but in selecting the optimal structure of the feature extractor or classifier; for instance, in [6] Takara, Higa, and Nagayama, used the GA in order to select the structure of a HMM for the problem of isolated word recognition.

Fisher's Linear Discriminant Ratio (FLDR), is used to measure class separability in the features space [1]; the bigger this ratio is, the more discriminant is the set of features involved in its calculation, and vice versa, thus providing better class separability; it is of application in l -dimensional as well as to multiclass problems.

In this paper, we propose the use of genetic algorithms in combination with multiclass-class (FLDR) to provide a robust and reliable means for spoken word recognition (or voice commands). The hypothesis in this method relies on the fact that the frequency contents of a limited set of words are sufficient to distinguish one from another, and that at the same time consistent enough so as to recognize precisely each word, with no further time-scale or time-frequency analysis required. This approach is to address the problem of recognizing a small vocabulary (2 to 10 spoken words), from one individual (mono-user), with training.

The genetic algorithm is used here to select the bandwidth (BW) and central frequency (Fc) of a set of bandpass digital filters; the FLDR is used in the cost function of the GA in order to increase both, class separability and compactness between classes ω_0 and ω_1 , being ω_0 the class of the word to be recognized at the moment and ω_1 the class of the rest of the words in the vocabulary. The features selected were the energy of the sub-bands, thus arriving to feature vectors with optimal class separability for the specific user and vocabulary. Our approach was implemented at the simulation level, showing a correct classification rate of 96%; also, it was implemented on a TMS-320LF2407 digital signal processor using bandpass elliptic filters [2], [8], with floating point, demonstrating real-time capability.

The rest of the paper is organized as follows. Section 2 is devoted to depict the concepts that are the foundation of this paper. In section 3, the proposed method is explained. Results and conclusions are drawn in section 4.

2. Background

2.1. Genetic Algorithms

In this section a brief description of a simple genetic algorithm is given. Genetic algorithms are based on concepts and methods observed in nature for the evolution of the species. Genetic algorithms were brought to the artificial intelligence arena by Holland and has been widely supported by Goldberg [7]. GAs apply certain operators to a population of solutions of the problem to be solved, in a such a way that the new

population is improved compared to the previous one according to a certain criterion function J [7]. Repetition of this procedure for a preselected number of iterations will produce a last generation whose best solution is the optimal solution to the problem.

The solutions of the problem to be solved are coded in the chromosome and the following operations are applied to the coded versions of the solutions [7], [1], in this order:

Reproduction: Ensures that, in probability, the better a solution in the current population is, the more replicates it has in the next population,

Crossover: Selects pair of solutions randomly, splits them in a random position, and exchanges their second parts.

Mutation: Selects randomly an element of a solution and alters it with some probability. It helps to move away from local minima.

Besides the coding of the solutions, some parameters must be set up:

N , number of solutions in a population. Fixed or varied.

p , probability with which two solutions are selected for crossover.

m , probability with which an element of a solution is mutated.

The performance of the GA depends greatly on these parameters, as well as on the coding of the solutions in the chromosome. The solutions can be coded in some of the following formats:

Binary: Bit strings represent the solution(s) of the problem. For instance, a chromosome could represent a series of integer indexes to address a database, or the value of a variable(s) that must be integer, or each bit could represent the state (present-absent) of a part of an architecture that is being optimized, and so on.

Real valued: The bit strings represent the value of a real valued variable, in fixed or floating point.

The aspect of one chromosome could be like this: $C = \{00010101010101\}$.

2.2. Fisher's Linear Discriminant Analysis

Fisher's Linear Discriminant Analysis, is used as an optimization criterion in several research fields, including speech recognition, handwriting recognition, and others [7]. Consider the definitions of "Within-class scatter matrix", "Between-class scatter matrix" and "Mixture scatter matrix" are given in equations (1-3):

$$S_w = \sum_{i=1}^M P_i S_i, \quad (1)$$

$$S_b = \sum_{i=1}^M P_i (\mu_i - \mu_0)(\mu_i - \mu_0)^T \quad (2)$$

and

$$S_m = E[(x - \mu_o)(x - \mu_o)^T] \quad (3)$$

Where S_i , P_i , traces $\{S_w\}$ and $\{S_b\}$, S_m and $E[\cdot]$ are the covariance matrix for class ω , a priori probability of class, a measure of the average variance of the features (descriptive elements of the class), a measure of the average distance of the mean of each individual class from the respective global value, the covariance matrix of the feature vector with respect to the global mean and the mathematical operator of the expected mean value, respectively. And μ_o is the global mean vector

$$\mu_o = \sum_i^M P_i \mu_i \quad (4)$$

Based on the just given definitions, the following criteria can be expressed:

$$J_1 = \frac{\text{trace}\{S_m\}}{\text{trace}\{S_w\}}, \quad (5)$$

$$J_2 = \frac{|S_b|}{|S_w|} = |S_w^{-1} S_b|$$

It can be shown that J_1 and J_2 take large values when the samples in the 1-dimensional are well clustered around their mean, within each class, and the clusters of different classes are well separated. Criteria in eq. 5 can be used to guide an optimization process, since they measure the goodness of data clustered; the data to be clustered could be the set of features representative of the items of a class.

3. Voice Command Recognition via GAs and FLDA

3.1. Overview

The objective of this work is that of isolated word recognition, with a small vocabulary, user dependant and with training. The goal is to make it real-time, using just a few training samples per word to be recognized, in order to avoid annoying the potential users with large training sessions. The general application is command voice recognition, meaning that with a lexicon with as few as 4 to 8 words a user can achieve home automation, voice-driven menu navigation, wheel-chair control, and so on. By real-time, it means that the recognition will take a just a few milliseconds.

Many traditional approaches for speech recognition take a long time to be executed, or require a lot of voice samples in the training phase to make it reliable, such as dynamic time warping (DTW) or Hidden Markov Models (HMM), respectively [1]. Here, to

achieve real-time execution, a set of tunable IIR digital filters are used; the parameters of the digital filters are calculated exploring simultaneously the frequency contents of all the words in the vocabulary in order to discover which sub-bands can be used to distinguish between different words and, at the same time, being capable of allowing good clustering of the utterances of each word in the vocabulary. Good clustering means that the words are separable in the features space. The search is done by a genetic algorithm, and the Fisher's Linear Discriminant Analysis is used to lead the search (cost function), effectively measuring the goodness of the sub-bands selected by each chromosome in the evolutive process. The parameters of the sub-bands are encoded within the chromosomes by pairs, one datum being the central frequency and the other one being the bandwidth; this way, the chromosome will consist of as many pairs as sub-bands to be searched. At the end of the genetic process, the final result is the set of sub-bands that provide the best intra-class grouping and inter-class separation, according to J_2 in eq. 5.

The discriminant feature used here is the energy of the signal within each sub-band. Thus, for each utterance of a word, a feature vector of length N (N being the number of sub-bands) will describe that word. To calculate the energy, the magnitude of the spectrum is used.

The classifier used is a simple one: the minimum distance classifier; the distance metric is the Mahalanobis distance, which takes into account the shape of the classes in the features space, allowing ellipsoidal clustering.

In the sequel, mathematical and configuration details of the process are given.

3.2. Characterization of the frequency spectrum using sub-bands

Consider the Fourier Transform of a signal:

$$S(\omega) = F(s(t)) \quad (6)$$

Where $S(\omega)$ is the Fourier Transform of the time signal $s(t)$.

Now consider the frequency spectrum split in sub-bands, as shown in fig. 1. Please notice in fig. 1 that $S(\omega)$ has been normalized to unitary amplitude. For each sub-band, the energy can be calculated as

$$E_i = k_0 \sum_{lb}^{ub} (S(\omega))^2 \quad (7)$$

Where E_i is the energy of the i -th sub-band, k_0 is a constant proportional to the bandwidth (BW_i) of the i -th sub-band, lb and ub are the lower and upper bounds of the i -th sub-band. The feature vector of n -th utterance of $s(t)$ in the frequency domain becomes:

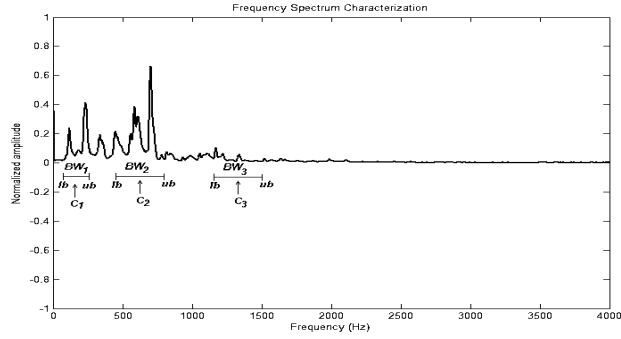


Fig. 1 Characterization of the magnitude of the frequency spectrum of a signal $s(t)$.

$$S_{r_n}(\omega) = [E_1 E_2 E_3 E_4 \dots E_M] \quad (8)$$

In which M is the number of sub-bands, and S_{r_n} is the reduced version of the n -th $S(\omega)$. In order to characterize a word in the vocabulary, N samples of $s(t)$ must be entered. From the example in fig. 1 and without loss of generality, the set of parameters of the respective filter bank to operate in the time domain is:

$$BF = [C_1 BW_1 C_2 BW_2 C_3 BW_3] \quad (9)$$

The number of filters to be applied in the time-domain signal in this case is 3, but in practice it could vary from 4 to 12. For a vocabulary of K words (K small), the spectra in which the solution must be searched is given by K matrices of N rows and 2000 columns.

3.3. Configuration of the Genetic Algorithm

3.3.1. Coding the Chromosome. The chromosome is comprised of the parameters of the filter bank described in eq. 9. The number of sub-bands is fixed, and each one of the centers and bandwidths are subject to the genetic algorithm. Real numbers are used.

3.3.2. The Cost Function. The cost function is given by eq. 5, criterion J_2 . The goal is to maximize J_2 as a function of the centers and bandwidths.

3.3.3. Restrictions. Only two restrictions apply: sub-bands overlap ≤ 50 Hz. And bandwidth is limited to range from 40 to 400 Hz, varying according to the performance of the genetic algorithm.

3.3.4. Operating Parameters of the Genetic Algorithm. The main parameters of the genetic algorithm are summarized in table 1. The values of the parameters are given according to the best results obtained by experimentation. The genetic algorithm was run in Matlab®, using the genetic algorithms

Table 1. Main Parameters of the Genetic Algorithm.

Parameter	Value
Population	80-120
Mutation	Gaussian (1-2, 1-2) (scale, shrink)
Selection	Tournament, size 4
Crossover	Scattered, One-point and Two-point

toolbox and the *gatool* guide. The nomenclature of the parameters in table 1 is the one used by Matlab®.

3.4. Recognizing voice commands

To make operational the methodology described so far, the following steps must be done sequentially:

1. *Vocabulary definition.* 2 to 10 words. In many real life applications, 6 to 8 words do the job.
2. *Database acquisition.* 15 to 20 utterances of each word from the vocabulary, for learning purposes. Sampling frequency can be set from 6000 to 8000 Hz.
3. *Running the GA.* Run the GA to find the sub-bands whose J_2 (eq. 5) is the maximum.
4. *Filters realization.* For each sub-band, compute the coefficients of the respective bandpass filters. For real-time implementation, order from 4 to 8 is recommended, type IIR, elliptic. Elliptic filters achieve great discrimination and selectivity.
5. *Modeling the commands.* To make comparisons and therefore classification, a Gaussian statistical model of each word is to be constructed for each command in the vocabulary. Proceed as follows for each command:
 - a. Construct a matrix C of 15-20 rows of $S_{r_m}(\omega)$ and M columns (one row per sample of the command, one column per sub-band selected by the AG),
 - b. Compute the mean value overall the samples, to find the average energy per sub-band, this is the feature vector of the command (μ_i)
 - c. Compute the covariance matrix of $\Sigma_i = \text{cov}(C_i)$.
 - d. At any time, the Mahalanobis distance between the model of the i -th command and the feature vector x of an incoming command is:

$$dM(\mu_i, x) = (\mu_i - x) * \Sigma_i^{-1} * (\mu_i - x)^{-1} \quad (10)$$

To make the real-time implementation, a digital system must be sampling the input microphone. Each sample of a command must be filtered by each filter (sub-band extraction). The integral of the energy of the signal outgoing from each filter over the period of the command is used to create the feature vector of that command, i.e., x in eq. 10. The feature vector is compared to each model in the vocabulary, and the command is recognized as the one with the shortest d_M . This is the so-called “minimum distance classifier”.

4. Results and Conclusions

To asses the performance of the proposed recognizer, two vocabularies were designed, one in Spanish and the other one in English.

4.1. Results in a Spanish Vocabulary

4.1.1. Definitions and set-up. The vocabulary $V_s = \{\text{'uno'}, \text{'dos'}, \text{'tres'}, \text{'cuatro'}, \text{'cinco'}\}$. 8 sub-bands were used. Sampling frequency of 6000 Hz. Training set (15 utterances per command); testing set (30 utterances per command). The total amount of Normalization of the spectra to unitary amplitude and 2000 points. The allowed bandwidth was in the range from 30 to 180 Hz.

4.1.2. Resulting sub-bands and performance. The genetic algorithm was executed 20 times, and the maximum Fisher’s ratio obtained was 62. The resulting best chromosomes (encoding the best sub-bands (Hz)) were:

BF = [254 180 526 132 744 118 1196 141 1483 115 2082 86 2295 171 2828 46].

From which the corresponding center and bandwidth are:

C = [254 526 744 1196 1483 2082 2295 2828],

BW = [180 132 118 141 115 86 171 46].

Fig. 2 shows the ideal transfer function of the respective filter bank. The recognition rate in the training set was 100% and in the tes set was 100% also. The amount of trials was 225.

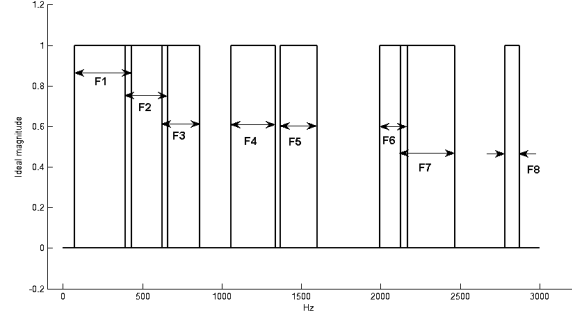


Fig. 2. Ideal transfer function of the filter bank. 8 sub-bands labeled as F1, F2, F3,..., F8.

4.1.3. Behavior of the Genetic Algorithm.

Consecutive executions of the GA produced variable Fisher’s ratio. It was observed here that the population size is critical, since a population of 20 produced Fisher’s ratios between 35 and 42, while a population of 120 individuals easily produced Fisher’s ratios above 58. It was noticed during experimentation that specific values for restrictions (subsection 3.3.3) also have a strong influence in the outcome.

4.2. Results in a English Vocabulary

4.2.1. Definitions and set-up. The English vocabulary was comprised of eight words to command a wheelchair, $V_e = \{\text{'faster'}, \text{'slower'}, \text{'left'}, \text{'right'}, \text{'stop'}, \text{'forward'}, \text{'reverse'}, \text{'brake'}\}$. 12 sub-bands were used. Sampling frequency of 8000 Hz. Training set: 20 utterances per command; testing set: 50 utterances per command. The total amount of Normalization of the spectra to unitary amplitude and 2000 points. The allowed bandwidth was in the range from 30 to 180 Hz.

4.2.2. Resulting sub-bands and performance. The genetic algorithm was executed 50 times, varying population size, probability of mutation, restrictions, number of sub-bands, and other parameters. The initial number of sub-bands was 8, then it was scaled to 9 and 12; in this scenario, the Fisher’s ratio varied from 23 (8 sub-bands, population size = 10) to 48 (12 sub-bands, population size = 200). The resulting centers and bandwidths are:

C = [250 446 648 1283 1483 1776 2018 2506 2737 3197 3383 3833],

BW = [5 138 187 157 139 43 207 106 105 98 148 224].

The performance in the training set was 100%; the performance in the testing set was 93.75%.

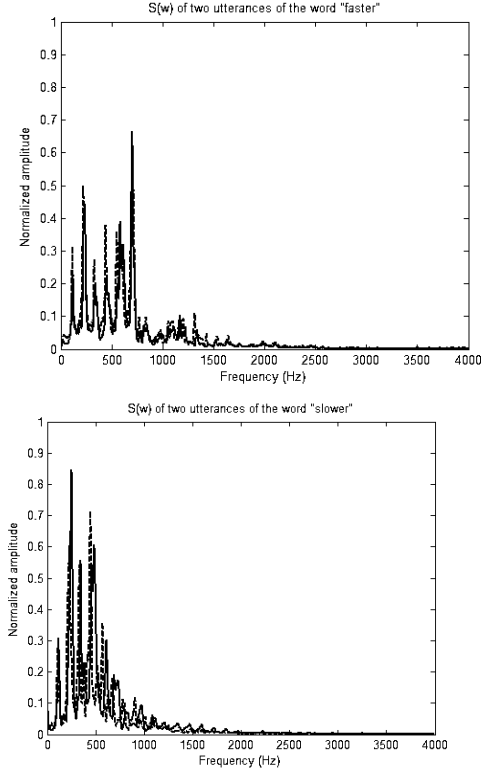


Fig. 3 The spectra of two words. Intra-class repeatability and inter-class distinction.

The number of trials was 560. The correct classification rate per word was [90 98 100 92 88 92 94 96] for {'faster', 'slower', 'left', 'right', 'stop', 'forward', 'reverse', 'brake'}, respectively.

Fig. 3 shows the normalized spectra of two utterances of the word 'faster' and the word 'slower'. In both cases notice the repeatability in the frequency domain, as well as the difference between both sets of spectra.

4.2.3. Real-time implementation in a DSP. The case of the Spanish vocabulary was portrayed to a digital signal processor TMS320LF2407, from Texas Instruments, in order to verify the performance using the filter bank in a nearly real-life application. No attempt was made to segment the incoming audio signal into voiced/no-voiced; instead, a push-button was used to start and finish capturing the voice. The DSP has a 10-bit analog to digital converter built-in, facilitating this way the interfacing task. The 8 sub-band filters (subsection 4.1.1) were implemented using an IIR topology, being of the type elliptic, 8th order. The filter coefficients (A , B), the covariance matrices (Σ_i) and the word models (μ_i) were transferred to the DSP program.

The analog-to-digital conversion was set-up to acquire one audio sample every T sec, ($T=1/6000$); each time a sample came into the device, the filters actuated and the respective output was squared to calculate the energy. The energy at each sampling instant was accumulated to obtain the energy over all the period of existence of the voice. Scaling issues had to be solved since the model was created in a real valued $[-1, 1]$ scale, while the DSP just "see" integer values. Once a whole command was processed, it was just a matter of a few milliseconds to apply the minimum distance classifier and provide the classification. The correct classification rate was in this case of the order of 93%, in 60 repetitions of the vocabulary (300 trials).

Other authors reported alternative methods such as HMM, neuronal networks, etc., to solve the same problem, showing efficiencies below 93%. For instance, de Luna Ortega reported 84% for HMM and 81% for a Kohonen Neural Network [9], and Lam *et al* reported 81.33%, using a combination of LPCC, VQ, and HMM, in applications similar to the one reported here.

4.3. Conclusions

In this paper was presented a method to implement a high performance, real-time, command voice recognition system, combining a genetic algorithm and the Fisher Linear Discriminant Analysis (FLDA). The features space used was the energy of the signal in certain sub-bands in the frequency domain; thanks to the Parseval's theorem, the same amounts of energy can be calculated in the time domain via a bank of digital filters, which enables a very fast way of applying the recognizer, since the process goes on at the same time as the occurrence of the command voice. Two experiments were shown, in Spanish and English command voice recognition, and in both cases high performance was attained. About the genetic algorithm, it consistently maximized the criterion of inter-class separability and intra-class compactness, under different conditions of population, probability of mutation, etc., and also varying the restriction set. Thus, the Fisher's ratio could be improved by a factor of 2 in some cases. It is remarkable and worth to mention that the genetic algorithm didn't gave the best result in its first execution, which means that the execution must be repeated to achieve really good results; increasing the population did not improve significantly the performance of the GA; instead, playing with the restriction set and some other parameters of the GA gave better results.

In the experiments reported, the performance, considering only the training sets, can be calculated as $(100 + 93.75 + 93) / 3 = 95.8\%$. The real-time implementation was shown with the use of the DSP.

5. References

- [1] S. Theodoridis, and K. Koutroumbas, *Pattern Recognition*, Academic Press, USA, 1998.
- [2] Madiseti, V. K. and Williams, D. B. *Digital Signal Processing Handbook*. CRC Press LLC, USA, 1999.
- [3] S. Kwong, Q.H. Heb, K.W. Kua, T.M. Chana, K.F. Mana, and K.S. Tang, "A genetic classification error method for speech recognition", *Signal Processing*, 83(5), 2002, pp. 737-748.
- [4] M.L. Lan, S.T. Pan, and C.C. Lai, "Using Genetic Algorithm to Improve the Performance of Speech Recognition Based on Artificial Neural Network", *First International Conference on Innovative Computing, Information and Control*, 2(30-01), 2006, pp. 527-530.
- [5] S. Kwong, C.W. Chau, and W.A. Halang, "Genetic algorithm for optimizing the nonlinear time alignment of automatic speech recognition systems", *Industrial Electronics*, IEEE Transactions, 43(5), 1996, pp. 559-566.
- [6] T. Takara, K. Higa, and I. Nagayama, "Isolated Word Recognition Using the HMM Structure Selected by the Genetic Algorithm", *IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP'97)*, 2, 1997, pp. 967.
- [7] D.E. Goldberg, *Genetic Algorithms in Search, Optimization and Machine Learning*, Kluwer Academic Publishers, Boston, MA., 1989.
- [8] S.W. Smith, *Digital Signal Processing. Demistifying Technology Series*, Newenes, USA, 2003.
- [9] de Luna Ortega, C.A. *Reconocimiento de Voz con Redes Neuronales y Modelos Ocultos de Markov*, Master Thesis, Instituto Tecnológico de Aguascalientes, México, 2005.
- [10] Lam, Y.M., Mak, M.W., and Leong, P. H. "Fixed-point implementations of speech recognition Systems". Proceedings of the ISPC 2003 Conference. 2003. Pp.

SADCoRH - a dynamic system to solve conflicts in the use of water through the generation of rules by genetic algorithms

Pedro Salves Arraes Neto¹
pedroarraes@gmail.com

Andréia Gentil Bonfante¹
andreiab@ufmt.br

Peter Zeilhofer²
pitalike@terra.com.br

Claudia Aparecida Martins¹
claudia@ufmt.br

Josiel Maimone de Figueiredo¹
josiel@ufmt.br

¹Instituto de Computação

²Departamento de Geografia

Universidade Federal de Mato Grosso (UFMT)

Av. Fernando Corrêa s/nº, Coxipó
Cuiabá- MT, 78060-900, Brasil

Abstract

The sustainability of the water uses in the future depends on the planning and the rational management of water resources. As many governments around the world, Brazil has endorsed in 1997 the federal law 9.433 on water resources, which previews the development of management tools to support decision making on how to use and regulate the rights of rational water use. This work presents a computational tool, based on genetics algorithms, able to optimize water use allocation. The system is validated in a case study in the Cuiabá river basin, in Mato Grosso State, Brazil.

1. Introduction

Decision-making in water resources (WR) allocation must be based on information on the actual availability of such resources, in terms of environmental factors (stream-runoff, water quality) and socio-economic determinants (existing and future users, etc.). The grants must guarantee qualitative and quantitative aspects of the basin. Meanwhile, ensure water availability, taking into account economic and social factors, is a measure that can create conflicts between users. Conflict solving can be ensured by law enforcement, since there are tools, able to subsidy the evaluation in the decision making process for water use granting ([5],[6],[4]). Legislation makes explicit that the use of water is a universal right. Meanwhile, the legislation does not grant water in the case of the low-flow stream discharge

is expected to become below 80% of the so-called "Ecological Discharge", estimated generally to be 80% of river low-flow statistics. In this context, the SADCoRH system was developed to help in the optimization of water resource allocation by means of genetic algorithms, generating rules by the comparison of simulated flows before and after the inclusion of a new water user. The SADCoRH works as a module coupled with the "Outorga" [2] water availability model, that allows the simulation of scenarios (addition of a new user) indicating if grants can be conceded or not under the current environmental conditions. If in a simulation scenario a new venture can not be granted, the SADCoRH is invoked, proposing an optimization of cotes conceded to WR users in the basin.

This work presents a system based on genetic algorithms to subsidy the management of water resources. The article is structured as follows. Section 2 summarizes the Outorga model. Section 3 presents the SADCoRH model, that is a genetic algorithms based system for conflict solving between water users. Section 4 presents the results and discussions of a case study. Finally, Section 5 concludes this article.

2. Outorga Model

Decision-making in WR allocation should be based on information on the actual availability of water resources, considering environmental (river-discharge, water quality) and socio-economic (existing and future users, etc.) factors. Da Hora [2] proposed a decision system called "Outorga" for the granting of water resources, where water availability is estimated according to qualitative and quantitative

aspects. The system is not able however, to subsidy the re-distribution of water grants between users, guaranteeing the universality of water use as endorsed by legislation.

For the concession of operating licenses to water users, estimates of water availability are generally based on probabilistic rates of low-flow rates, such as Q95 (95% percentile from the flow duration curve) or $Q_{7,10}$ (minimum 7-day average discharge with a return period of 10 years). Under the quantitative aspects, the concession can be granted as long as Equation 1 is true.

$$\sum_{t=0}^n u(i).v(i, t) < vMax(i, t), \forall \quad (1)$$

Where:

- $u(i)$ is the coefficient of use of water resource in place i . For example, if the rate of use is 0.1 means that 90% of the captured water returns to river;
- $v(i, t)$ is the captured flow by a user in a place by the time interval Δt in m^3/s ;
- $vMax(i, t)$ is the maximum grants in i , with the reliability desired by $Q_{95\%}$ or $Q_{7,10}$ in m^3/s ;
- n is the maximum grants.

At river reaches with no withdrawal or effluent, concentration C of a water quality parameter is estimated as a function of its decay rate k and the time t needed by a water volume to pass one reach.

$$C = C_i e^{-kt} \quad (2)$$

Where:

- C is the concentration of effluent in the n th stretch of the river
- C_i is the concentration of effluent in the begining
- k is the rate of deterioration of effluent
- t is the deterioration time needed for a volume of water passing in the stretch of river

Runoff and concentration of a water quality parameter in reaches which are receiving an effluent are calculated by Equation 3.

The flow and concentration in the excerpts that are receiving an effluent are calculated in Equation 3.

$$Q_F C_F = Q_i C_i + Q_E C_E \quad (3)$$

Where:

- Q_F is the total ow in the river after launch, in m^3/s

- C_F is the concentration of effluent after launch
- Q_i is the ow early in the stretch of river
- C_i is the concentration of measure of quality early in the stretch of river
- Q_E is the ow of effluent released by the user in the stretch of river
- C_E is the concentration of effluent released by the user

For modeling, each river reach of a basin has to have associated attributes of flow, flow velocity and water quality parameters at its beginning and end (the parameters at the end of the reach are the parameters of the start of the next downstream reach and so on). Water users are allocated on a specific river reach and have attributed the flow of its withdrawal and effluent, as well as the concentrations of water quality parameters in its effluent. The simulation of a scenario begins with the insertion of a new user of water. The model run will than be executed for all reaches downstream the new user, where water availability will be affected.

Briefly, the model estimates the influence of already allocated uses and checks the stream flow grantable under quantitative aspects with Equation 1. Equation 2 is applied in reaches where no users are allocated and Equation 3 is used in reaches with water users. Decision on water allocation is done by the comparison between simulated data and the limits according to environmental criteria, verified by the result returned from Equation 1.

The model proposed by da Hora is limited to only quantitative and qualitative aspects of available WR and does not take into account socio-economic factors, neither simulates proposals for the conflicts solving, resulting only in the granting or negation of a requested water use. Under this aspect the proposed SADCoRH model is a complementation to Da Hora's model to subsidy decision support in WR allocation of new water users.

3. A solution through genetic algorithms

Genetic algorithms (GAs) are known as evolutionary algorithms, which are being studied since the beginning of the decade of 50. They are computer systems for problem solving inspired by the theory of natural evolution ([1],[3]). They operate on a population of individuals, based on the fact that those with favorable genetic characteristics have greater chances of survival and produce increasing offspring, while those with poorer characteristics tend to disappear.

3.1. SADCoRH model

Proposed SADCoRH is a GA based system, projected to solve the problem of conflicts between water users as presented in section 1. The generation of rules by the use of GAs allows the optimization of the flow withdrawals and the introduction of polluted effluents. The rules are generated through the comparison between the initial water availability estimates and the final optimized availability.

The first step in the implementation of a GA is the generation of a random population. Then it is verified if that population is the solution to the problem. If so, the GA has reached the end, as shown in the Algorithm 1. Otherwise, the steps of crossover, mutation, fitness and selection of individuals are executed. Finally, simulation is checked again if now a viable solution has been generated. The algorithm works until a solution is found, the number of cycles is unlimited.

Data: chromosomes

Result: a solution

```
create a new population();
while not found the solution(population) do
    descendants ← crossover(population);
    descendants ← mutation(population);
    descendants ← fitness(population);
    new populaion ← selection(descendants);
end
```

Algorithm 1: GA function

This scheme can be broken. If there is limited water availability for the negotiation between the existent and the new users, the GA will not finish. To overcome this possibility a pre-processing routine is done with the users who participate in the negotiations. Water availability for negotiation is described by the equation 4.

$$\frac{\sum_{i=1}^n flow_i * negotiationRate}{100} \quad (4)$$

Where:

- $flow_i$ is the amount of water used by the venture
- $negotiationRate$ is the maximum rate (1% to 100%) that can be withdrawn from the venture

If there is water availability which can be negotiated between users, the next step is to generate an initial population. In the following the algorithms used by SADCoRH are exposed.

3.1.1 Function to create a random population

This function uses data from the allocated ventures along the studied river basin. The created individuals have randomly attributed withdrawals (effluents are transformed in volumes, necessary for dilution of pollutants). Only this attribute is part of the chromosome, the other data are maintained in order to preserve the features of the venture. As the optimization procedure is only based on the withdrawal flow, only this attribute is created randomly in the shape of a chromosome. The Algorithm 2 then generates random chromosomes. The decimal part was separated from the whole part of the flow volume due to the characteristics of the problem. Therefore, an individual with two chromosomes is created, a procedure not described in the literature. As the ventures have a withdrawals below the $1 m^3/s$, the entire part needs not to be generated and is not part of the process of the GA. The GA's next step is the validation if the population is the solution to the problem.

Data: population from database

Result: a population

```
Array newPopulation ;
AleatoryBit seed ;
for i=0 to size (population) do
    char[10] flowDECCap ;
    for j=0 to 9 do flowDECCap[j] = seed.nextBit();
    individDataBase = population[i] ;
    individ.add(flowDECCap and individDataBase
    features);
    newPopulation.add(individual);
end
```

Algorithm 2: Create-population function

3.1.2 The algorithm found a solution?

To find a solution for the domain problem (conflicts in the use of WR), the lowest possible flow rates should be assigned to the ventures installed in the basin, thus increasing the water availability and allowing the installation of a new venture. The evaluation of the solution was divided into three stages:

1. verify if new simulated flow rates are below the maximum limit of withdrawals
2. verify if the standard deviation between the flow rates of the involved users are found below the limit set by a technician and
3. determine if the amount of water withdrawal of the existent businesses is sufficient for the assignment of a WR use licence for the new venture.

Data: Array of individuals, allowedWithdrawal

Result: boolean value

```
for i=0 to size(individuals) do
  if individuals[i].flowCapt >= allowedWithdrawal
    then
      | return false
    end
  end
end
Return true
```

Algorithm 3: Found solution function

The second criterion is based on the standard deviation of the water volume percentage drawn from each individual (user). If the standard deviation is greater than the limit defined by the technician, the function returns a false boolean value, returning to the initial phase of the GA, resulting in the generation of new individuals.

The third criterion, described by the Equation 5 and the Algorithm 4, is the sum of the difference of initial flow with the flow optimized for all ventures. If this sum is greater than or equal to the flow of the new user, the criterion has been obeyed.

$$totalFlow = \sum_{i=1}^n withdrawalQuant_i \quad (5)$$

Data: RequiredFlow

Result: boolean value

```
if totalFlow > requiredFlow then
  | return true
end
Return false
```

Algorithm 4: Check availability function

If none of the stopping criteria has been met, the GA runs through the steps of crossover, mutation, fitness and selection, resulting in a new generation. In the following each of these steps is detailed.

3.1.3 Crossover function

The crossover scheme adopted in SADCoRH was to generate two children from two individuals. The crossing divides the chromosome into two chains. The first chain of A individual is concatenated with the second chain of the B individual. And, the second chain of A is concatenated with the first chain of B. The results are two new individuals with characteristics of both parents.

3.1.4 Mutation function

SADCoRH does the mutation at random. The purpose of this function is to help the convergence of the population to a viable solution. In Algorithm 5 the mutation scheme adopted in SADCoRH is presented.

Data: Array of Descendants

Result: Descendants

```
if seed.nextAleatoryValue == True then
  DescendantsSelected = seed.nextNumber;
  ChromosomePosition = seed.nextNumber;
  MutationBit = seed.nextBit;
  Descendants.alterBit(DescendantsSelected,
    ChromosomePosition, MutationBit)
end
```

Algorithm 5: Mutation function

The mutation can occur or not. At first a boolean value is selected that determines whether the mutation occurs or not. The next step is to determine which individuals will change. And finally, the bit can be reversed in the chromosome of the individual. Equation 6 determines the probability of occurrence (in percentage) of a mutation on a specific bit.

$$P(\text{mutationBIT}) = (80 * (\text{DescendantsNumbers}))^{-1} * 100 \quad (6)$$

For example, if 10 children are generated from a crossing, the likelihood of a mutation to occur in a bit is 0.125%.

In the next topic, the fitness function is exposed.

3.1.5 Fitness function

In SADCoRH, the fitness was determined in accordance with the assessment of an expert in the field of water resources. The Algorithm 6 defines the function of fitness with the likelihood of individuals to proceed to the next generation. The fitness function simply compares the individual with the original one generated from the database. When the flow is reduced, a note is attributed to the new individual. The note varies between 1 and 100 and determines the likelihood of an individual to be in the next generation.

3.1.6 Selection function

The selection function in SADCoRH uses a roulette([1]) that chooses a number between 1 and 100. The individual is select or not by its fitness. The higher the fitness, the greater the chances.

For example, consider a roulette with two individuals A and B, the first with a probability of 40% to be selected and

Data: Array of Array of Descendants

Result: Descendants evaluated

```

for i=1 to size(Descendants) do
  Descendants desc = Descendants[i] ;
  for j=0 to j < 2) do
    if reduction < 0.0 and reduction >= -5.0 then
      desc[j].setFitness(70);
    if reduction < -5.0 and reduction >= -25.0
    then desc[j].setFitness(10);
    if reduction < -25.0 and reduction >= -50.0
    then desc[j].setFitness(5);
    if reduction < -50.0 and reduction >= -85.0
    then desc[j].setFitness(3);
    if reduction < -85.0 and reduction >= -95.0
    then desc[j].setFitness(2);
    if reduction < -95.0 and reduction >= -100.0
    then desc[j].setFitness(1);
    if reduction > 0.0 then desc[j].setFitness(1);
  end
end
end

```

Algorithm 6: Fitness function

the second with a probability of 60%. If a number between 1 and 60 is chosen B will be selected; if the number is between 61 and 100, A will be selected. Algorithm 7 details this concept.

Data: Array of Array of Descendants

Result: New Population

```

while there are Descendants do
  randomly select a number from 1 to 100 ;
  if number <= fitness(individual A) then
    choose(individual A);
  else choose(individual B)
end

```

Algorithm 7: Selection function

Passing these six steps, the GA can reach a solution or it repeats the processing steps until the data converge toward a desired solution. In section 4 some simulations results are shown.

4. Results and discussions

The optimization procedure of the GAs to find acceptable river flows is as follows: from the moment that a solution for a specific conflict is found, there is no further need to apply the GA, this because the system is capable to store solutions, to score them positively or negatively according

to the evaluation of a specialist, and to make a reuse of the generated rules in similar cases. This technique brings a vast economy of processing, since GAs may have a high cost in more complex simulations. The sequence below shows some possible simulations.

There are three groups of simulations, according to the complexity level of the solution. The first level uses a maximum water volume for a withdrawal of up to 20% of the original user; the second up to 15% and third 10% at maximum. The maximum standard deviation for reducing a withdrawal was limited to 1.5% for all three simulations and the flow required for the new venture to be installed ranged between $0.1 \text{ m}^3/\text{s}$ and $0.6 \text{ m}^3/\text{s}$. The selected river reach for the simulation was always the same, to facilitate the comparison between the results. In Figure 1 it can be observed that the greater the required flow required by a new venture the larger will be the number of WR users involved in the negotiations.

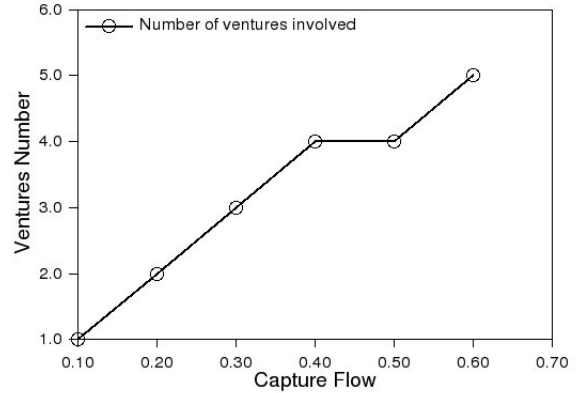


Figure 1. Number of ventures involved X Captured flow

Figure 2 was developed based on the same parameters used for Figure 1. In Figure 2 it is possible to see a trend line showing an exponentially increase of number of generations, if the complexity of the problem increases (increase of requested withdrawal). However, this behavior is not a rule. In a new run, GA can find a solution for the same problem using lesser or more generations. This fact occurs due to the nature of GA algorithms which main feature is the randomness of their results within a parameters set. It can be seen that 4740 generations were necessary to ensure a withdrawal of $0.5 \text{ m}^3/\text{s}$, while for a simpler problem ($0.4 \text{ m}^3/\text{s}$), the number of needed generations was higher. Comparing figures 1 and 2 it is possible to observe a typical behavior of the SADCoRH for a same simulation: (1) the increase of ventures involved in the negotiation process if the required flow volume increases and (2) an increase in

computational demand if more ventures are involved.

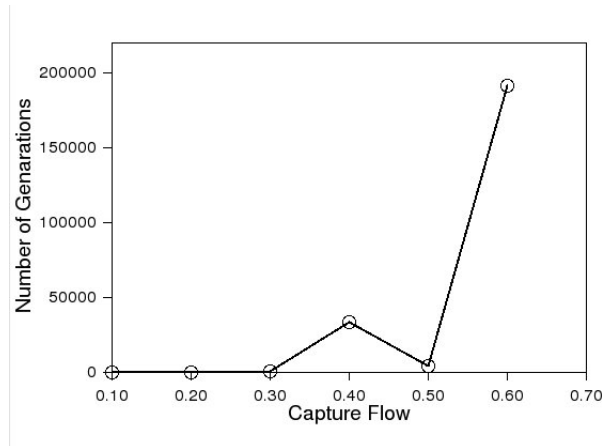


Figure 2. Number of generations X Captured flow

Simulation: (20% of maximum reduction, standard deviation of 1.5%, required flow $0.2 \text{ m}^3/\text{s}$):

```
***** GENERATION: 434 *****
DNAAE cod: 6610000 - T: 229
  Old Flow: 00.9600
  New Flow: 00.7960
  Reduction: -17.0833%
DNAAE cod: 6610000 - T: 96
  Old Flow: 00.9000
  New Flow: 00.7460
  Reduction: -17.1111%

-----
Average      : -17.0972%
Variance     : 00.0004%
Standard Deviation: 00.0196%
-----

Founded Flow : 00.3180
Required Flow : 00.2000
Maximum Standard Deviation: 01.5000
-----
```

5. Conclusions

This paper presents only a preliminary development stage of what is planned for the system as a whole, but it could be evidenced, that the use of intelligent agents can be helpful in decision making in the application of decision support systems for WR allocation. Machine learning, by computer or evolutionary learning through stochastic databases, has a key role in intelligent systems able to interact with the human and can propose solutions to problems to a defined area of human knowledge. The use GAs for the generation of dynamic rules for resolving conflicts

could be proved to be efficient, and exceeded initial expectations, since there were no references on the use of GA on optimizing flow rates of WR in river basins. To limit high processing costs typical for GAs, the SADCoRH permits the storage of rules and its application in similar situations without the necessity to develop new GAs. SADCoRH will hopefully support future research in the field of the decision support systems for WR allocation and other environmental problems in general. Future development of the system should include improvements of the learning schema for compensation and punishment of the rules stored in the database, as well as the inclusion of projections about the future WR availability (ex. climate change) and the introduction of tools in the system to generate future scenarios.

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References

- [1] T. Bäck and U. Hammel. Evolutionary computation: Comments on the history and current state. *IEEE Transactions on Evolutionary Computation*, 1997.
- [2] A. F. Da Hora. *Metodologia para outorga do uso de recursos hídricos. Ênfase: Usinas hidrelétricas*. PhD thesis, Engenharia de Produção - Universidade Federal do Rio de Janeiro - COPPE, 2001.
- [3] T. Mitchell. *Machine Learning*. McGraw-Hill Book Co., United States, 1997.
- [4] R. B. Rodrigues. *Sistema de suporte a decisão para a gestão quali-quantitativa dos processos de outorga e cobrança pelo uso da água*. PhD thesis, Engenharia Civil - Universidade de São Paulo - USP, São Paulo, SP, 2005.
- [5] I. M. Santos and P. Zeilhofer. Modelagem hidrológica integrada em sistemas de informação geográfica. *Geodesia online*, 4:1–16, 2005.
- [6] P. Zeilhofer and R. M. Lima. Spatial patterns of water quality in the cuiabá river basin, central brazil. *Environmental Monitoring and Assessment*, 123:54–61, 2006.

On the estimation of the Pareto optimal set using an evolutionary parameter adjustment of the normal distribution function

Luis M. Torres-Treviño
Corporación Mexicana de Investigación en Materiales S. A.
Ciencia y Tecnología 790
Saltillo, Coahuila, México. CP 25290
ltorres@comimsa.com.mx

Abstract

In multi-objective evolutionary optimization it is desirable to obtain a set of solutions usually called the Pareto optimal set. In this paper, some preliminary results are presented to approximate a Pareto optimal set considering an evolutionary adjustment of the parameters of a normal distribution function. The univariable and multivariable normal distribution functions are considered. Four test functions are used considering two forms of evaluation. Results indicate a novel way to represent the Pareto front and suggest different ideas for this area of optimization.

1 Introduction

There are several problems of optimization where there is more than one objective and usually these objectives are in conflict with each other. These classes of problems are called multiobjective optimization. The use of evolutionary algorithms for these optimization tasks have been increased in recent years [3, 2]. The principal reason is the easy integration and implementation of evolutionary algorithms to solve a variety of problems. The Multi-objective optimization [5] can be defined as follows:

Defining Ω as the search space where there are n objective functions, optimize $F(X) = \{f_1(X), f_2(X), f_3(X), \dots, f_n(X)\}$ where $X \in \Omega$. The Pareto dominance can be defined as follows, considering a minimization problem: X_1, X_2 are two vectors (solutions) from Ω where X_1 dominates X_2 if $F_i(X_1) \geq F_i(X_2)$ $i = 1, 2, \dots, n$

If S is a subset of Ω then all solutions not dominated by S are called non-dominated with respect to S . A local Pareto solutions can be defined if this set of solutions are not dominated by a defined subset S . Solutions that are

non-dominated with respect to the entire search space Ω are called Pareto Optimal solutions. When the search space is a subset of R , then Pareto Optimal set may be represented as:

1. A set of points
2. A set of disjoint intervals
3. A recombination of the last two

The Pareto Optimal set is usually a union of continuous sets. When discrete solutions are used, there is some information loss. Considering this definition, a multiobjective optimization may be reduced to the search of a distribution function that represents a Pareto Optimal Set. The solution may be a distribution function, and as a preliminary result a normal distribution function. The evolutionary estimation of a normal distribution function parameters (Evonorm) [8, 9] is an easy implementation of an estimation of distribution algorithms (EDAs) [6] where the crossover and mutation operators are substituted by an estimation of a distribution function to represent a population of fittest solutions. The use of EDAs in multiobjective optimization can be situated since the origins of this paradigm. The first reference is given in the use of IDEA for multiobjective optimization called MIDEA [1]. Authors use a mixture of distributions for modeling complicated dependencies between variables and to preserve the diversity. An interesting study was made by Kumara Sastry et al. [7] This paper presents a study about the scalability of EDAs in multiobjective optimization paying attention to the number of function evaluations required, the number of Pareto-optimal solutions and the population size. The principal conclusion of the study is the exponential increasing of the Pareto optimal solutions with the problem size due to failure in niching mechanism. This causes a problem to maintain a good Pareto Optimal front. This can be corrected when the growth rate of the Pareto optimal solutions is controlled. Finally, Chang Wook Ahn made

some numeric comparisons between EDAs for multiobjective optimization [11]. In his paper is presented a multiobjective real -coded Bayesian optimization algorithm (MrBOA). The author shows a comparison of his approach versus NSGAII and MOIDEA achieving a better performance.

2 Evonorm and Correlated Evonorm to solve multiobjective optimization

Firsts versions of Evonorm works with the assumption that there are independent variables; however, this assumption is not really true, in this paper a Correlated Evonorm (COEvonorm) is presented. This algorithm uses correlated variables represented in a covariance matrix. Evonorm includes a heuristic to orientate the search considering the best solution found or the calculated mean from selected solutions. Any distribution function can be used; however, Evonorm emphasizes the use of normal distribution functions considering its simplicity.

The principal idea of Evonorm is the generation of a model that represents a condensed version of the population. It is expected a population of solutions with high fitness so it is possible to get a set of solutions with the same algorithm. The principal difference of Evonorm with an EDA is the use of a heuristic to generate the new population where 50% of the time the best individual found is used as a mean of the random number generator and the other 50% of the time is used the calculated mean of the selected population. The general algorithm of Evonorm is illustrated in Table 1. Usually the number of selected individuals to generate a new population (PS) is lower than the number of individuals of the original population (P is a matrix where the number of rows represents the number of individuals and the number of columns represents the number of parameters or decision variables). A fitness (FE is a vector where the number of individuals determines its size) is determined considering the normalized value of the evaluation function. A mean(MN) and a standard deviation (SD) is calculated per parameters considering the selected population (These structures are vectors that stores the calculated mean and standard deviation per decision variable, the number of decision variables determines the size of these vectors). A new population is generated using a normal random generator considering the mean and standard deviation calculated per parameter. Considering a probability of 50%, the calculated mean (MN) is used. The other 50% is used the best solution found.

Evonorm has been applied to solve global and constrain satisfaction optimization and it is possible to extend the application of Evonorm in Multiobjective optimization (MO). The solution will be a probabilistic model that represents a set of solutions (very similar among them) with high fitness. This aspect is very important in MO and usually is re-

Table 1. The general algorithm of Evonorm

1)	Create a population P using a uniform random generator
2)	Evaluate P and generate a fitness FE
3)	Select the best individuals of P using FE to generate PS
4)	Calculate the parameters of MN and SD using PS (Randomly uses the calculated mean and the best solution found every 50 % of the time)
5)	Generate a new population P using MN and SD
6)	Repeat steps (2-5) until end condition is satisfied

Table 2. Evonorm algorithm considering the evaluation function.

1)	Create basic population P
2)	Evaluation(P) $\rightarrow$ (MFE,Imax)
3)	Selection(P,MFE) $\rightarrow$ PS
4)	Calculation(PS,Imax) $\rightarrow$ (MN,SD)
5)	Generation(MN,SD) $\rightarrow$ P
6)	if end condition not satisfied go to step (2)

lated with the Pareto Optimal set; however, the low diversity in the final population generates a tiny standard deviation. The increment of the standard deviation could generate an approximation of the Pareto Optimal set.

Two forms to establish the fitness evaluation has been considered. The first form uses a normalized version of the numeric functions. The evaluation function gives a high fitness when there is a solution that generates a minimum value (as possible) in all the functions. The second form is considering an evaluation of the dominance of every solution considering all the population except itself. The evaluation of individuals considering the non-dominate grade has been used in NSGA II and SPEA2 algorithms [4, 12]. The evaluation function gives a high value of fitness when the solution dominates a high number of solutions of the population. The use of Evonorm considering the first form of evaluation mentioned above is shown in Table 2). A second version of Evonorm considering as the evaluation function the number of individuals dominated is shown in Table 3. COEvonorm imply the calculation of a covariance matrix. The same evaluations forms used in Evonorm are considered in COEvonorm too (Table 4 and Table 5)The covariance generator procedure consider the generation of a triangular matrix from the covariance one to generate a random number from a multivariable normal distribution function.

Where Imax is the best individual found (A vector where the number of its elements in the number of decision vari-

Table 3. Evonorm algorithm considering non-dominate evaluation

1)	Create basic population P
2)	Evaluation(P) → (MFE,Imax)
3)	Non dominate evaluation(P,MFE) → FED
4)	Non dominante selection(P, FED) → PS
5)	Calculation(PS,Imax) → (MN,SD)
6)	Generation(MN,SD) → P
7)	If end condition not satisfied go to step (2)

Table 4. COEvonorm algorithm considering the evaluation function

1)	Create basic population P
2)	Evaluation(P) → (MFE,Imax)
3)	Selection(P,MFE) → PS
4)	Covariance calculation(PS) → MCOV
5)	Eigen calculation(MCOV) → (V, D)
6)	Covariace generation(V, D, Imax) → (P)
7)	If end condition not satisfied go to step (2)

ables). MFE is a matrix to store the evaluation per function where every row represents the solutions and its fitness is represented in every column of the matrix. PS is the population with selected individuals. MN and SD store the mean and standard deviation per parameter respectively. FED is the non-dominate evaluation per individual and this structure has the same size than FE. MCOV is the covariance matrix (a square matrix where the number of decision variables determines its size), V is the Eigen vector and D contains the Eigen values in the principal diagonal.

3 Experimentation

A set of numerical experiments was made to visualize the performance of Evonorm and COEvonorm considering

Table 5. COEvonorm algorithm considering nondominate evaluation

1)	Create basic population P
2)	Evaluation(P) → (MFE,Imax)
3)	Non dominate evaluation(P,MFE) → FED
4)	Non dominante selection(P, FED) → PS
5)	Covariance calculation(PS) → MCOV
6)	Eigen calculation(MCOV) → (V, D)
7)	Covariace generation(V, D, Imax) → (P)
8)	If end condition not satisfied go to step (2)

Table 6. Test functions

Schaffer F_1	$F_1(x) = x^2$ $F_2(x) = (x - 2)^2$
Schaffer F_2	$F_1(x) = \begin{cases} x^2 & \text{if } x \leq 1 \\ x^2 & \text{if } 1 < x \leq 3 \\ x^2 & \text{if } 3 < x \leq 4 \\ x^2 & \text{if } x > 4 \end{cases}$ $F_2(x) = (x - 5)^2$
Valenzuela-Rendon F_1	$F_1(x) = 1/(x_1^2 + x_2^2 + 1)$ $F_2(x) = x_1^2 + 3 * x_2^2 + 1$
Valenzuela-Rendon F_2	$F_1(x_1, x_2) = x_1 + x_2 + 1$ $F_2(x_1, x_2) = x_1^2 + 2 * x_2 - 1$

the two forms of the evaluation function mentioned above. The objective is to find a normal distribution function that approximate a Pareto Optimal set assuming dependence and independence between variables. Evonorm and COEvonorm operate as an evolutionary algorithm so the diversity usually diminishes as the algorithms runs. The standard deviation in Evonorm and the Covariance matrix in COEvonorm tend to be zero; however, it is possible to get a set of solutions when the standard deviation or the covariance matrix increases its values. Four simple test functions were used in Evonorm and COEvonorm [2, 10] (Table 6) considering the preliminarily of the study.

A population of 100 individuals is used where 25 of them are selected to produce the models in order to generate the new population per generation through 100 generations. Every algorithm runs 100 times so the results illustrate the average performance. The plots were made generating 1000 solutions from the distribution function evolved. The extended versions were made setting a standard deviation to one in Evonorm and to the identity matrix in COEvonorm. The results for Schaffer functions are illustrated from Figure 1 to Figure 4. The results for Valenzuela-Rendon function are illustrated from Figure 5 to Figure 8.

4 Discussion, conclusion and future work

The Pareto optimal set of previous approaches are very similar. There is not a difference to use a specific form of evaluation. The objective of this preliminary study is to find a normal distribution function that represents a Pareto optimal set. This assumption may not be true because the Pareto Optimal set could be represented with a different distribution function. Evonorm and COEvonorm generate a low diversity so the use of niches could improve the solution set; however, the solution generated can be useful when the standard deviation (o covariance matrix) is extended to generate more solutions near of the Pareto Optimal set.

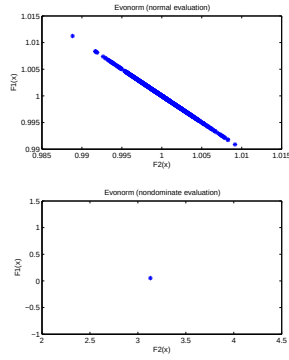


Figure 1. Evonorm performance considering Schaffer function 1 and two forms of the evaluation function. The standard deviation is set to 1 in the extended version.

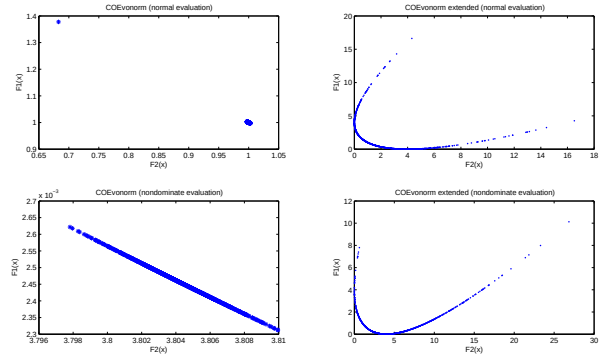


Figure 2. COEvonorm performance considering Schaffer function 1 and two forms of the evaluation function. The covariance matrix is set to 1 in the extended version.

Future work includes the use of clustering in order to generate a multiple representation of the Pareto optimal set. This procedure will work in some problems where there are multimodal functions. Finally, this is a preliminary result of the proposal, so it is necessary to use other test functions, the use of niches (every niche can be represented by a distribution function), the use of empiric distribution function, the use of a numeric performance measures and the comparison of the present approach with common used approaches like NSGA II and SPEA 2.

References

- [1] P. A. N. Bosman and D. Thierens. Multiobjective optimization with diversity preserving mixture-based iterated density estimation evolutionary algorithms. *International journal of approximate reasoning*, 31:259–289, 2002.
- [2] C. A. C. Coello, G. B. Lamont, and D. A. V. Veldhuizen. *Evolutionary Algorithms for Solving Multi-Objective Problems. Second edition*. Springer, New York, 2007.
- [3] C. A. Coello-Coello. *20 Years of Evolutionary Multi-Objective Optimization: What Has Been Done and What Remains to be Done*. IEEE Computational Intelligence Society, 2006. pp. 73-88.
- [4] K. Deb, A. Pratap, S. Agarwal, and T. Meyarivan. A fast and elitism multiobjective genetic algorithm: Nsga ii. *IEEE Transactions on evolutionary computation*, 6(2):182–197, 2002.
- [5] D. Dumitrescu, C. Grosan, and M. Oltean. *Evolving Continuous Pareto Regions*, pages 167–199. Abraham A., Jain L. and Goldberg R. (Eds.). Springer-Verlag, 2005.
- [6] P. Larranaga and J. A. Lozano. *Estimation of Distribution Algorithms: A New Tool for Evolutionary Computation (Genetic Algorithms and Evolutionary Computation)*. Kluwer Academic Publishers, 2002.

- [7] K. Sastry, M. Pelikan, and D. E. Goldberg. Descomposable problems, niching and scalability of multiobjective estimation of distribution algorithms. Technical Report IlliGAL Report no. 2005004, Illinois Genetic Algorithms Laboratory, 2005.
- [8] L. M. Torres-Treviño. Evonorm, a new evolutionary algorithm to continuous optimization. In *Workshop on Optimization by Building and Using Probabilistic Models (OBUPM-2006) Genetic and Evolutionary Computation Conference (GECCO-2006) CD Proceeding Tutorials and Workshops*, Seattle, WA, 2006.
- [9] L. M. Torres-Treviño. Evonorm: Easy and effective implementation of estimation of distribution algorithms. *Research in Computing science*, 23:75–83, 2006.
- [10] M. Valenzuela-Rend'on and E. Uresti-Charre. A non generational genetic algorithm for multiobjective optimization. In *Proceedings of the Seventh International Conference on Genetic Algorithms*, pages 658–665, Seattle, WA, 1997.
- [11] C. Wook-Ahn. *Advances in Evolutionary Algorithms: Theory, Design and Practice*. Springer-Verlag, 2006.
- [12] E. Zitzler, M. Laumanns, and L. Thiele. SPEA2: Improving the strength pareto evolutionary algorithm. Technical Report Technical Report 103, Computer Engineering and Networks Laboratory (TIK), ETH Zurich, Zurich, Switzerland, 2001.

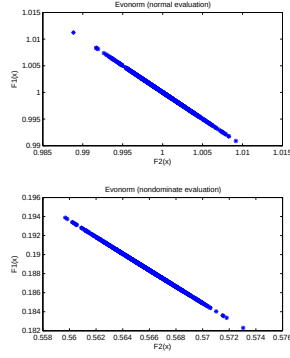


Figure 3. Evonorm performance considering Schaffer function 2 and two forms of the evaluation function. The standard deviation is set to 1 in the extended version.

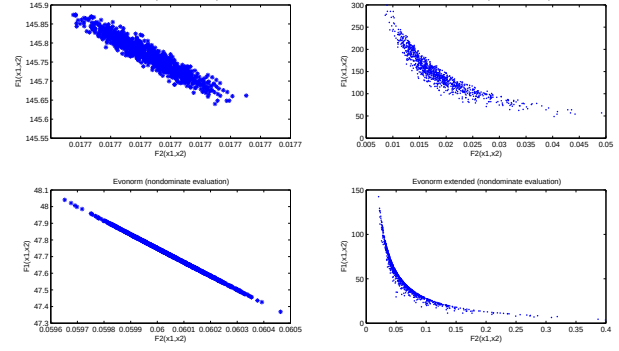


Figure 5. Evonorm performance considering Valenzuela-Rendon function 1 and two forms of the evaluation function. The standard deviation is set to 1 in the extended version.

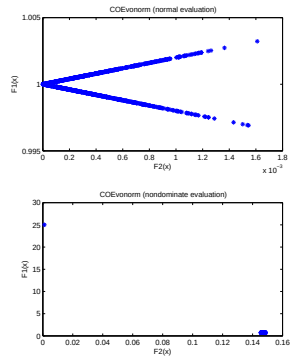


Figure 4. COEvonorm performance considering Schaffer function 2 and two forms of the evaluation function. The covariance matrix is set to 1 in the extended version.

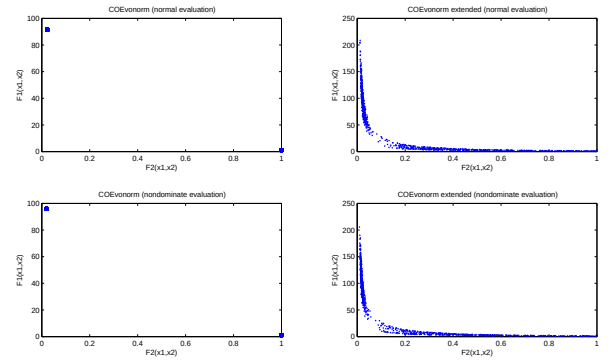


Figure 6. COEvonorm performance considering Valenzuela-Rendon function 1 and two forms of the evaluation function. The covariance matrix is set to 1 in the extended version.

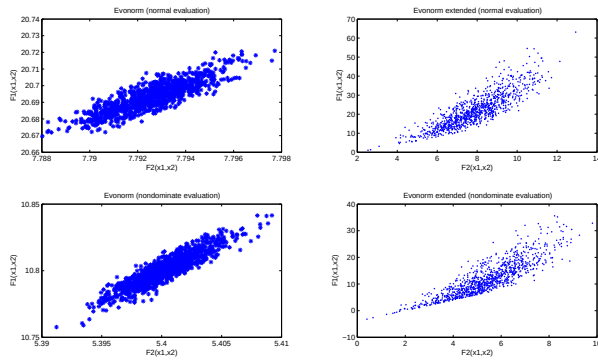


Figure 7. Evonorm performance considering Valenzuela-Rendon function 2 and two forms of the evaluation function. The standard deviation is set to 1 in the extended version.

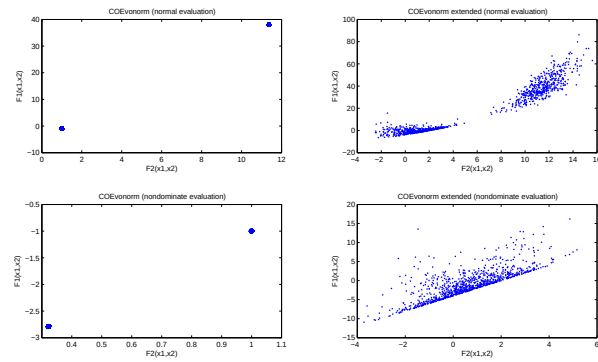


Figure 8. COEvonorm performance considering Valenzuela-Rendon function 2 and two forms of the evaluation function. The covariance matrix is set to 1 in the extended version.

Simulated Annealing for SAT Problems using Dynamic Markov Chains with Linear Regression Equilibrium

Felix Martinez-Rios¹ and Juan Frausto-Solis²

¹Universidad Panamericana, Campus Ciudad de México, Augusto Rodín 498,
Col. Insurgentes Mixcoac, 03920, México DF, México
fmartin@up.edu.mx

²Instituto Tecnológico y de Estudios Superiores de Monterrey, Campus Cuernavaca, Autopista
del Sol km 104, Colonia Real del Puente, 62790,
Xochitepec, Morelos, Mexico
juan.frausto@itesm.mx

Abstract

Since the appearance of Simulated Annealing (SA) algorithm it has shown to be an efficient method to solve combinatorial optimization problems. This algorithm is based on two cycles: the external or temperature cycle and the internal or Metropolis Cycle. In this paper a new SA method named LRSA is presented. LRSA dynamically finds the equilibrium in the Metropolis cycle by using Linear Regression. Experimentation shows that the proposed method is more efficient than the classical one, since it obtains the same quality in the final solution with less processing time.

1. Introduction

It is well known that the Satisfiability (SAT) problem belongs to NP class [1] and Simulated Annealing (SA) is one of the most efficient for solving problems of this class. Since SA was proposed by Kirkpatrick et al. [2] and Cerny [3], commonly the SA implementations have two cycles: the temperature and the metropolis cycles; therefore SA can be considered as an extension of the Metropolis algorithm [4]. In fact SA is applied in solving other NP problems as well. However SA finds the optimum or near-to-optimum solutions with a faster convergence rate if it is well tuned. The determination of a well cooling scheme of SA is a main issue in SA due to its efficiency (measured by its execution time) and its effectiveness (measured by its quality solution). The cooling scheme of SA is given by the temperature parameters used in the external cycle as well as the function temperature ($f(c_k)$) that determines the way that the initial temperature (c_i) is decreased until the final temperature

(c_f) is reached. Another important tuning parameter is the maximum length of the Markov chain (L_k) establishing the length of the Metropolis cycle. For each value of the control parameter c_k (temperature in the k metropolis cycle), SA accomplishes a certain number of Metropolis decisions, each of them establishing whether or not a new solution is accepted according to the Boltzmann distribution. In this sense, in order to obtain a better performance of SA a relation between any k temperature c_k and the length L_k of the Metropolis cycle must be correctly determined. Because of this, experimental and analytical parameters tuning strategies are currently being studied. By using experimental strategies, the parameter values of the function of temperature have been widely studied for the main cooling schemes:

- Geometrical $c_{k+1} = \alpha c_k$
- Exponential: $c_{k+1} = \exp(-\alpha) c_k$ and
- Logarithmic: $c_{k+1} = c_k / \ln(\alpha)$

Where α is a parameter and k is the iteration number of the metropolis cycle (k is i or f for the initial and final temperature respectively). Reasonable α values have been previously determined through experimentation [5]; on the other hand, analytical tuning methods have been proposed to determine c_i and c_f as well as the relation between c_k and L_k published for the geometrical cooling scheme [6]; in this method L_k represents the expected value (or an average value) of the number of iterations required for the k metropolis cycle to reach its stochastic equilibrium [6]. If each k Metropolis cycle is stopped before L_k a poor quality solution can be obtained; in contrast if it is stopped after L_k , the execution time is increased without a significant quality solution.

As was pointed out before, the objective of these tuning methods is to find a better way to reduce the required execution time in good quality in the final solution. However it is known that a good SA performance is obtained only if the Metropolis cycle is stopped when it really reaches the stochastic equilibrium.

In this paper an adaptive method that establishes the length of each Markov chain in a dynamic way into a new SA algorithm is proposed. The paper proposes to use Linear Regression in every Metropolis cycle to detect its stochastic equilibrium. The new method named Linear Regression Simulated Annealing (LRSA) also tunes the initial and final through an analytical model. This method is implemented in an algorithm (also named LRSA) which is tested with SAT instances and is compared against those obtained by an SA algorithm that uses the average L_k value (TMSA: Tuning with the Markov model the Simulated Annealing algorithm) [6]. Experimentation shows that LRSA obtains a similar quality solution but the execution time is considerable reduced.

2. Background

In this section we are presenting the mathematical background of SAT, Markov Chains, Cooling Function and Neighborhood.

2.1 SAT problem

SAT was the first problem referred to be as NP-complete [7] and it is fundamental to the analysis of the computational complexity of many problems [8].

An instance of SAT is a boolean formula which consists of the following components:

- A set of n variables $x_1, x_2, \dots, x_n$.
- A set of literals; a literal is a variable x_i or its negation $\neg x_i$.
- A set of m clauses: $C_1, C_2, \dots, C_m$ linked by the logical connective AND ($\wedge$) where each clause consists of literals linked by the logical connective OR ($\vee$).

Then a SAT instance Φ which can be written as follows:

$$\Phi = C_1 \wedge C_2 \wedge \dots \wedge C_m \quad (1)$$

Using the last notation, the SAT problem can be enunciated as follows:

Definition 1. Given a finite set $\{C_1, C_2, \dots, C_m\}$ of clauses, determine whether there is an assignment of truth-values to the literals appearing in the clauses which makes all the clauses true.

2.2 Neighborhood, Markov Chains and Cooling Function

In [9] it is established that SA requires a well defined neighborhood structure and the value of the maximum and the minimum cost incremented in the objective function can be used to calculate c_i and c_f temperatures. The solution selected from i - can be any solution j - defined by the following neighborhood scheme:

Definition 2.-Let

$\{ \forall S_i \in S, \exists \text{ a set } V_{S_i} \subset S \mid V_{S_i} = V : S \rightarrow S \}$ be the neighborhood of a solution S_i where V_{S_i} is the neighborhood set of S_i , $V : S \rightarrow S$ is a mapping and S is the solution space of the problem being solved.

It can be seen from the above definition that neighbors of a solution S_i only depend on the neighborhood structure V established for a specific problem. Once V is defined, the maximum and minimum cost deteriorations can be written as:

$$\Delta Z_{V_{\max}} = \text{Max} \{ Z(S_j) - Z(S_i) \} \quad (2)$$

$$\forall S_j \in V_{S_i}, \forall S_i \in S$$

$$\Delta Z_{V_{\min}} = \text{Min} \{ Z(S_j) - Z(S_i) \} \quad (3)$$

$$\forall S_j \in V_{S_i}, \forall S_i \in S$$

Where $Z(S)$ is the cost associated to S . Then the initial and final temperatures c_i and c_f are [9]:

$$c_i = \frac{-\Delta Z_{V_{\max}}}{\ln(P_A(\Delta Z_{V_{\max}}))} \quad (4)$$

$$c_f = \frac{-\Delta Z_{V_{\min}}}{\ln(P_A(\Delta Z_{V_{\min}}))} \quad (5)$$

The c_k control parameter is set by a cooling function like $c_{k+1} = f(c_k)$. At the beginning of the process the temperature has a high value and the probability of accepting one proposed solution is high. When the

temperature decreases this probability also decreases and only good solutions are accepted at the end of the process. In this sense every Markov chain makes a stochastic walk in the solution space until the stationary distribution is reached. In general L_k can be established as:

$$L_k \leq L_{\max} = g(|V_{S_i}|) \quad (6)$$

In which $g(|V_{S_i}|)$ is a function that gives the maximum number of samples that must be taken from the neighborhood V_{S_i} in order to evaluate an expected fraction of different solutions at c_f . Usually an SA algorithm uses a uniform probability distribution function given by a random replacement sampling method to explore V_{S_i} at any temperature c_k [6]. In this way, the probability of getting the solution S_j in N samples is:

$$P(S_j) = 1 - \exp\left(-\frac{N}{|V_{S_i}|}\right) \quad (7)$$

Therefore in any SA algorithm the maximum Markov chain length (when $c_k = c_f$) may be set as:

$$L_{\max} = -\ln(1 - P(S_j))|V_{S_i}| \quad (8)$$

Moreover an exploration level C can be defined as $C = -\ln(P_R(S_j))$ and $L_{\max} = C|V_{S_i}|$. C ranges from 1 to 4.6 which guarantees a good exploration level of the neighborhood at the final temperature, if 99% of the solution space is going to be explored, the rejection probability is 0.01 and $C=4.6$; if 95% of the solution space will be explored then $C=3$.

At the beginning of the process ($c_k = c_i$), any solution has the same acceptance probability. Therefore, the first Markov chain length must be very small ($L_1 \approx 1$) because its stationary probabilistic state is reached in only one iteration. When k is increased, c_k is decremented until it reaches c_f . Therefore, for consecutive values of c_k , SA is forced to increment its Markov chain length in order to reach its stationary probabilistic state. Thus, L_k is incremented from one at c_i to $L_{\max}$ at c_f . An incremental Markov chain function can be proposed as: $L_{k+1} = \beta L_k$ where $\beta > 1$ and L_k is the length of the Markov chain at c_k , L_{k+1} representing the length of the Markov chain at c_{k+1} and β is the increment coefficient.

Because the Markov chain length is incremented from L_1 to $L_{\max}$ when c_k varies from c_i to c_f in a Markov process we have $L_{\max} = \beta^n L_1$ and:

$$\beta = \exp\left(\frac{\ln(L_{\max}) - \ln(L_1)}{n}\right) \quad (9)$$

For each cooling scheme (Geometrical, Exponential and Logarithmic) the n number of steps performed from c_i to c_f can be calculated from the next formulae derived from their cooling function:

$$n = \frac{\ln(c_f) - \ln(c_i)}{\ln(\alpha)} \quad (10)$$

$$n = \frac{\ln(c_i) - \ln(c_f)}{\alpha} \quad (11)$$

$$n = \frac{\ln\left(\frac{c_i}{c_f}\right)}{\ln(\ln(\alpha))} \quad (12)$$

1. Initialization ($c_i, c_f, L=1, S_p, \beta$)
2. $c = c_i$
3. Repeat
4. $k=0$
5. Repeat
6. Generate S_i
7. if $E(S_i) < E(S_j)$
8. $S_i = S_j$
9. else
10. if (random < Boltzman)
11. $S_i = S_j$
12. $k++$
13. While ($k \leq L$)
14. $c = *c$
15. $L = \beta * L$
16. While ($c > c_f$)

Fig. 1. TMSA pseudocode.

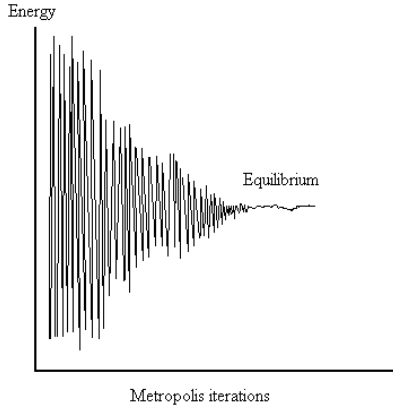


Fig. 2. Energy variation in Metropolis cycle.

3. Modification to the TMSA algorithm

The initial temperature c_i , the final temperature c_f and the Markov chain length increment (β) can be calculated from the Expressions 4, 5 and 8 respectively, which has been previously published and implemented in TMSA algorithm [6, 9]; this algorithm is shown on figure 1.

In the Metropolis Cycle, the typical behavior of energy (i.e. the objective function for a given temperature) can be observed in Fig. 2; the Metropolis Cycle reaches the equilibrium average at the L_k value, but probably more iterations are required by Metropolis to really reach equilibrium. The equilibrium zone can be found by applying a linear regression (LR) method into the Metropolis Cycle. In this case, LR fits the energy values to a straight line by using a simple minimum square method.

Let us suppose LRSA is running with $C=4.6$ (i.e 99 percent exploration level) and L_{max} is obtained by Equation 8. In this case LRSA obtains L_{max-1} using $C=3.0$ (i.e 95 percent exploration level). LRSA starts the LR's application just when the iteration L_{max-1} is reached, and the slope of the straight line is obtained by applying a minimum square method. When the slope of this line is equal or smaller than zero LRSA infers that the stochastic equilibrium is reached and the metropolis cycles' number k should be stopped. Then the same approach is applied to the next metropolis cycle.

Assuming that the data set consists of points (x_i, y_i) with $i=1,2,...,n$. The goal is to find a function f such that $f(x_i) \approx y_i$. The linear function required is defined as $f(x_i) = ax_i + b$, where a and b are

parameters that should be determined. The values of a and b are determined by minimizing the sum of the squares of the residuals:

$$S = \sum_{i=1}^n (y_i - f(x_i))^2 \quad (13)$$

Differentiating with respect to a and b we obtain the following system of equations:

$$\sum_{i=1}^n x_i^2 \times a + \sum_{i=1}^n x_i \times b = \sum_{i=1}^n x_i y_i \quad (14)$$

$$\sum_{i=1}^n x_i \times a + \sum_{i=1}^n 1 \times b = \sum_{i=1}^n y_i \quad (15)$$

From Equations 14 and 15 the slope a is trivially obtained:

$$a = \frac{Cn - BD}{An - B^2} \quad (16)$$

where:

$$A = \sum_{i=1}^n x_i^2, B = \sum_{i=1}^n x_i, C = \sum_{i=1}^n x_i y_i \text{ and } D = \sum_{i=1}^n y_i \quad (17)$$

1. Initialization ($c_i, c_f, L=1, S_i, \beta$)
2. $c = c_i$
3. Repeat
4. $k=0$
5. Repeat
6. Generate S_i
7. if $E(S_i) < E(S_i)$
8. $S_i = S_j$
9. Calculate C' and D' (Equation 18)
10. else
11. if (random < Boltzman)
12. $S_i = S_j$
13. $k++$
14. if ($k > L_{max-1}$)
15. Calculate a (Equation 18)
16. if ($a \leq 0$) exit loop
17. While ($k \leq L$)
18. $c = *c, L = \beta * L$
19. While ($c > c_f$)

Fig. 3. LRSA algorithm.

But $x_i = 1, 2, \dots, n$, so the final expressions are:

$$a = \frac{12C' - 6(n-1)D'}{n^3 - n} \text{ where} \quad (18)$$

$$C' = \sum_{i=1}^n iE_i \text{ and } D' = \sum_{i=1}^n E_i$$

Table 1. SAT instances tested.

SAT Instance	Id	σ	Sat?
aim-50-1_6-yes1-3	a2	1.60	Y
aim-200-1_6-no-1	a7	1.60	N
aim-50-1_6-no-2	a8	1.60	N
aim-50-2_0-no-4	a10	2.00	N
aim-50-2_0-yes1-1	a3	2.00	Y
aim-50-2_0-no-3	a9	2.00	N
g2_V100_C200_P2_I1	g1	2.00	Y
dubois26	d1	2.67	N
dubois21	d2	2.67	N
dubois27	d3	2.67	N
BMS_k3_n100_m429_161	b1	2.83	Y
g2_V300_C900_P3_I1	g15	3.00	Y
g2_V50_C150_P3_I1	g17	3.00	Y
BMS_k3_n100_m429_368	b2	3.08	Y
hole6	h2	3.17	N
par8-1	p1	3.28	Y
aim-50-3_4-yes1-2	a4	3.40	Y
hole7	h3	3.64	N
par8-3-c	p2	3.97	Y
g2_V100_C400_P4_I1	g3	4.00	Y
hole8	h4	4.13	N
uuf225-045	u4	4.27	N
RTI_k3_n100_m429_150	r1	4.29	Y
uf175-023	u1	4.30	Y
uuf100-0789	u8	4.3	N
uf50-01	u7	4.36	Y
uuf50-01	u9	4.36	N
ii8a2	i3	4.44	Y
g2_V50_C250_P5_I1	g19	5.00	Y
hole10	h1	5.10	N
ii32e1	i2	5.34	Y
anomaly	a6	5.44	Y
aim-50-6_0-yes1-1	a5	6.00	Y
g2_V50_C300_P6_I1	g20	6.00	Y
jnh201	j1	8.00	Y
jnh215	j3	8.00	N
medium	m1	8.22	Y
jnh301	j2	9.00	Y

In the case of SA, the energy E_i of the system is equal to the number of satisfied clauses. LRSA applies the same approach of TMSA algorithm to determine the other tuning parameters (i.e. initial and final

temperature, and β value). Figure 3 shows the LRSA algorithm.

4. Computational Results

In order to test LRSA we used SAT instances shown in Table 2. Table 2 shows several SAT instances with different σ relations of clauses/variables [10] which were taken from SATLIB [11] or generated with Hories algorithm [12].

The measurement of the efficiency of LRSA algorithm was based on the execution time and the solution quality. The solution quality of an instance was measured as the number of “true” clauses in an instance at the end of the program execution. Both algorithms (LRSA and TMSA) were implemented in a Dell Latitude Laptop with 1Gb of RAM and Pentium 4 processor running at 2.13 GHz.

Each instance was executed 40 times and the average execution time and the average quality solution were obtained. The quality of the solution was established with the following expression:

$$Q = \frac{\text{clauses true}}{\text{total clauses}} \times 100 \quad (17)$$

Table 2. Q_q and Q_t average for each cooling schemes. In this table with C value, LRSA algorithm obtains $L_{\max}$ and with CLRSA value LRSA obtains $L_{\max-1}$ using Equation 8.

C=2.0, C _{LRSA} =1.0		
	Q_q	Q_t
Geometrical	99.50	24.12
Exponential	99.45	23.60
Logarithmic	99.38	23.35
Average	99.44	23.69
C=3.0, C _{LRSA} =2.0		
	Q_q	Q_t
Geometrical	99.49	27.17
Exponential	99.44	26.50
Logarithmic	99.46	26.37
Average	99.46	26.68
C=4.6, C _{LRSA} =3.0		
	Q_q	Q_t
Geometrical	99.57	27.64
Exponential	99.40	27.18
Logarithmic	99.52	26.97
Average	99.50	27.26

LRSA results were compared to those of TMSA, by using two quotients denominated quality improvement Q_q and time improvement Q_t :

$$Q_q = \frac{Q_{LRSA}}{Q_{TMSA}} \times 100 \quad (18)$$

$$Q_t = \frac{Time_{LRSA}}{Time_{TMSA}} \times 100 \quad (19)$$

Table 2 shows that LRSA maintains the quality of solution values very similarly to those obtained with TMSA in each cooling scheme (96.47%), but LRSA execution times are always lower than those obtained by TMSA (25.87% in average).

5. Conclusions

In this paper a new LRSA algorithm for solving SA is proposed. LRSA uses a Linear Regression (LR) procedure by applying the classical minimum square method as a way to find the stochastic equilibrium zone in the Metropolis cycle. As a result the execution time is considerably lower than a classical SA with all other tuning parameters fixed exactly at the same value. The paper also presents relations for the temperature parameter and the expected value of the length of the Metropolis cycle with several cooling schemes. Experimentation presented in the paper for these three cooling schemes shows that LRSA is more efficient than a classical TMSA where the stochastic equilibrium of the Metropolis cycle is not detected during its execution.

6. References

[1] P. Crescenzi, V. Kann, "How to find the best approximation results a follow-up to Garey and Johnson", *ACM SIGACT, News*, 1998, pp. 90–97.

- [2] S. Kirkpatrick, C.D. Gelatt and M.P. Vecchi, "Optimization by Simulated Annealing", *Science*, Number 4598, 13 May 1983, pp. 671–680.
- [3] V. Cerny, "Thermodynamical approach to the traveling salesman problem: An efficient simulation algorithm", *Journal of Optimization Theory and Applications*, 45, 1985, pp. 41–51.
- [4] N. Metropolis, A.W. Rosenbluth, M.N. Rosenbluth, A.H. Teller, A.H. and E. Teller, "Equation of State Calculations by Fast Computing Machines", *Chem. Phys.* **21**, 1953, pp. 1087–1092.
- [5] J. Frausto-Solis and F. Martinez-Rios, "Golden Ratio Annealing for Satisfiability Problems using Dynamically Cooling Schemes", 17th International Symposium on Methodologies for Intelligent Systems (ISMIS'08), Toronto, Canada, May 20, (2008), Accepted as a full paper for publication in ISMIS 2008 proceedings will appear in Springer's Lecture Notes in Artificial Intelligence (LNAI).
- [6] J. Frausto, H. Sanvicente and F. Imperial, "ANDYMARK: An analytical Method to Establish Dynamically the Length of the Markov Chain in Simulated Annealing for the Satisfiability Problem", *Springer Verlag*, 2006, ISSN: 0302-9743.
- [7] S.A. Cook, "The complexity of theorem proving procedures", In: *Proceedings of 3rd Annual ACM symposium on the Theory of Computing*, ACM, 1971, pp. 151–158.
- [8] C.H. Papadimitriou, *Computational Complexity*. Addison Wesley Longman, 1995
- [9] H. Sanvicente and J. Frausto, "A Method to Establish the Cooling Scheme in Simulated Annealing Like Algorithms", In *ICCSA 2004*, Assisi, Italy, May 14-17, 2004, pp. 755–763.
- [10] M. Mezard, G. Parisi and R. Zecchina, "Analytic and algorithmic solution of random satisfiability problems", *Science*, June 27, 2002.
- [11] J.E. Beasley, "OR Library", Imp. College, Management School, <http://mscmga.ms.ic.ac.uk/info.html>.
- [12] S. Horie, and O. Watanabe, "Hard instance generation for SAT", Technical Report TR97-0007, Dept. of Computer Science, Tokyo Inst. of Tech., 1997 (Available from CS Dept. TR Archive; the extended abstract appeared in Proc. ISAAC'97, *Lecture Notes in Computer Science* 1350).

Equivalence of the constriction factor and inertia weight models in particle swarm optimization: A geometric series analysis

Julio Barrera
julio.barrera@gmail.com

Juan J. Flores
juanf@umich.mx

División de Estudios de Posgrado
Facultad de Ingeniería Eléctrica
Universidad Michoacana de San Nicolás de Hidalgo

Abstract

In this work we show the equivalence of the models of constriction factor and weighted inertia for particle swarm optimization. This equivalence is shown through expressing the formula for updating a particle's velocity as a geometric series in both models. We also present the mathematical basis for the convergence conditions on an unified model based on geometric series. These mathematical results tell us directly what are adequate numeric values for the model's parameters, which is a very useful pragmatic result.

1 Introduction

Evolutionary algorithms have been used in the last years for solving optimization problems, their application has been successful in problems where traditional optimization algorithms do not produce good results or can not be applied. In particular, the particle swarm algorithm have gained interest from the research community. The particle swarm algorithm (PSO) was introduced by Kennedy and Eberhart [1] "inspired by social behavior of bird flocking or fish schooling" it differs from other evolutionary algorithms, like genetic algorithms, in a simple rule for updating the particles' "position" from one generation to the next, to reach an optimum.

Initially the velocity of a particle is zero. In each generation, a velocity is computed and added to the current velocity of a particle, a component of velocity directed towards the optimum in the swarm (social component) and an other one directed towards the point with the best fitness value recorded at the moment by the particle (cognitive component). Then the position of a particle is updated by adding its velocity to its current position. But this simple rule of

update has a drawback, the velocity of the particle is accumulated and it can increase and diverge.

Since the presentation of the PSO algorithm various methods have been developed to overcome the divergence of velocity; limiting the velocity is the simplest method and consists of truncating the velocity of a particle if it passes a given limit. Later, two models were developed to reduce the velocity of a particle in each generation by multiplying it by a factor [1], both models have been reviewed as different models and their differences have been addressed in various works [2]. Although in some circumstances it has been observed they behave "somewhat similar" [3] or it is believed that mathematically, both update rules are equivalent if certain conditions are achieved.

In this paper we formalize the equivalence of the inertia weight and constriction factor models, by writing the update rules of each model as a geometric series. Using the results of geometric series we state the conditions for convergence, and show a graphical method for selection of the inertia weight factor.

The rest of the paper is organized as follows: Section 2 describes the models for limiting the velocity of a particle, Section 3 examines the equivalence of the models of inertia weight and constriction factor, in Section 4 we state the results of geometric series that guarantee convergence in models that can be expressed as geometric series, and in Section 5 we wrap up with the conclusions of the paper and the directions of research in future work.

2 Limiting particles' velocity

An issue with particle swarm optimization is that particles tend to "fly" away not only from the optimum of the function, but from the search space itself. From the beginning of its development, the PSO algorithm has been improved; the bare bones version of PSO has two update rules for velocity and position of a point in the search space

$$v_{t+1} = v_t + v_{ct} \quad (1)$$

$$x_{t+1} = x_t + v_{t+1} \quad (2)$$

where

$$v_{ct} = C_1 r_1 (x_g - x_t) + C_2 r_2 (x_p - x_t) \quad (3)$$

C_1 and C_2 are the learning constants, r_1 and r_2 are random numbers, x_g is the position of the global best particle, x_p is the position where the particle had the best value of fitness through the iterations, and x_t is the current position.

In this model, update Equation 1 tends to increase velocity as t increases, so the particle does not converge to an optimum of the function. One attempt to avoid non-convergence is to limit the velocity magnitude.

$$v_{t+1} = \begin{cases} v_{t+1}, & \text{if } v_{t+1} < v_{max} \\ v_{max}, & \text{otherwise} \end{cases} \quad (4)$$

Another model developed by Eberhart [4] incorporates an inertia weight factor to the velocity update rule

$$v_{t+1} = \omega v_t + v_{ct} \quad (5)$$

The inertia weight factor provides a way to regulate the exploration and exploitation of the particles. Based on experimental results Eberhart suggests that the inertia weight must be less than 1.

In the work of Clerk [5] he studies the particle swarm optimization algorithm as a dynamic system, and proposes the constriction factor model, where the velocity update rule is

$$v_{t+1} = \chi(v_t + v_{ct}) \quad (6)$$

where

$$\chi = \frac{2\kappa}{|2 - \phi - \sqrt{\phi^2 - 4\phi}|} \quad (7)$$

$\phi = C_1 + C_2$, κ is an arbitrary constant that is used to adjust the value of χ and it takes on values from $(0, 1]$. A detailed description of the constriction factor can be found in [5]. A restriction on the computation of the constriction factor is that $\phi \geq 4$, if $\phi = 4$ and $\kappa = 1$, we have $\chi = 1$ and the model becomes the bare bones PSO. If one of the values (or both) of the learning coefficients is greater than 2, the particles oscillate.

3 Equivalence of inertia weight and constriction factor

We first represent particles' velocity evolution through time as a geometric series in both models: inertia weight and constriction factor. In the inertia weight model the evolution of the velocity is determined by the equation

$$v_{t+1} = \omega v_t + v_{ct} \quad (8)$$

where v_{ct} is the same as in Equation 3. In the initialization of the swarm all particles have zero velocity. That is, for the first evaluation at time $t = 1$, $v_1 = 0$ and

$$v_2 = \omega v_1 + v_{c1} = v_{c1} \quad (9)$$

for $t = 2$

$$\begin{aligned} v_3 &= \omega v_2 + v_{c2} \\ &= \omega(v_{c1}) + v_{c2} \\ &= \omega v_{c1} + v_{c2} \end{aligned} \quad (10)$$

for $t = 3$

$$\begin{aligned} v_4 &= \omega v_3 + v_{c3} \\ &= \omega(\omega v_{c1} + v_{c2}) + v_{c3} \\ &= \omega^2 v_{c1} + \omega v_{c2} + v_{c3} \end{aligned} \quad (11)$$

and for any given t we can write v_t as

$$v_{t+1} = \omega^{(t-1)} v_{c1} + \dots + \omega v_{c(t-1)} + v_{ct} \quad (12)$$

Now, for the constriction factor model we have the evolution rule for velocity

$$v_{t+1} = \chi(v_t + v_{ct}) \quad (13)$$

where again v_{ct} is computed according to Equation 3. The initial velocity at $t = 1$ is $v_1 = 0$ and

$$v_2 = \chi(v_1 + v_{c1}) = \chi v_{c1} \quad (14)$$

for $t = 2$

$$\begin{aligned} v_3 &= \chi(v_2 + v_{c2}) \\ &= \chi(\chi v_{c1} + v_{c2}) \\ &= \chi^2 v_{c1} + \chi v_{c2} \end{aligned} \quad (15)$$

for $t = 3$

$$\begin{aligned} v_4 &= \chi(v_3 + v_{c3}) \\ &= \chi(\chi^2 v_{c1} + \chi v_{c2} + v_{c3}) \\ &= \chi^3 v_{c1} + \chi^2 v_{c2} + \chi v_{c3} \end{aligned} \quad (16)$$

If we continue the process in the same way, we obtain the expression for v_{t+1}

$$v_{t+1} = \chi^t v_{c1} + \chi^{t-1} v_{c2} + \cdots + \chi^2 v_{c(t-1)} + \chi v_{ct} \quad (17)$$

Observing Equations 12 and 17, the only difference between both models is a power of their corresponding constants ω , and χ ; the inertia weight model contain a “constant” velocity factor v_{ct} that is not multiplied by a power ω , and the highest power of ω is $t - 1$. In the constriction factor all velocities are multiplied by a power of χ , and the highest power of χ is t .

From Equation 3 we can factor a constant value by multiplying by 1

$$\begin{aligned} v_{ct} &= C_1 r_1 (x_g - x_t) + C_2 r_2 (x_p - x_t) \\ &= \frac{C_3}{C_3} C_1 r_1 (x_g - x_t) + \frac{C_3}{C_3} C_2 r_2 (x_p - x_t) \\ &= C_3 \left[\frac{C_1}{C_3} r_1 (x_g - x_t) + \frac{C_2}{C_3} r_2 (x_p - x_t) \right] \end{aligned} \quad (18)$$

If we choose $C_3 = 1/\chi$ and we factor each v_{ct} in Equation 17, we have

$$\begin{aligned} v_{t+1} &= \chi^t (1/\chi) v'_{c1} + \chi^{t-1} (1/\chi) v'_{c2} + \cdots \\ &\quad + \chi^2 (1/\chi) v'_{c(t-1)} + \chi (1/\chi) v'_{ct} \\ &= \chi^{t-1} v'_{c1} + \chi^{t-2} v'_{c2} + \cdots \\ &\quad + \chi v'_{c(t-1)} + v'_{ct} \end{aligned} \quad (19)$$

Where v'_{ck} have new learning constants $C'_1 = \chi C_1$ and $\chi C'_2$. Equation 19 is the same expression that we derived for the inertia weight model described by Equation 12. Thus both models have the same form at time t , so both models are equivalent.

4 Convergence

Equation 12 it can be expressed as a series

$$\begin{aligned} v_{t+1} &= \omega^{(t-1)} v_{c1} + \cdots + \omega v_{c(t-1)} + v_{ct} \\ &= \sum_{n=0}^{t-1} \omega^n v_{c(t-n)} \end{aligned} \quad (20)$$

Which is a geometric series. Results for geometric series are well known [6, 7]; if $\sum r_n$ is a geometric series

$$\sum r_n = \sum_{n=0}^{\infty} ar^n \quad (21)$$

with a a fixed constant, the series converge if $|r| < 1$, else (i.e. $|r| \geq 1$) the series diverges. An other known result is that, if we have two series $\sum r_n$, and $\sum s_n$, and $s_n > r_n$ for all n , if $\sum s_n$ converges, then $\sum r_n$ converges, this is called comparison test. With these two results we can prove that models like inertia weight and constriction factor always converge if the factor (inertia or constriction) is less than 1.

The results for convergence stated above are valid for $a, r_n \in R$, but it can be easily extended for $r_n \in R^k$. If we have a series $\sum r_n$, with $r_n \in R^k$, a_n converges if each $r_{n,i}$ for $i = 1, \dots, k$ converges, i.e. each coordinate of r_n converges. In the case of the inertia weight (or constriction factor) the search space is frequently a subset of R^k , and we have the product of a scalar and a vector in the update rule

$$\begin{aligned} \omega v_{ct} &= \omega(v_{ct,1}, v_{ct,2}, \dots, v_{ct,k}) \\ &= (\omega v_{ct,1}, \omega v_{ct,2}, \dots, \omega v_{ct,k}) \end{aligned} \quad (22)$$

for a given t , each coordinate of the velocity can be viewed as a geometric series.

To prove the convergence of the inertia weight model we compare Equations 20 and 21; ω is the factor r in Equation 21, but the factor a is required to be constant and in the case of factors $v_{ct,i}$ is likely to change its value for each t . To use the results on convergence of series we need that $|r| < 1$ or in the case of inertia weight that $|\omega| < 1$. The factor a is needed to be constant as stated above, but if we can find a bound v_c for v_{ct} , that is $v_{c,i} > v_{ct,i}$ for $i = 1, \dots, k$, we have the series

$$\sum \omega'_t = \sum_{i=0}^t v_c \omega^i \quad (23)$$

were $v_{c,i} \omega^t > v_{ct,i} \omega^t$ for $i = 1, \dots, k$, and as mentioned in the beginning of the section (comparison test), as $\sum \omega'_t$ converges, Equation 20 converges.

4.1 Computing the constriction factor

We have shown that the constriction factor and inertia weight models are equivalent, and proved that any model that can be expressed as a geometric series converges if the multiplying factor is less than 1. A question arises: is necessary to compute the constriction factor using Equation 7? Following the results of previous sections the answer is no.

We shown that the constriction factor model can be expressed in the same way as the inertia weight model, and one condition for convergence is that χ (or ω) is less than 1. Thus we do not need to compute the constriction factor using Equation 7. One of the side effects of computing the constriction factor from Equation 7 is that the sum of the learning coefficients is required to be greater than 4. This restriction produces an unnecessary oscillation of the particles, research works have been done for reducing it [3, 8, 9]. As stated above, it is not necessary to compute the constriction factor from Equation 7 nor is it needed to restrict the learning coefficients.

In this section we state two conditions for convergence of velocity in inertia weight model: $|\omega| < 1$ and we need to find a bound for all v_{ct} . For the bound of v_{ct} the limits of the search space can be used to ensure convergence. Moreover, we can ensure convergence without limiting the values of the learning constants.

5 Conclusions and future work

We have presented a mathematical argument to show the equivalence of inertia weight model and constriction factor, not only the update rule of velocity in both models can be written as a geometric series, given learning constants for constriction factor we can compute values for the inertia weight model that produce identical numerical values. By writing each step in the process of computing the velocity of a particle as a geometric series we have exposed the conditions for convergence of the velocity and shown that neither the computing of constriction factor is necessary, nor it is the restriction of the learning coefficients.

References

- [1] Kennedy, J., Eberhart, R.C.: Particle swarm optimization. Proceedings of IEEE International Conference on Neural Networks (1995) 1942–1948
- [2] Kennedy, J., Eberhart, R.C., Shi, Y.: Swarm intelligence. Morgan Kaufmann, San Francisco (2001)
- [3] Lam t. Bui, O.S., Abbass, H.A.: A modified strategy for the constriction factor in particle swarm optimization. Lecture Notes in Computer Science **4828/2007** (2007) 333–344
- [4] Shi, Y., Eberhart, R.C.: A modified particle swarm optimizer. Proceedings of the IEEE Congress on Evolutionary Computation (CEC 1998) (1998)
- [5] Clerc, M., Kennedy, J.: The particle swarm: explosion, stability, and convergence in a multidimensional complex space. IEEE Transactions on Evolutionary Computation **6** (2002) 58–73
- [6] Rudin, W.: Principles of Mathematical Analysis. McGraw-Hill, New York (1976)
- [7] Apostol, T.M.: Mathematical Analysis. Addison-Wesley, Massachusetts (1974)
- [8] Suganthan, P.N.: Particle swarm optimiser with neighbourhood operator. Evolutionary Computation, 1999. CEC 99. Proceedings of the 1999 Congress on Evolutionary Computation **3** (1999) 1958–1962
- [9] Fan, H.: A modification to particle swarm optimization algorithm. Engineering Computations **19** (2002) 970–989
- [10] Torn, A., Zilinskas, A.: Global optimization. Lecture Notes in Computer Science **350** (1989)
- [11] H. Muhlenbein, D.S., Born, J.: The parallel genetic algorithm as function optimizer. Parallel Computing **17** (1991) 619–632

On the Path Towards Autonomic Computing: Combining Swarm Intelligence and Excitable Media Models

Luis M. Fernández-Carrasco, Hugo Terashima-Marín, Manuel Valenzuela-Rendón

Center for Intelligent Systems

Instituto Tecnológico y de Estudios Superiores de Monterrey

Monterrey, México

{A00789695, terashima, valenzuela}@itesm.mx

Abstract

The present article presents an approach towards the emergent field of autonomic computing by combining swarm intelligence and excitable media. It is believed that the combination of these scientific fields is capable of generating two important self-★ properties required by autonomic systems: self-organization and self-configuration.

An agent-based simulation is used to generate swarms, which are then thought as computing tasks; the excitable media is, on the other hand, implemented as a cellular automaton, which is what excites swarms to show autonomic computing features.

The results show that combining swarm intelligence and excitable media is a feasible way towards autonomic computing and its desired characteristics, reducing systems' complex design and implementation.

1. Introduction

Autonomic computing is an emergent field in the research world that aims at solving the current complexity that new computing systems show at the design and implementation phases. This issue is due to the recent and constant advances in networking and computing technologies, which has lead to an explosive growth in applications and information services [16]. Moreover, the underlying information infrastructure (e.g., the Internet) aggregates large numbers of independent computing and communication resources, which is itself similarly large, heterogeneous, dynamic, and complex [16]. As a consequence, the need for a new vision for system application and design has emerged, a vision that is currently based on strategies used by biological systems when dealing with similar challenges. This new paradigm has been called autonomic computing [15].

The autonomic computing paradigm is inspired by the

human autonomic nervous system and its main goal is to have computer and software systems that can manage themselves [16]. Autonomic systems are characterized by their self-★ properties including *self-configuration* and *self-organization* [16]¹. This document, therefore, explores the possibility to achieve such properties by combining two nature-inspired techniques: *swarm intelligence* and *excitable media*.

A swarm is defined as a set of (mobile) agents which are liable to communicate directly or indirectly with each other [6]. Following this concept, many researchers have been able to simulate termite, ant and bee's behavior by implementing simple algorithms [6]. As such, researchers have good reasons to find swarm intelligence appealing as software systems have become so intractable that they can no longer be managed by any human. This methodology offers an alternate way of designing *intelligent* systems, in which autonomy, emergence, and distributed functioning replace control, preprogramming, and centralization [1]. These characteristics are perfect for achieving autonomic systems.

On the other hand, and based on life itself, excitable media is an attempt to provide a mathematical model to excitation waves. Many successful models show the common property of treating any *cell* as an *automaton* [12]. An automaton is basically a black box which receives information from the surrounding environment and generates output accordingly. In mathematics, the automata are often considered to be discrete [19]. However, automata with gradually varying input and output signals and a continuous temporal evolution may also be covered. Moreover, automata with stochastic dynamics are possible [12].

When an automaton which mimics the individual behavior of a biological organism is constructed, a population of such automata can be considered whose members communicate by generating, sending and receiving signals [12]. As

¹Self-★ refers to self-configuration, self-optimization, self-healing and self-protection.

such, excitable media is a communicating automaton. If the swarm is placed on this communicating automaton, then the swarm can be *excited* to behave in such a way that self- $\star$ properties emerge.

Thus, this document explores an approach towards autonomic computing by having swarm agents live in an excitable environment based on cellular automata. These swarm agents react according to the levels of excitation, instead of chemical-based communication models, and, through this excitation, self-organization and self-configuration is achieved. There are not any related work that follows this same approach. Most publications in the excitable media field are in the biological and medical areas. Some examples are the works of Romanes [17], Mayer [10] in the biology field. In the medical area, the publications of Mines [14], Lewis [7], Garrey [5], and Wiener and Rosenblueth [18] are some examples. For swarm intelligence there is the book written by Bonabeau et al [1].

Another related work is Eiben's article [4] that provides an overall look at the link between nature-based models and autonomic computing; whereas Mamei et al [9] offers a self-organizing methodology called the TOTA approach. Another article tackling self-organization is provided by d'Inverno and Saunders [3], where an agent-based modeling is used. However, the proposed approach in this article makes use of excitable media as the engine to generate self- $\star$ properties for autonomic systems.

The next sections of this paper present a general description of what excitable media is; then, the mathematical model used in this implementation is explained. After, the agent-based implementation where swarms are placed is described. Finally, the conducted experiments, the obtained results and the conclusions of this novel approach are presented.

2. What is Excitable Media?

Excitable media refers mainly to spatially distributed systems which have the ability to propagate signals without damping. Let us look at a forest fire, for instance, which travels as a wave from its initiation point, and regenerates with every tree it ignites [2]. This is in contrast to passive wave propagation, which is characterized by a gradual damping of signal amplitude due to friction.

An impulse over a certain threshold initiates a wave of activity moving across the excitable media. As each element undergoes an excursion from steady state, it causes its neighbors to move over threshold at a rate determined by the diffusion coefficient (a *passive* property of the media), and the rate of rise of the diffused species of the excited element (an *active* property of the media). Figure 1 shows this idea graphically where the arrival of a perturbation initiates a transition from the rest state (white) into the estate of ex-

citation (dark grid). After that, the element goes into the refractory state (light gray) and then returns to the rest state [12].

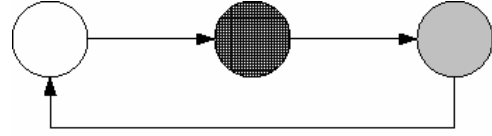


Figure 1. The schematic representation of a cycle of an excitable element.

Excitable media can be found in natural events such as heart palpitation. Moreover, the heart can be approximated as a continuous excitable media [2]. A variety of cardiac tachycardias have been attributed to formation of large scale patterns of excitation such as the formation and break up of spiral waves [2]. Excitable media can be modeled using both partial differential equations and cellular automata, and some examples are the works of Lewis [7], Wiener and Rosenblueth [18], Zykov and Mikhailov [13]. This last model is explained in the next section.

3. The Zykov-Mikhailov Model

The model proposed by Zykov-Mikhailov [13] is the one to be used in the implementation phase. Its simplicity and few number of variables makes it ideal to test the design. Moreover, it allows agent-based implementation simulating cellular automata.

Let us consider a two-dimensional square lattice occupied by excitable entities. Each lattice site is specified by a pair of coordinates i and j ($i, j = 1, 2, \dots, N$), where N is the size of the square. The state of an individual element at a discrete moment of time n is described by two variables ϕ_{ij}^n and s_{ij}^n . The first of these variables, ϕ , represents the integer *phase* of an entity. The zero phase ($\phi = 0$) corresponds to the rest state. The phases in the interval $0 < \phi < \tau_e$ correspond to excited states (τ_e is the duration of the excitable period). The refractory states correspond to the interval $\tau_e < \phi < \tau_e + \tau_r$, where τ_r is usually the duration of the refractory period.

The second variable s specifies the *accumulated energy* received by the element from its neighbors. If this variable happens to exceed a certain threshold h , the entity goes from the state of rest into the first excitable state and its cycle is thus initiated. The accumulated signal of the element (i, j) is obtained by summation of signals coming from its neighbors. Also, to consider temporal accumulation, it is assumed that the signal at the next time step includes a fraction $g < 1$ of the signal. Hence, there is [12]:

$$s_{ij}^{n+1} = g s_{ij}^n + \sum_{k,l} C(i-k, j-l) \theta_{kl}^n \quad (1)$$

where the coefficients C determine the range of spatial summation. The variables θ are defined here as $\theta_{kl}^n = 1$ if the respective element (k, l) is currently in the state of *excitation*, i.e., $0 < \phi_{kl}^n < \tau_e$, and $\theta_{kl}^n = 0$ otherwise.

The updating laws for the phase variable are [12]:

$$\phi_{ij}^{n+1} = \begin{cases} 0 & \text{if } \phi_{ij}^n = 0 \text{ and } s_{ij}^{n+1} < h \\ 1 & \text{if } \phi_{ij}^n = 0 \text{ and } s_{ij}^{n+1} > h \\ \phi_{ij}^n + 1 & \text{if } 0 < \phi_{ij}^n < \tau_e + \tau_r \\ 0 & \text{if } \phi_{ij}^n = \tau_e + \tau_r \end{cases} \quad (2)$$

Both Equation 1 and Equation 2 mean that the element operates like a clock: the cycle is started if the threshold h is exceeded and then lasts for the time $\tau_e + \tau_r$, returning the element to its original state [12].

4. The Zykov-Mikhailov Model Implementation

The implementation of the simulated environment was done using TurtleKit [11], a MadKit plugin for multi-agent simulation [8]. The size of the *world* where the *swarm* is to reside is 300×300 , a property also set using the plugin. From this last sentence, there are two aspects to consider for the implementation: the swarm, implemented using agents, and the world, implemented using a cellular automaton.

4.1. The Swarm Agents

Two kinds of swarm agents were implemented in order to make use of the excitable media model. One kind was in charge of *exciting* the world. From a swarm's point of view, these agents could be considered as the initiators of a process, either because something was found, a new order is meant to be followed, etc. These agents can be placed anywhere in the world (as new events can take place anywhere). For the purposes of the simulation, these agents are fixed and do not move.

The other kind of agents, randomly placed in the simulated world, are the *swarm* agents with the only assigned task of finding the agent that excited the environment. They achieve this by reading two variables embedded in the world, *phase* and *energy* (see Equation 1 and Equation 2). These variables also make the swarm agents behave in different ways. The greater the phase and energy variables the closer the agent is to finding the origin of the excitation. The idea behind this approach is to simulate the levels of excitement in swarms, not from the chemical point of view [1],

but considering that the swarm agents live in a world where excitation takes place and, consequently, behave according to such excitement. In the proposed simulated world, 1500 of these agents were instantiated. Figure 2 presents an initial state of the agents in the world.

4.2. The Simulated Excitable World

For the purposes of the simulation, the world is a square lattice of 300×300 which behaves as a cellular automaton using Equation 1 and Equation 2 as updating rules. There are five exciting swarm agents randomly placed; and 1500 swarm agents that behave according to the level of excitation initiated by an exciting agent. Each exciting agent is *excited* after a fixed number of time steps.

The TurtleKit platform is the simulation engine used to process the rules presented in Equation 2. This engine offers tools for modeling, using, and exploiting *multi-agent* simulations, where agents evolve in a discretized world and act on it according to the environment. Moreover, one characteristic that helped a lot in the simulation process was the fact that, thanks to the plugin, each cell in the world could hold variables (i.e., phase and energy). These variables change at each time step and *guide* each swarm agent's behavior.

In this simulated world one has the ability to excite any of the already mentioned exciting agents and see how such excitation affects the other agents. In order to better visualize the wave propagation, a coloring based on the phase variable, is imposed on each cell of the world. The darker the color, the higher the phase is.

Figure 2 shows the initial state of the world. As it can be seen, the state in all cells is zero, consequently, no excitation has taken place yet.

5. Simulation & Experimentation

In order to test the model, some initial values have to be set (see Equation 1 and Equation 2). The values for these variables were set following Zykov and Mikhailov's model [13], and the examples presented by Mikhailov and Calenbur's book [12]. Thus:

$$\text{Setup Variables} = \begin{cases} h & = 2.4 \\ g & = 0.5 \\ \tau_e & = 3.0 \\ \tau_r & = 5.0 \end{cases} \quad (3)$$

The variable h represents the threshold that has to be reached in order to *be* ready for excitation. The lower this number, the easier to excite the media. At the beginning the world is started with the phase and state variables set to

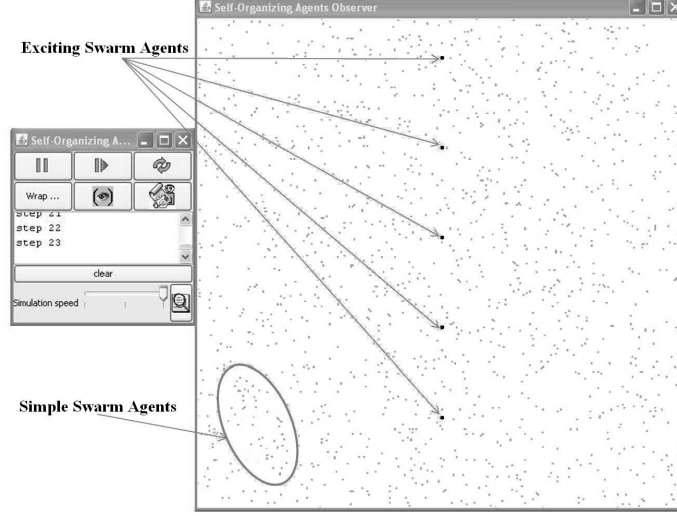


Figure 2. A view of the simulated agents and world.

zero. Once the excitation threshold is reached, excitation takes place and the waves spread throughout the media.

As the excitation takes place, the embedded variables tell agents how to behave. As the simulation goes on, the swarm agents start to move and organize moving towards and according to the excitation origin. Figure 3(a) represents the movement of the swarm towards the origin of the excitation, whereas, Figure 3(b) presents the organization that the swarm forms as the excitation is found.

A better visualization of the organization is seen in Figure 4(a), where one observes the beginning of the organization around a point of excitation. Figure 4(b) shows some clusters already formed after some time steps of the simulation.

The next section of this paper presents the obtained results using this simulation.

6. Obtained Results

In order to measure organization one thing was considered, the existing distance among the swarm agents. This characteristic was chosen as it is considered to be task and environment independent.

The distance, as time goes by, decreases as the swarm agents organize themselves based on the level of excitation. Figure 5 presents the maximum distance among all agents at each time step. This was done by measuring the Euclidean distance d_i between one agent and the others. These results were stored in a vector D_j . Of each vector D_j the maximum distance was calculated and then, the biggest value out of all these was assigned to the current step. In other words,

$$\max_{i=1 \dots T} \{ \max_{j=i \dots T} d_{ij} \}; \quad T = \text{total number of agents} \quad (4)$$

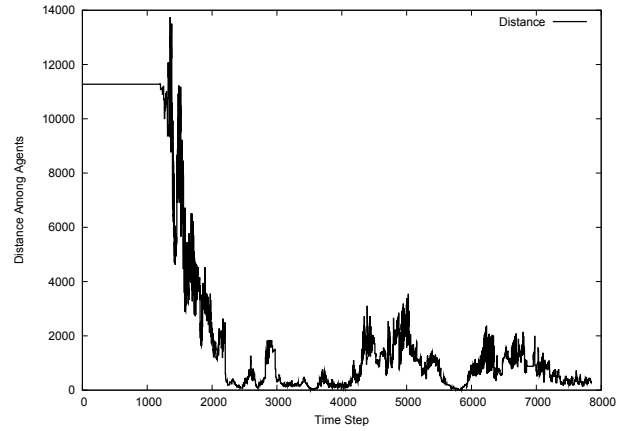


Figure 5. Maximum distance among agents as they try to organize.

Figure 5 shows some peaks. Each peak means an increase in the distance. This was so because of the process explained in the previous section. After a fixed time, the world is reset and the next exciting swarm agent is excited. While the agents react to the new levels of excitation, they tend to separate more. However, as time goes by, this distance is reduced.

Figure 6 is in a way similar to the Figure 5 as it also presents a distance measure. However, in this case what it is being measured is the minimum Euclidean distance among agents. Mathematically this means,

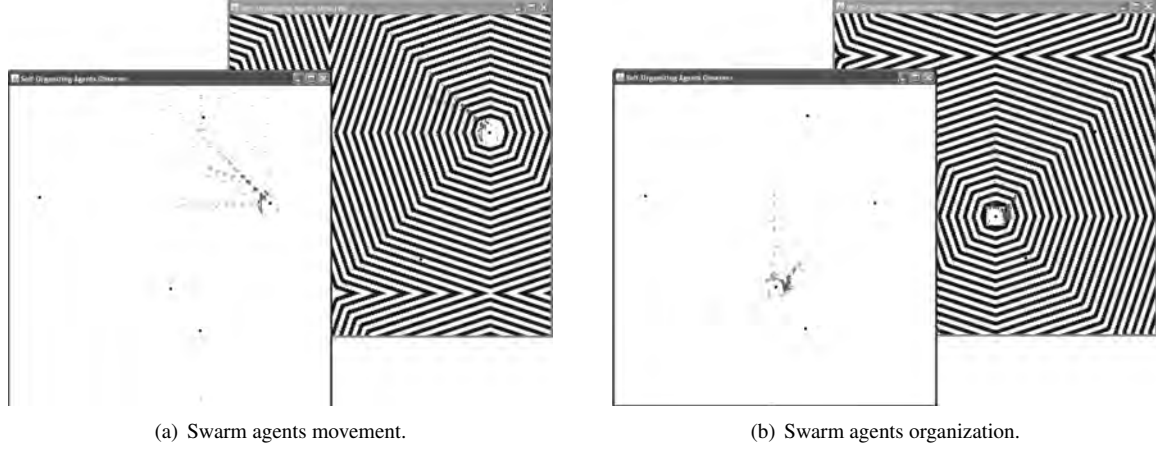


Figure 3. Swarm movement induced by the media.

$$\min_{i=1...T} \{ \min_{j=i...T} d_{ij} \}; T = \text{total number of agents} \quad (5)$$

Following the same idea, Figure 7 shows the averaged Euclidean distance at each step.

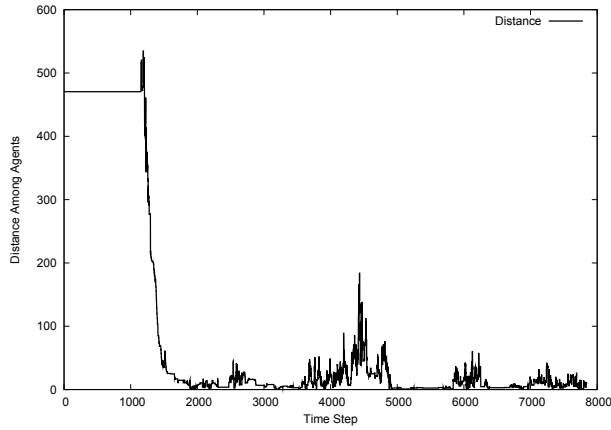


Figure 6. Minimum distance among agents as they try to organize.

The reason why these plots were generated was mainly to see the evolution of the distance as swarm agents configure according to the disturbance in the media. The distance is considered to be a task independent way to measure organization among agents. It is interesting to see that, at the beginning of the simulation, all agents seemed to be *confused*; thus, greater distances among them. However, as the simulation continued, the distance decreases. This could be an indication of better configuration and organization as they find the origin of the disturbance.

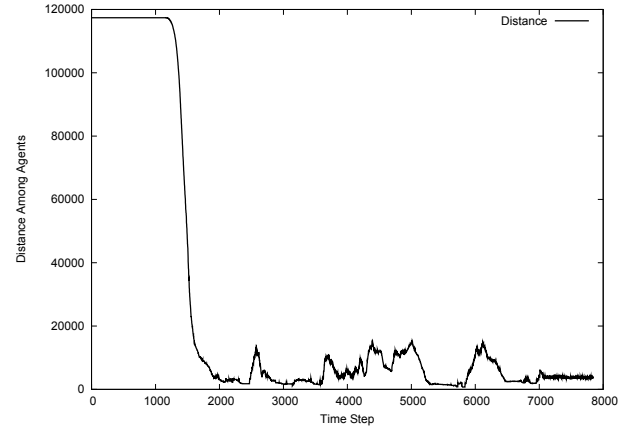


Figure 7. Averaged distance among agents as they try to organize.

The next section of this document presents the conclusions that the experimentation and analysis phase provided.

7. General Conclusions

The main conclusion that can be given is that the combination of excitable media and swarm agents is an interesting and promising way towards autonomic systems. Such combination can guide software components (in this case represented as swarm agents) to achieve self-organization and self-configuration according to the needs of the environment as it is excited (represented here by the excitable media). In other words, and more specifically, the conclusions are:

- Swarm agents can be organized depending on the

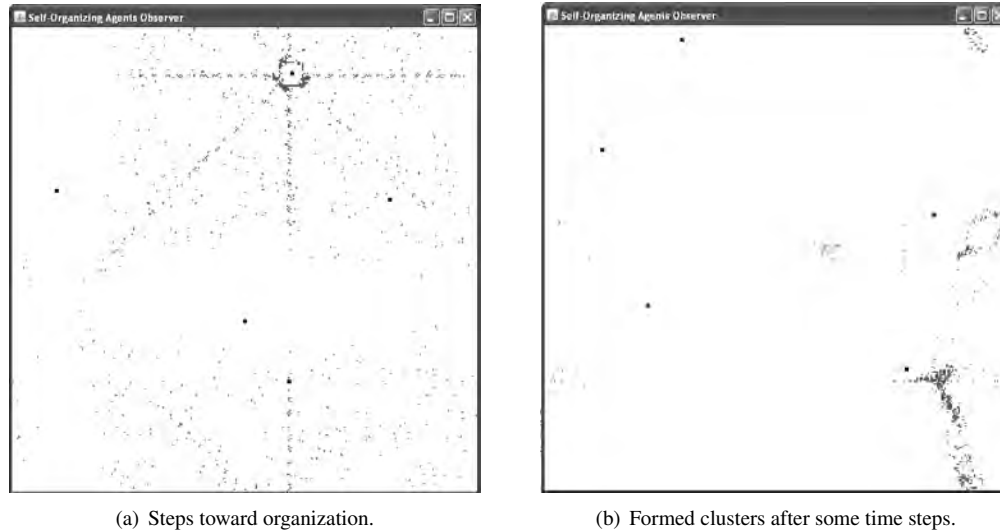


Figure 4. Swarm organization using excitable media.

source of disturbance in an excitable environment.

- Organization of agents is given by two variables, phase ϕ and energy s , embedded in the world.
- There are ways to change the way agents self-configure by tuning the variables of the excitable media model (See Equation 3). Different values produce different kinds of behavior that could meet the requirements of the system.
- By using excitable media, more complex approaches, such as agent negotiation, can be avoided resulting in a simpler way to achieve self- $\star$ properties.

Therefore, swarm agents living in an excitable media is a feasible way towards autonomic systems. These agents can be software components that are required to change their behavior according to the level and to the origin of excitation.

Nevertheless, this is still a simulation and it is necessary to transfer the model to real systems and see the evolution of this combination of methodologies as the system faces real-world situations.

This document presented an approach towards autonomic computing by discussing the possibility of combining swarm intelligence and excitable media to achieve self- $\star$ properties; properties that are necessary in autonomic systems.

8. Acknowledgments

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References

- [1] E. Bonabeau, M. Dorigo, and G. Theraulaz. *Swarm Intelligence: From Natural to Artificial Systems*. Oxford University Press, New York, 1999.
- [2] Center for Nonlinear Dynamics - McGill University. Excitable media. <http://www.cnd.mcgill.ca/bios/bub/excitablomain.html>, 2007. Web Page.
- [3] M. d’Inverno and R. Saunders. Agent-based modelling of stem cell self-organisation in a niche. In S. Brueckner, G. D. M. Serugendo, A. Karageorgos, and R. Nagpal, editors, *Engineering Self-Organising Systems: Methodologies and Applications*, pages 52 – 68. Springer-Verlag, 2005.
- [4] A. E. Eiben. Evolutionary computing and autonomic computing: Shared problems, shared solutions? In O. Babaoglu, M. Jelasity, A. Montresor, C. Fetzer, S. Leonardi, A. van Moorsel, and M. van Steen, editors, *Self-Star Properties in Complex Information Systems*, pages 36 – 48. Springer-Verlag, 2005.
- [5] W. Garrey. The nature of fibrillary contraction of the heart. its relation to tissue mass and form. *American Journal of Physiology*, 23(397), 1914.
- [6] J. Hoffmeyer. Swarm semiotics. <http://www.molbio.ku.dk/MolBioPages/abk/PersonalPages/Jesper/Swarm.html>, 2008. Web Page.
- [7] T. Lewis. *The Mechanism and Graphic Registration of the Heart Beat*. Shaw and Sons, London, third edition, 1925.
- [8] MadKit. The MadKit project. <http://www.madkit.org/>, 2005. Web Page.
- [9] M. Mamei, M. Vasirani, and F. Zambonelli. Self-organising spatial shapes in mobile particles: The TOTA approach. In S. Brueckner, G. D. M. Serugendo, A. Karageorgos, and R. Nagpal, editors, *Engineering Self-Organising Systems: Methodologies and Applications*, pages 138 – 153. Springer-Verlag, 2005.

- [10] A. G. Mayer. Rhythmical pulsation in scyphomeduse. In *Papers from the Tortugas Laboratory*, volume I, pages 115 – 131. 1908.
- [11] F. Michel. Introduction to TurtleKit: A platform for building logo based multi-agent simulations with MadKit. Technical Report RR LIRMM 02215, Laboratoire d'Informatique, de Robotique et de Microélectronique de Montpellier, 2002.
- [12] A. S. Mikhailov and V. Calenbuhr. *From Cells to Societies: Models of Complex Coherent Action*. Springer-Verlag, Germany, 2006.
- [13] A. S. Mikhailov and V. S. Zykov. *Foundations of Synergetics I. Distributed Active Systems*. Springer, Germany, second edition, 1994.
- [14] C. R. Mines. On dynamic equilibrium in the heart. *Physiology*, XLVI(349), 1913.
- [15] R. Murch. *Autonomic Computing*. IBM Press, USA, 2004.
- [16] M. Parashar and S. Hariri. *Autonomic Computing: Concepts, Infrastructure, and Applications*. CRC Press, USA, 2007.
- [17] C. J. Romanes. Preliminary observations on the locomotor system of meduse. *Phil. Trans. Roy. Soc.*, CLXVI(269), 1876.
- [18] N. Wiener and A. Rosenblueth. The mathematical formulation of the problem of conduction of impulses in a network of connected excitable elements, specifically in cardiac muscle. *Arch. Inst. Cardiol.*, 16(205), 1946.
- [19] A. Wuensche. Discrete dynamic lab: Tools for investigating cellular automata and discrete dynamical networks. In A. Adamatzky and M. Komosinski, editors, *Artificial Life Models in Software*, pages 263 – 297. Springer-Verlag, 2005.

Computer Vision and Image Processing

Multichannel Median M-Type L Filter to Remove Impulsive Noise from Color Images

Antonio Toledo-Lopez, Francisco J. Gallegos-Funes, Volodymyr Ponomaryov

National Polytechnic Institute of Mexico

Mechanical and Electrical Engineering Higher School

UPALM Zacatenco, Av. IPN s/n, Edificio Z, Acceso 3, 3er. piso;

Col. Lindavista, 07738, Mexico D.F., Mexico,

Phone/Fax +52-55-57296000x54622, email: fgallegosf@ipn.mx

Abstract

In this paper, we present the Multichannel Median M-type L (MMML) -filter to remove impulsive noise from color images. The proposed technique uses the MM (Median M-Type) -estimator with simplest influence function in the L filtering algorithm and these are adapted to multichannel image processing applications. The results from known techniques are compared with the proposed method to demonstrate its performance in terms of noise suppression, detail preservation, and color retention.

1. Introduction

There are investigated different algorithms applied in the multichannel image processing during the last decade [1,2]. One of the useful and promising approaches being proposed was the multichannel signal processing based on vector processing [1,2,3]. In this case the correlation in chromacity that exists between the channels is employed. The value of each a 2D pixel is represented by a 3D vector, so, the color image is translated into a set of vectors with the directions and lengths that are related to the chromatic properties of the pixels [3,4].

Different filtering techniques have been proposed for color imaging. Particularly, nonlinear filters applied to color images have been designed to preserve edges and details, and remove impulsive noise [5,6].

In this paper, we present the Vector Median M-Type L (VMML) -filter. This filter utilizes the Median M-Type L (MML) algorithm [7,8] whose is adapted to multichannel image processing applications. The proposed filter uses the combined RM-estimator [9] into the L-filter [6] by following way. The redescending M-estimator [10] with the simplest

influence function [11,12] is combined with the R-(median) estimator into the L filtering scheme to obtain sufficient impulsive noise suppression for each channel by using the vector approach. We also introduce an impulsive noise detector [13] to improve the properties of noise suppression and detail preservation of proposed filter.

2. Vector Median M-type L (VMML) Filter

The proposed VMML filter employs the L algorithm [6] and the MM-estimator [9]. The following representation of L filter is often used,

$$\theta_L = \sum_{k=1}^N a_k \cdot X_{(k)} \quad (1)$$

$$a_k = \int_{k-1/N}^{k/N} h(\lambda) d\lambda / \int h(\lambda) d\lambda \quad (2)$$

where $X_{(k)}$ is the ordered data sample, a_k are the weighted coefficients, and $h(\lambda)$ is a probability density function. The weighted coefficients a_k can be computed by using the exponential, Laplacian, or uniform distribution functions [6].

For convenience the VL (Vector L) filter is written below as

$$\theta_L = \sum_{m=1}^N a_m \cdot \psi(y_m) y_m \quad (3)$$

where y_m are the noisy image vectors in sliding filter window, which includes $m=1, \dots, N$ (N is odd)

vectors $y_1, y_2, \dots, y_N$ located at spatial coordinates in

the filter window, and $\psi(y_m) = \begin{cases} 1 & m \leq (2L+1)^2 \\ 0 & \text{otherwise} \end{cases}$ is

the influence function.

The robustness of the VL filter can be improved with the use of the Median M-type estimator [7-9],

$$\theta_{\text{MM}} = \text{MED}\{Y_k \varphi(Y_k - \text{MED}\{Y_N\}), k=1, \dots, N\} \quad (4)$$

where Y_k is data sample, φ is the normalized influence function $\psi: \psi(Y) = Y \varphi(Y)$, and Y_N is the primary data sample.

So, the Vector Median M-type L (VMML) -filter can be written as:

$$\theta_{\text{VMML}} = \text{MED}\{a_m \cdot [Y_m \psi(Y_m - \text{MED}\{Y_N\})]\} / a_{\text{MED}} \quad (5)$$

where $Y_m \psi(Y_m - \text{MED}\{Y_N\})$ is set of values of vectors y_m which are weighted by value in accordance with the used influence function $\tilde{\psi}(y_m)$ in a sliding filter window, y_m are the noisy image vectors in a sliding filter window, which includes vectors $y_1, y_2, \dots, y_N$ located at spatial coordinates (i, j) in the filter window, a_m are the weighted coefficients used into the proposed filtering scheme, and a_{MED} is the median of coefficients a_m used as a scale constant.

The simplest influence function is used in the proposed VMML filter [7-9, 11-12],

$$\psi_{\text{cut}(r)}(y_m) = y_m \cdot 1_{[-r, r]}(y_m) = \begin{cases} y_m, & |y_m| \leq r \\ 0, & \text{otherwise} \end{cases} \quad (6)$$

where the parameter r is connected with restrictions on the range of influence function.

Finally, we propose for enhancement of the removal ability of VMML filter to involve an impulsive noise detector, this detector chooses that pixel is or not filtered. The impulsive detector used here is defined as [13]:

$$[(\text{rank}(y_c) \leq T_1) \vee (\text{rank}(y_c) \geq N - T_1)] \wedge (|y_c - \text{MED}(Y_N)| \geq T_2) \quad (7)$$

where y_c is the vector of interest (the central vector in the filtering window), $T_1 > 0$ and $T_2 \geq 0$ are thresholds, and N is the length of the data.

3. Experimental Results

The criteria used to compare the restoration performance of various filters were the *peak signal-to-noise ratio* (PSNR) and *normalized mean square error* (NMSE) for the evaluation of noise suppression, the *mean absolute error* (MAE) for quantification of edges and detail preservation, and the *normalized color difference* (NCD) for the quantification of the color perceptual error [1-6]:

$$\text{PSNR} = 10 \cdot \log \left[\frac{(255)^2}{\text{MSE}} \right], \text{dB} \quad (8)$$

$$\text{NMSE} = \frac{\sum_{i=1}^{M_1} \sum_{j=1}^{M_2} \|y(i, j) - y_0(i, j)\|_{L_2}^2}{\sum_{i=1}^{M_1} \sum_{j=1}^{M_2} \|y_0(i, j)\|_{L_2}^2} \quad (9)$$

$$\text{MAE} = \frac{1}{M_1 M_2} \sum_{i=1}^{M_1} \sum_{j=1}^{M_2} \|y(i, j) - y_0(i, j)\|_{L_1} \quad (10)$$

where $\text{MSE} = \frac{1}{M_1 M_2} \sum_{i=1}^{M_1} \sum_{j=1}^{M_2} \|y(i, j) - y_0(i, j)\|_{L_2}^2$ is the *mean square error*, M_1, M_2 are the image dimensions, $y(i, j)$ is the 3D vector value of the pixel (i, j) of the filtered image, $y_0(i, j)$ is the corresponding pixel in the original uncorrupted image, and $\|\cdot\|_{L_1}, \|\cdot\|_{L_2}$ are the L_1 - and L_2 -vector norms, respectively.

$$\text{NCD} = \frac{\sum_{i=1}^{M_1} \sum_{j=1}^{M_2} \|\Delta E_{Luv}(i, j)\|_{L_2}}{\sum_{i=1}^{M_1} \sum_{j=1}^{M_2} \|E_{Luv}^*(i, j)\|_{L_2}} \quad (11)$$

where $\|\Delta E_{Luv}(i, j)\|_{L_2} = \left[(\Delta L^*)^2 + (\Delta u^*)^2 + (\Delta v^*)^2 \right]^{1/2}$ is the norm of color error; ΔL^* , Δu^* , and Δv^* are the difference in the L^* , u^* , and v^* components, respectively, between the two color vectors that present the filtered and uncorrupted original images for each pixel (i, j) , and $\|E_{Luv}^*(i, j)\|_{L_2} = \left[(L^*)^2 + (u^*)^2 + (v^*)^2 \right]^{1/2}$ is the norm or magnitude of the uncorrupted original image pixel vector in the $L^* u^* v^*$ space.

The described VMML filter has been evaluated, and its performance has been compared with *Vector Median* (VM) [14], *Adaptive VM* (AVM) [15], *Fast Adaptive Similarity VM* (FASVM) [16], and *Adaptive Multichannel Non parametric Vector Median M-type K-Nearest Neighbor* (AMNVMMKNN) [17] filters.

The 320x320 RGB color images “Peppers” and “Mandrill” were corrupted by impulsive noise

intensities that change in the range from 0% (noise free) to 20% with the step size of 5% for spike occurrence in each a channel. Table 1 shows the performance criteria in the case of image “Peppers”.

From Table 1, we observe that the proposed VMML filter provides better performance results in comparison with other filters in the most of cases. In some cases the VMML filter lost performance in terms of PSNR and NMSE criteria, but the MAE and NCD performances have the best results. This is observed in the visual results of Figure 1 where the AMNVMMKNN filtered image presents the best noise suppression but the proposed VMML filtered image shows the better detail preservation and color perceptual error.

Table 2 presents the performance results for test image “Mandrill” and Figure 2 exhibits the processed images in the case of noise probability occurrence of 10% for each color channel according with Table 2. From the results of Table 2, one can see that the performance of proposed filter is better in comparison with the performance of filters proposed as comparative when the noise level is high, but when the noise level is low the FASVM filter has better performance in terms of noise suppression. Figure 2

shows that the proposed VMML filter has a better subjective quality in comparison with AMNVMMKNN and FASVM filters.

Figure 3 shows the subjective visual quantities of restored zoom part of 320x320 color image “Lena” with spike occurrence of 20%. From this Figure we observe that the proposed VMML with and without impulsive noise detector appears to have better visual qualities in comparison with the reference filters.

From Tables 1-2 and Figures 1-3, one can see that the proposed VMML filter consistently outperforms other filters by balancing the tradeoff between noise suppression, detail preservation, and color retention.

To demonstrate the performance of the proposed filtering scheme we applied it for filtering of UHF SAR images, which naturally have speckle noise. The filtering results are presented in Figure 4 for the image “Manzano” (forest near Manzano State Park, New Mexico) 2 meter resolution, UHF SAR, 380MHz, vertical polarization. It is possible to see analyzing the filtering images that speckle noise can be efficiently suppressed, while the sharpness and fine feature are preserved using the proposed filter.

Table 1. Comparative restoration results for different impulsive noise percentages for color image “Peppers”

% Noise	Algorithm	PSNR	MAE	NCD	NMSE
0	AVM	31.54	1.51	0.0057	0.0025
	FASVM	34.07	0.34	0.0016	0.0014
	AMNVMMKNN	27.29	5.56	0.0217	0.0068
	VM	30.89	2.86	0.0108	0.0030
	VMML	36.59	0.44	0.0010	0.0014
5	AVM	30.81	1.97	0.0080	0.0031
	FASVM	31.19	1.14	0.0045	0.0028
	AMNVMMKNN	29.21	4.42	0.0170	0.0046
	VM	30.30	3.14	0.0120	0.0034
	VMML	32.03	0.89	0.0016	0.0034
10	AVM	29.79	2.49	0.0095	0.0039
	FASVM	29.01	2.07	0.0081	0.0046
	AMNVMMKNN	28.71	4.66	0.0182	0.0049
	VM	29.44	3.49	0.0132	0.0042
	VMML	30.03	1.33	0.0021	0.0050
15	AVM	28.66	3.13	0.0119	0.0050
	FASVM	25.63	3.70	0.0142	0.0101
	AMNVMMKNN	28.32	4.92	0.0193	0.0054
	VM	28.44	3.95	0.0151	0.0053
	VMML	28.04	1.87	0.0026	0.0079
20	AVM	27.30	3.92	0.0150	0.0069
	FASVM	24.45	4.84	0.0161	0.0131
	AMNVMMKNN	27.82	5.26	0.0209	0.0061
	VM	27.19	4.53	0.0172	0.0071
	VMML	26.30	2.50	0.0031	0.0117

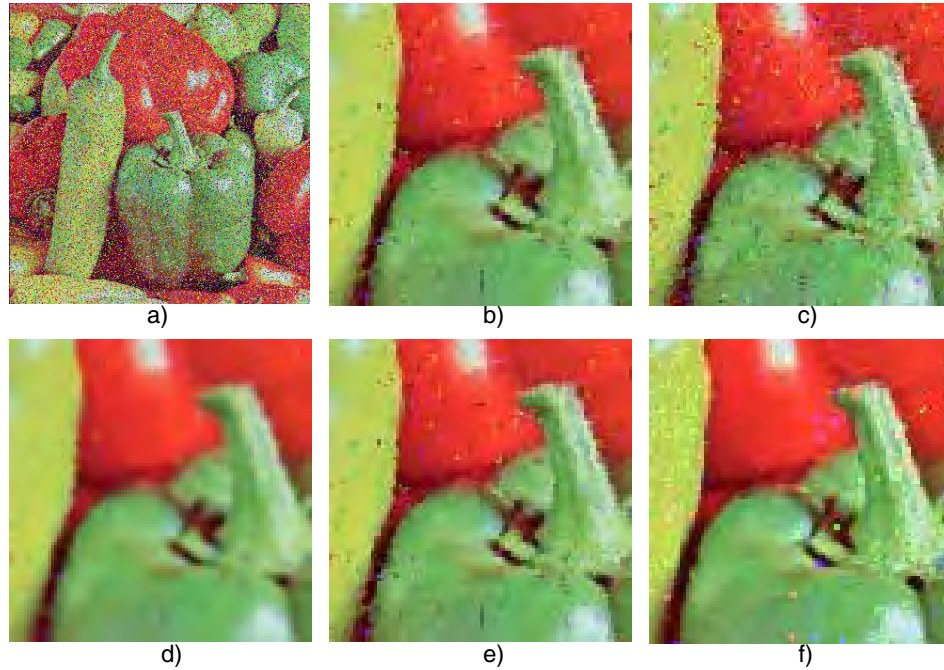


Figure 1. Subjective visual quantities of restored zoom part of color image “Peppers”, a) Input noisy image corrupted by 30% impulsive noise in each a channel; b) AVM filtered image; c) FASVM filtered image; d) AMNVMMKNN filtered image, e) VM filtering image, and f) VMML filtering image.

Table 2. Comparative restoration results for different impulsive noise percentages for color image “Mandrill”

% Noise	Algorithm	PSNR	MAE	NCD	NMSE
0	AVM	24.44	6.98	0.0280	0.0171
	FASVM	29.97	0.99	0.0041	0.0047
	AMNVMMKNN	21.99	13.60	0.0532	0.0300
	VM	24.15	8.55	0.0336	0.0182
	VMML	27.39	3.60	0.0025	0.0093
5	AVM	24.27	7.36	0.0293	0.0177
	FASVM	27.21	2.54	0.0100	0.0090
	AMNVMMKNN	23.62	10.76	0.0350	0.0210
	VM	24.02	8.71	0.0340	0.0188
	VMML	26.66	3.16	0.0028	0.0124
10	AVM	23.97	7.87	0.0310	0.0190
	FASVM	25.29	4.06	0.0159	0.0140
	AMNVMMKNN	23.48	11.04	0.0432	0.0213
	VM	23.78	8.96	0.0349	0.0199
	VMML	25.90	4.03	0.0032	0.0138
15	AVM	23.48	8.60	0.0334	0.0213
	FASVM	23.68	5.83	0.0223	0.0203
	AMNVMMKNN	23.28	11.38	0.0443	0.0223
	VM	23.35	9.43	0.0363	0.0219
	VMML	24.33	5.44	0.0035	0.0180
20	AVM	22.88	9.49	0.0362	0.0244
	FASVM	22.47	7.69	0.0354	0.0268
	AMNVMMKNN	23.07	11.77	0.0455	0.0233
	VM	22.79	10.11	0.0384	0.0249
	VMML	23.48	6.26	0.0038	0.0217

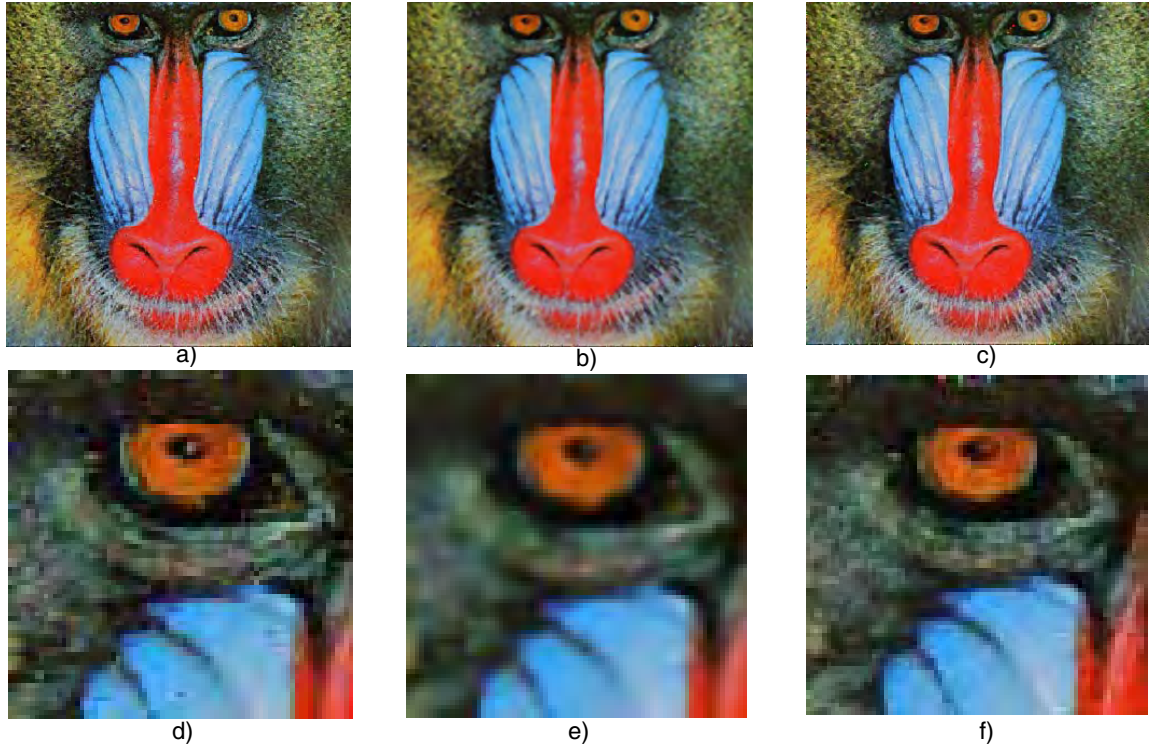


Figure 2. Subjective visual quantities of restored color image "Mandrill" in the case of 10% of impulsive noise, a) FASVM filtered image, b) FASVM filtered zoom part of (a), c) AMNVMMKNN filtered image, d) AMNVMMKNN filtered zoom part of (c), e) Proposed VMML filtered image, and f) Proposed VMML filtered zoom part of (e).

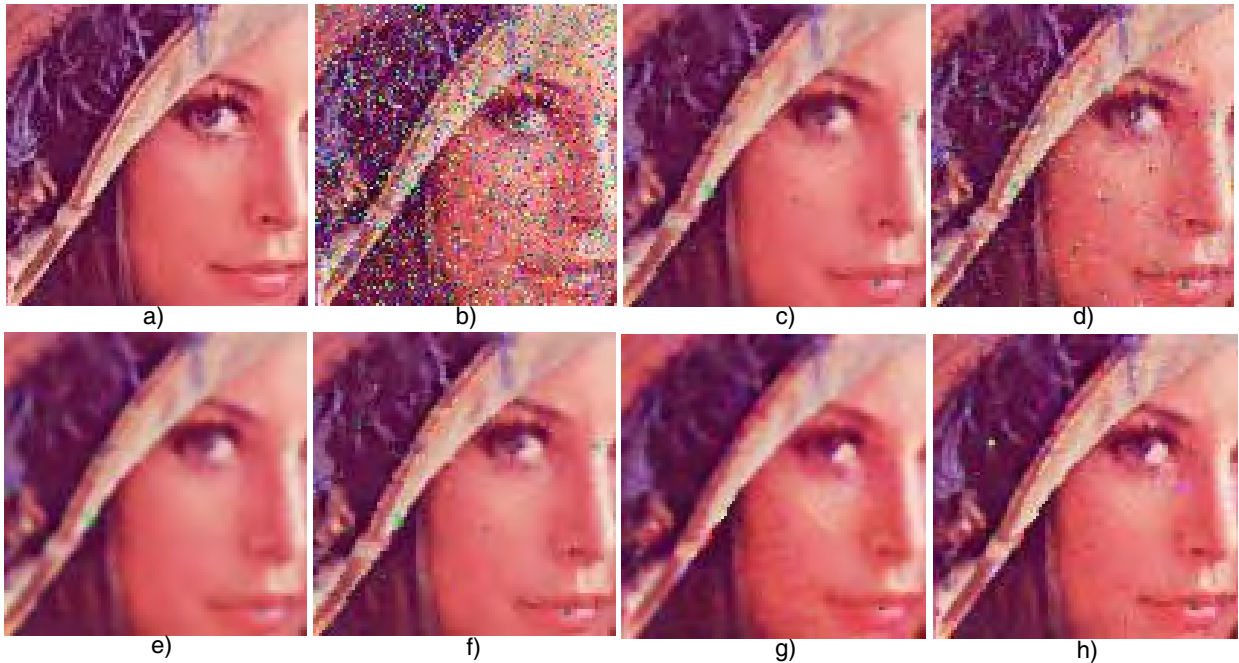


Figure 3. Subjective visual quantities of restored zoom part of color image "Lena", a) Original image, b) Input noisy image corrupted by 20% impulsive noise in each a channel; c) AVM filtered image; d) FASVM filtered image; e) AMNVMMKNN filtered image, f) VM filtered image, g) Proposed VMML filtered image (without impulsive noise detector), and h) Proposed VMML filtered image (with impulsive noise detector).

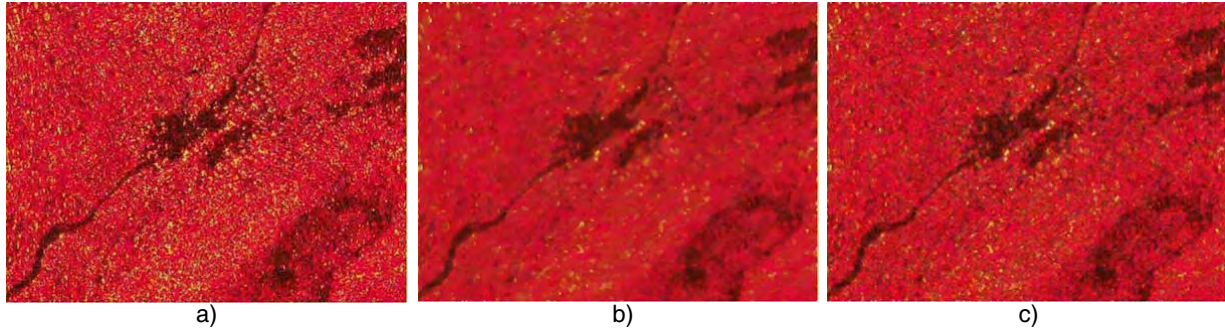


Figure 4. Visual results of despeckled SAR image. a) Original image “Manzano”, resolution 2m, source Sandia National Lab., b) Despeckled image with the proposed VMML filter (without impulsive noise detector), c) Despeckled image with the proposed VMML filter with impulsive noise detector).

4. Conclusions

The proposed VMML filter is able to remove impulsive noise and preserve the edges and details in color imaging. The proposed filter uses the robust MM-estimator with the simplest influence function and utilizes an impulsive noise detector to provide better noise suppression, detail preservation, and color retention. The VMML filter has demonstrated better quality of image processing, both in visual and analytical sense in comparison with different known color image processing algorithms.

5. Acknowledgments

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6. References

- [1] Bovik, A., *Handbook of Image and Video Processing*, Academic Press, San Diego, 2000.
- [2] Plataniotis, K.N., A.N. Venetsanopoulos, *Color Image Processing and Applications*, Springer Verlag, Berlin, 2000.
- [3] K.N. Plataniotis, D. Androutsos, S. Vinayagamoorthy, A.N. Venetsanopoulos, “Color image processing using adaptive multichannel filters”, *IEEE Trans. Image Process.*, vol. 6-7, 1997, pp. 933-949.
- [4] P.E. Trahanias, D.G. Karakos, A.N. Venetsanopoulos, “Directional processing of color images: Theory and experimental results”, *IEEE Trans. Image Process.*, vol. 5, 1996, pp. 868-880.
- [5] Astola J., P. Kuosmanen, *Fundamentals of Nonlinear Digital Filtering*, CRC Press, Boca Raton-New York, 1997.
- [6] Pitas I., A.N. Venetsanopoulos, *Nonlinear Digital Filters*, Kluwer Academic Publishers, Boston, 1990.
- [7] J.L. Varela-Benitez, F.J. Gallegos-Funes, V. Ponomaryov, “RM L-filters for Real Time Imaging”, *IEEE 15th International Conference on Computing*, IEEE Press, Mexico City, Mexico, 2006, pp. 43-48.
- [8] J.L. Varela-Benitez, F.J. Gallegos-Funes, V.I. Ponomaryov, “Real-time speckle and impulsive noise suppression in 3-D imaging based on robust linear combinations of order statistics”, *SPIE 6496 Real-Time Image Processing 2007*, SPIE Press, San Jose, USA, 2007, pp. 64960H.
- [9] F.J. Gallegos-Funes, V.I. Ponomaryov, “Real-time image filtering scheme based on robust estimators in presence of impulsive noise”, *Real Time Imaging*, vol. 10, no. 2, 2004, pp. 69-80.
- [10] M.E. Zervakis, A.N. Venetsanopoulos, “M-estimators in robust nonlinear image restoration”, *Optical Engineering*, vol. 29, no. 5, 1990, pp. 455-470.
- [11] Hampel, F.R., E.M. Ronchetti, P.J. Rouseew, W.A. Stahel, *Robust Statistics: The approach based on influence function*, Wiley, New York, 1986.
- [12] S. Peltonen, P. Kuosmanen, J. Astola, “Output distributional influence function”, *Proc. IEEE EURASIP Workshop on Nonlinear Signal and Image Processing*, Antalya, Turkey, 1999, pp. 33-37.
- [13] I. Aizenberg, J. Astola, T. Bregin, C. Butakoff, K. Egiazarian, D. Paily, “Detectors of the impulsive noise and new effective filters for the impulsive noise reduction”, *SPIE 5014 Image Processing, Algorithms and Systems II*, SPIE Press., 2003, pp. 419-428.
- [14] J. Astola, P. Haavisto, Y. Neuvo, “Vector median filters”, *Proc.IEEE.*, vol. 78, 1990, pp. 678-690.
- [15] R. Lukac, “Adaptive vector median filtering”, *Pattern Recognition Letters*, vol. 24, 2003, pp. 1889-1899.
- [16] B. Smolka, R. Lukac, A. Chydzinski, K.N. Plataniotis, W. Wojciechowski, “Fast adaptive similarity based impulsive noise reduction filter”, *Real-Time Imaging*, vol. 9, no. 4, 2003, pp. 261-276.
- [17] V.I. Ponomaryov, F.J. Gallegos-Funes, A. Rosales-Silva, “Real-Time Color Imaging Based on RM-Filters for Impulsive Noise Reduction”, *J. Imaging Science and Technology*, vol. 49, no. 3, 2005, pp. 205-219.

A Comparison of the Bandelet, Wavelet and Contourlet Transforms for Image Denoising

Osslan Osiris Vergara Villegas¹, Humberto de Jesús Ochoa Domínguez¹ and Vianey Guadalupe Cruz Sánchez²

¹Universidad Autónoma de Ciudad Juárez (UACJ)

Avenida del Charro No. 450 Norte Ciudad Juárez, Chihuahua, México

²Centro Nacional de Investigación y Desarrollo Tecnológico (cenidet)

Interior Internado Palmira S/N. Col. Palmira Cuernavaca, Morelos, México
{overgara, hochoa}@uacj.mx, vianey@cenidet.edu.mx

Abstract

The bandelet transform take advantage of the geometrical regularity of the structure of an image and is appropriate for the analysis of edges and textures of the images. Denoising is one of the most interesting and widely investigated topics in image processing area. The main problem in denosing is the tradeoff between the noise suppression and oversmoothing of image details. In order to solve that problem, in this paper we exploit the geometrical advantages offered by the bandelet transform to solve the problem of image denoising. We present the results obtained with the bandelet transform for denoising process with additive white Gaussian noise and salt and pepper noise. A comparison is made with those results obtained with wavelets and contourlets. We show that bandelets can outperform the wavelets and contourlets in terms of subjective and objective measures.

1. Introduction

A very important stage of any computer vision system is the image preprocessing. This step is performed in order to obtain better image quality, than the obtained at the acquisition stage.

In computer vision, we need to deal with a lot of data sets. Many scientific data sets are contaminated with noise, either because of the data acquisition process or because of naturally occurring phenomena [1]. Image denoising is the process of separating the noise out of the image, from a single observation of a degraded image.

Denoising is currently one of the most interesting and widely topics, investigated in image processing from both, a theoretical and an empirical point of view

[2]. A common way to overcome the denoising difficulties is to remove the noise by using some techniques, to compute an approximation of the exact data that belongs to the proper function space [3].

The goal of image restoration is to relieve human observers from this task, by reconstructing a plausible estimate of the original image from the distorted or noisy observation. The problem of image denoising can be stated by equation 1 [2]:

$$g = f + n \quad (1)$$

Where g is the noisy signal, f is the clean signal and n the noise. The performance of the denoising method mainly depends on a suitable representation, able to compact as much as possible, the original information of the image in a few coefficients.

Denoising involves partitions of the data into the desirable components (signal) and undesirable components (noise). Separating the data into two parts requires milder assumptions than a complete analysis of all sources present. Validation is easier than those for more general techniques, and the tools require less expertise and pose less risk of misuse by inexperienced practitioners [3].

The classical techniques on image denoising are based on filtering [3]. However, new techniques based on the wavelet transforms [4], principal components analysis (PCA) [5] and independent component analysis (ICA) [6] have recently appear.

There are a lot of proposed algorithms that use wavelet transform to solve the image denoising problem [1], [2], [4]. Nevertheless, the wavelet denoising methods present some disadvantages when applied to natural images. To overcome this problem, new methods based on the contourlet transform have been proposed [7]. However, the image denoising

problem is still a great challenge. In this paper, we present the application of the bandelet transform to offer a novel solution to the image denoising problem. In the next sections we present the theory for denoising, the results obtained with bandelets and comparisons with contourlets and wavelets.

2. The Bandelet Transform

When natural images are processed, the geometrical structures inside become very important. Natural images are made of these structures which mostly are irregular or diffuse edges, high frequencies and several features that need very complicated methods to process. Therefore, it is important to know the geometrical structures of the images in order to exploit them. The geometry defines changing zones and gives important tracks for human perception.

There are several transforms that tackle the problem of image geometry such as the contourlet or bandelet transform. The second generation bandelet transform is a 2-D wavelet transform followed by a bandeletization. The bandelet is an orthogonal, multiscale transform able to preserve the geometric content of images and surfaces [8].

The orthogonal bandelets use an adaptive segmentation and a local geometric flow, which is well suited to capture the anisotropic regularity of edge structures. They are constructed with a “bandeletization” which is a local orthogonal transformation applied to wavelet coefficients [8].

The bandeletization removes the geometric redundancy of an image using an operator family. The operators can adapt the coefficients of a wavelet transform to the image geometry in order to capture the singularities of the image edges [9].

The bandelet transform use the image dyadic partition scheme, proposed for the x-lets. However, in each square, the geometry is not defined by localization; instead, it is defined by the orientation [10].

The orientation is defined by a vector field called geometric flow. The orientation is parallel to the discontinuities in every image point, and this generates a band. Each band is straightened into horizontal or vertical direction to put the geometric flow parallel to some shaft. The straighten operation is called warping.

The bandelet transform exploit the anisotropic regularity of natural images by constructing orthogonal vectors that are elongated in the direction where the function has a maximum of regularity [8].

3. Image Denoising with Bandelets

The simplest technique for image denoising is the thresholding method. In this method, we have as an input a noisy signal, and we perform a soft or hard thresholding scheme. The hard technique, sets to zero coefficients with values less than or equal to the threshold T . The soft technique, performs equal to the hard technique, but subtracts T from any coefficient that is greater than the threshold [11].

In the bandelet transform, a geometric flow of vectors is defined to represent the edges of an image. These vectors give the local directions in which the image has regular variations. Orthogonal bandelet bases are constructed by dividing the image support in regions which the geometric flow is parallel. In the bandelet basis $B(I) = \{b_v\}_v$, a thresholding algorithm defines an estimator $\tilde{f}$ that can be computed with equation 2.

$$\tilde{f} = \sum_v S_T(\langle Y, b_v \rangle) b_v \quad (2)$$

Where Y is the noisy image, and the threshold is set to $T = \lambda\sigma$. Where σ is the noise variance, $\lambda = \sqrt{2\log_e(N)}$, and N is the number of pixels [8]. The function for thresholding is defined by equation 3.

$$S_T(x) = \begin{cases} x & \text{if } |x| > T \\ 0 & \text{otherwise} \end{cases} \quad (3)$$

The steps to compute the image denoising are:

1. Select an input square image.
2. Define the number of image partition squares (we select a value of 8 and we define none overlapping).
3. Compute the number of geometries in the image.
4. Perform a wavelet transform.
5. Compute the Lagrangian function (as a result we obtain the image geometries).
6. Compute the bandelet transform; we use a fixed segmentation tree.
7. Select the threshold (for this paper we select at least 3 noise levels).
8. Perform soft thresholding in the bandelet subbands.
9. Compute the inverse bandelet transform.

4. Tests and Results

In order to make the tests we use a set of 256 x 256 images with different frequency content. We classify the images in three groups: low, medium and high

frequency groups. The images were classified by measuring their frequency content. In order to define this classification we transform the images into the wavelet domain using Daubechies db9 filter, and then we compute the energy of the wavelet coefficients in each subband using equation 4.

$$E_1 = \frac{1}{N^2} \sum_{k=1}^N |C_k|^2 \quad (4)$$

Where N is the image size and C_k is a wavelet domain coefficient. This classification measures the performance of images with different frequency content. The results of the classification is shown in table 1. The ranges were determined using [12] and are: medium frequency greater or equal than 95%, medium frequency from 99 to 99.49% and low frequency lower to 99%. The selected images are shown in figure 1.

The first step is to add noise to the images using the equation 1. Then, we compute the standard deviation of the complete input image (f). The value obtained is divided by a parameter called ρ which can take one values out of three. We use two different types of noise: Additive White Gaussian Noise (AWGN) and the Salt and Pepper Noise (SAPN) which are used to test denoising applications.

We compute the image threshold for each image in the wavelet, contourlet and bandelet domains. In figure 2, we show the recovered images after denoising with AWGN and $\rho = 0.01$, the numerical results of the Peak signal to Noise Ratios (PSNRs) and the Signal to Noise Ratios (SNRs) are shown in table 2. The recovered images obtained with denoising from SAPN are shown in figure 3 and the numerical results in table 3.

Table 1. Numerical results of Image classification.

Classification	Image	Wavelet Subband			
		LL	LH	HL	HH
Low frequency	Apple	99.9697	0.0243	0.0041	0.0018
	Peppers	99.7610	0.1054	0.1151	0.0184
	Clown	99.7500	0.0775	0.1569	0.0157
	Goldhill	99.6635	0.1528	0.1587	0.0250
	Tomography	99.5456	0.2027	0.2311	0.0206
	Lena	99.5420	0.1080	0.2911	0.0588
Medium frequency	Barbara	99.3720	0.1685	0.3807	0.0789
	Lighthouse	99.3386	0.2590	0.3815	0.0209
	Star	99.2432	0.2637	0.2663	0.2267
	Oil filter	99.1432	0.3051	0.5021	0.0496
	Buthfish	99.0839	0.6491	0.8810	0.3859
	House	99.0748	0.3403	0.5597	0.0251
High frequency	Camman	98.9378	0.6033	0.3897	0.0692
	Baboon	98.9158	0.6017	0.2888	0.1937
	Satellite	98.7691	0.7717	0.4137	0.0454
	Nift7	98.3234	0.8943	0.6764	0.1058
	Letters	98.1810	0.5890	0.8155	0.4145
	Zone	80.2468	7.6932	7.6932	4.3669

Table 2. Numerical results obtained after image denoising with AWGN.

$\sigma = 0.01$	Noisy		Wavelet		Contourlet		Bandelet	
Image	PSNR	SNR	PSNR	SNR	PSNR	SNR	PSNR	SNR
Apple	75.63	10.21	76.65	11.22	82.60	17.18	75.45	10.03
Peppers	74.17	12.51	72.69	11.03	73.80	12.14	73.19	11.53
Clown	73.59	13.34	72.28	12.03	73.94	13.69	72.00	11.75
Goldhill	74.60	13.13	72.31	10.84	73.53	12.06	73.02	11.55
Tomography	72.60	11.96	72.24	11.61	75.04	14.41	71.80	11.17
Lena	74.23	12.31	73.26	11.33	74.92	12.99	73.09	11.16
Barbara	75.00	12.35	71.77	9.13	71.98	9.34	73.97	11.33
Lighthouse	74.14	11.61	73.07	10.54	75.03	12.50	73.60	11.07
Star	71.35	9.757	70.45	8.85	70.51	8.92	71.22	9.62
Oil filter	72.55	14.38	70.69	12.52	71.54	13.37	70.79	12.61
Buthfish	73.24	13.79	69.76	10.31	70.05	10.59	71.76	12.31
House	73.68	12.56	71.32	10.20	72.07	10.95	72.76	11.64
Camman	72.98	12.63	72.15	11.80	73.50	13.15	71.98	11.63
Baboon	75.95	11.61	72.12	7.78	72.12	7.78	74.94	10.60
Satellite	74.18	8.039	72.57	6.43	73.97	7.82	73.61	7.474
Nift7	72.53	13.65	69.40	10.51	69.46	10.57	71.18	12.29
Letters	71.69	12.38	70.35	11.03	68.35	9.03	71.69	12.37
Zone	72.32	15.15	66.82	9.66	66.71	9.55	71.11	13.95

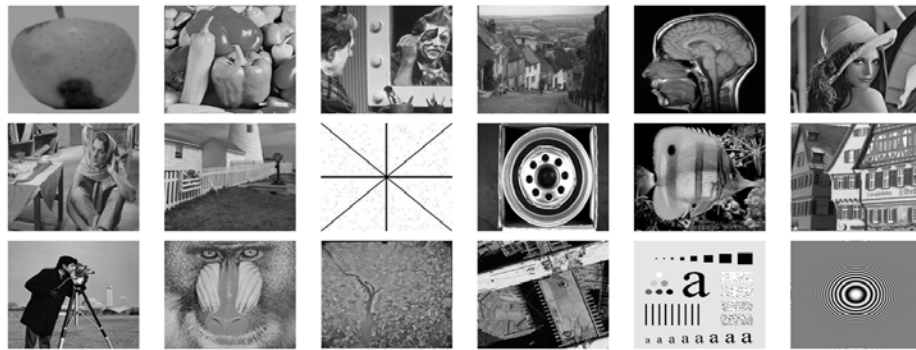


Fig. 1. Test images. First row: Apple, Peppers, Clown, Goldhill, Tomography, Lena. Second row: Barbara, Lighthouse, Star, Oil filter, Buthfish, House. Third row: Camman, Baboon, Satellite, Nift7, Letters, Zone.



Fig. 2. AWGN Denoised images. First column: Noisy images with AWGN and $\sigma = 0.01$. Second column: Denoised images using wavelets. Third column: Denoised images using contourlets. Fourth column: Denoised images using bandelets.



Fig. 3. SAPN denoised images. First column: Noisy images with AWGN and $\sigma = 0.01$. Second column: Denoised images using wavelets. Third column: Denoised images using contourlets. Fourth column: Denoised images using bandelets.

Table 3. Numerical results obtained after image denoising with SAPN.

$\sigma = 0.01$		Noisy		Wavelet		Contourlet		Bandelet	
Image		PSNR	SNR	PSNR	SNR	PSNR	SNR	PSNR	SNR
Apple		88.17	22.75	85.63	20.20	84.88	19.46	88.17	22.74
Peppers		88.13	26.47	75.15	13.49	74.10	12.44	79.79	18.13
Clown		88.09	27.84	74.82	14.57	74.23	13.98	76.88	16.64
Goldhill		88.11	26.64	74.14	12.67	73.76	12.29	77.49	16.02
Tomography		88.16	27.52	76.33	15.69	75.60	14.97	79.36	18.72
Lena		88.12	26.20	76.22	14.30	75.31	13.38	79.27	17.34
Barbara		88.14	25.50	73.24	10.59	72.17	9.52	80.33	17.69
Lighthouse		88.15	25.62	76.02	13.49	75.60	13.07	82.51	19.97
Star		88.12	26.52	75.22	13.62	71.11	9.52	85.69	24.10
Oil filter		88.13	29.96	72.41	14.24	71.84	13.67	75.24	17.07
Butthfish		88.09	28.64	71.02	11.57	70.18	10.73	76.78	17.33
House		88.12	27.00	73.14	12.02	72.40	11.28	79.38	18.26
Camman		88.11	27.76	74.92	14.57	73.79	13.44	78.56	18.21
Baboon		88.14	23.79	73.50	9.16	72.36	8.01	80.99	16.65
Satellite		88.13	21.99	76.05	9.91	75.01	8.87	82.08	15.94
Nift7		88.19	29.30	70.69	11.81	69.70	10.81	76.47	17.59
Letters		88.14	28.83	73.77	14.46	68.71	9.40	89.46	30.15
Zone		88.14	30.97	67.58	10.42	66.99	9.82	76.64	19.48

4.1. Discussion

We measure the PSNR and the SNR because they are typical measures widely used for denoising. All the transforms performs better when AWGN is present, this can be probed with the numerical results.

The performance of the denoising algorithm using bandelet performs well even when we have an image with very high frequencies. It is important preserve the details in order to avoid oversmoothing them. The higher the noise variance the worse the performance of the algorithm however, the edges of the images are preserved.

The objective measures are not always a good parameter to measure the quality of the denoised image. Therefore, we perform subjective tests asking some people their opinion of each recovered image. The images with best qualification were always those processed with the bandelet transform.

In figure 4 we show the bandelet SNR plot with different noise variances for three different images pertaining to one of the three groups.

Image cleaning is more difficult when SAPN is present. In both types of noise, the results of the measurements are low. However, if we made a visually test we can see the edges better preserved with the bandelet transform.

With the results obtained we can conclude that bandelet transform is a good tool for image denoising even when we have very high frequency images as input.

In figure 5 we show a close up of the Barbara image. The wavelet and the contourlet miss a part of the elbow while the bandelet preserve this part of the image after the denoising process.

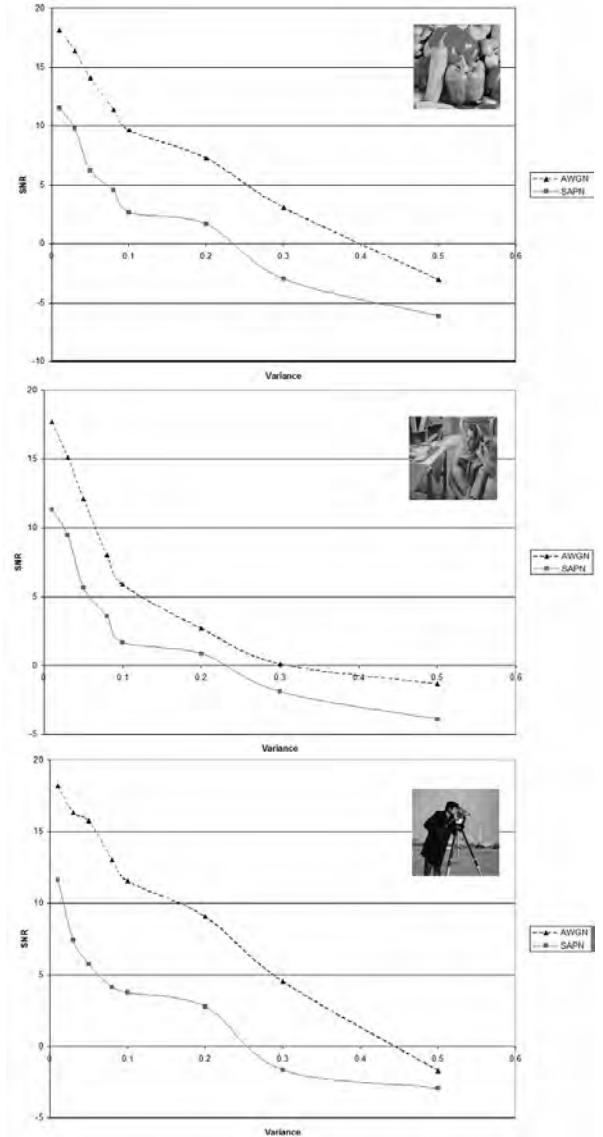


Fig. 4. Bandelet SNR plots for Peppers, Barbara and Cameraman.

5. Conclusions and Further Works

Image denoising is still an open problem in digital image processing. Several algorithms have been proposed in the literature. In this paper we presented a comparison of three novel methods applied to denoising and presented a new denoising algorithm using the bandelet transform. The algorithm performs very well even with images of high frequency content, because it can tackle the classic problem of image oversmoothing and edge preservation.

The comparison presented with three transforms allows us to check the superiority of the bandelet

transform over the wavelet and contourlet. We need to design an intelligent system to select the better threshold for each image according to its frequency content.

In the future we are going to work into defining a strategy to exploit the bandelet transform for image compression, we are planning to design a coder that allow us to preserve the important features of the images such as textures, edges and edge associated details.



Fig. 5. Close up of the denoised Barbara elbow. The upper left corner is the noisy image, the upper right corner is the result obtained with wavelets, the lower left corner the result obtained with contourlets and the lower right corner the result obtained with bandelets.

6. References

- [1] K. Q. Huang, Z. Y. Wu, S. K. Fung and H. Y. Chan, "Color Image Denoising with Wavelet Thresholding Based on Human Visual System Model", *Signal Processing: Image Communication*, vol. 20, no. 2, February 2005, pp. 115 – 127.
- [2] V. Bruni and D. Vitulano D., "Combined Image Compression and Denoising Using Wavelets", *Signal Processing: Image Communication*, vol. 22, no. 1, January 2007, pp. 86 – 101.
- [3] A. Cheveigné and J. Simon, "Denoising Based on Spatial Filtering", *Journal of Neuroscience Methods*, vol. 171, no. 2, June 2008, pp. 331 – 339.
- [4] D. Lazzaro and L. Montefusco, "Edge-Preserving Wavelet Thresholding for Image Denoising", *Journal of Computational and Applied Mathematics*, vol. 210, no. 1-2, December 2007, pp. 222 – 231.
- [5] S. Bacchelli and S. Papi, "Image Denoising Using Principal Component Analysis in the Wavelet Domain", *Journal of Computational and Applied Mathematics*, vol. 189, no. 1, May 2006, pp. 606 – 621.
- [6] A. Hyvärinen and E. Oja, "Independent Component Analysis: Algorithms and Applications", *Neural Networks*, vol. 13, no. 4-5, May-June 2000, pp. 411 – 430.
- [7] A. L. da Cunha, J. Zhou and M. N. Do, "The Nonsubsampled Contourlet Transform: Theory, Design, and Applications", *IEEE Transactions on Image Processing*, vol. 15, no. 10, October 2006, pp. 3089 – 3101.
- [8] S. Mallat and G. Peyré, "A Review of Bandlet Methods for Geometrical Image Representation", *Numerical Algorithms*, vol. 44, no. 3, March 2007, pp. 205 – 234.
- [9] S. Mallat and E. LePennec, "Sparse Geometric Image Representation with Bandelets", *IEEE Transaction on Image Processing*, vol. 14, no. 4, April 2005, pp. 423-438.
- [10] M. N. Do and M. Vetterli, "The Contourlet Transform: An Efficient Directional Multiresolution Image Representation", *IEEE Transactions on Image Processing*, vol. 14, no. 12, December 2005, pp. 2091 – 2106.
- [11] D. L. Donoho, "De-noising by Soft-Thresholding", *IEEE Transactions on Information Theory*, vol. 41, no. 3, May 1995, pp. 613-627.
- [12] H. J. Barnard, Image and Video Coding Using a Wavelet Decomposition, Ph. D. Dissertation, Delft Technological University, Electrical Engineering Department, The Netherlands, 1994.

Color Morphological Image Segmentation on a New Opponent Color Space Based on the Notion of Critical Functions

Israel Santillan
Facultad de Ingeniería
Universidad Autónoma de Querétaro
Cerro de las Campanas, 76000, Qro., México
santilis@gmail.com

Iván R. Terol-Villalobos
CIDETEQ
Parque Tecnológico Querétaro S/N
Sanfandila Pedro-Escobedo, 76700, Qro. México
famter@ciateq.net.mx

Gilberto Herrera-Ruiz
Facultad de Ingeniería
Universidad Autónoma de Querétaro
Cerro de las Campanas, 76000, Qro. México
gherrera@uaq.mx

Abstract

In the present paper color image segmentation, based on the notion of critical functions, is investigated. The notion of critical functions is derived from the well-known waterfall transformation. In the one-dimensional case the critical functions and the waterfall transformation provide similar results. However, in the two-dimensional case, the critical functions give better results for segmenting images than the waterfall transformation, since they take into account the fusion of regional minima (or maxima) to transform the image, providing a good control of the regions merging process. Moreover, contrary to the waterfall transformation which is non-parametric, critical functions allow the introduction of image segmentation criteria to compute an optimal partition. To carry out our study, a color space based on the opponent colors is proposed that we call YO_1O_2 model. The main reason to propose a new color space is that despite the numerous perceptually color spaces (HSV, HSL, HSI, among others) proposed in the literature, they have several drawbacks for color image processing. The performance in image segmentation of both proposals, the color model and critical functions, is illustrated with several examples. Also, some interesting properties of the color model YO_1O_2 and the critical functions are shown.

1. Introduction

In the last years much research has been made on color image filtering and segmentation, as well as on color im-

age retrieval. This research has not only been focused on the use of numerous color spaces that exist in the literature, but also to propose new color spaces and new schemes for color image processing. For color representation one has the perceptual color spaces such as HSL (hue, saturation, luminance), HSV (hue, saturation, value) [21] and HSI (hue, saturation, intensity) [6] as well as perceptually uniform color spaces such as $L^*u^*v^*$ and $L^*a^*b^*$ (luminance L^* , chrominance u^* , v^* , a^* , and b^*) and the CIECAM02 [14]. The first class, the perceptual color spaces, are more frequently used in image processing while the perceptually uniform color spaces are mainly focused in measuring the difference between colors. However, even if these last spaces were developed for measuring color differences, both spaces, $L^*u^*v^*$ and $L^*a^*b^*$ have also been applied to image processing. For example, in [10], a new framework for chromatic filtering of color images was introduced. The chromatic content of a color image is encoded in the CIE uv chromaticity coordinates, and colors are added according to the center of gravity law of additive color mixtures [8], whereas the achromatic content is encoded as CIE Y tristimulus value. In Lucchese et al [9] the same concept (gravity center law) is used for introducing a method for color image enhancement. These ideas [10, 9] were used in [15] for the restoration of ancient chinese paintings and in [5] to propose a morphological multiscale contrast approach for color images. On the other hand, in Hanbury and Serra [7], the use of the mathematical morphology in the CIE $L^*a^*b^*$ space is discussed. In particular, the authors show that it is possible to impose a total order on the color vectors in this space by using a weighting function

and a lexicographical order. However, representations of color in terms of hue, saturation and brightness coordinates are used in image processing since these models are more intuitive (i.e. for color mixing), simpler to calculate, and do not require any calibration information. Nevertheless, these representations suffer from some defects, such as the presence of unstable singularities and non-uniform distributions of their components. Recently, Serra [20] proposed various new perceptual color spaces in order to eliminate some of the drawbacks of these spaces; in particular, luminance and saturation do not verify any norm (L1, L2 or the pseudo-norm Max-Min). A consequence of the use of saturation and luminance definitions that verify the concept of norm is that new colors are not created. These new color models were used by Angulo and Serra [1] for segmenting images. Finally, in Sarifuddin and Missaoui [17], a new perceptually uniform color space, called HCL (hue, chroma, luminance), was proposed and used for content-based image and video retrieval by taking into account that human vision reacts non linearly (logarithmic). This color model was successfully applied for vehicle license plate recognition in [12] and for fingerprint recognition in [4]. Here, a new color space, called YO_1O_2 , is introduced based on the opponent colors concept. It is well-accepted that human vision processes images based on the opponent colors, R-G, (2B-(R+G)); therefore, this color codification is used to specify the saturation and hue, whereas the luminance is codified according to bit-mixing taking into account the order green, red and blue. One perceives better the green color than the red color and the red color better than the blue. This color codification provides an appropriate color specification for color image processing. This is particularly illustrated in this paper in image segmentation using an hierarchical approach based on the critical functions notion. Critical functions work by a flattening process of regional minima (or maxima). The flattening process is stopped to form a critical function at a given level of the hierarchy, when the minima (or maxima) disappear or become another feature. The neighborhood analysis of the minima (or maxima) on the critical function is easily carried out in order to take a decision, according to some homogeneity criteria, to determine whether a set of minima (or maxima) can be merged to form a feature in another level of hierarchy or they remain intact. This paper is organized as follows. In Section 2, the concept of morphological filter and filters by reconstruction are presented. In Section 3, the new color space YO_1O_2 is introduced illustrating some interesting properties of the color model. Next, in Section 4, the notion of critical function is proposed and analyzed. In the same Section, some homogeneity color criteria are introduced in the critical functions to control the merging process.

2 Some Basic Concepts of Morphological Filtering

The basic morphological filters ([19], [22]) are the morphological opening $\gamma_{\mu B}$ and the morphological closing $\varphi_{\mu B}$ with a given structuring element; where, in this work, B is an elementary structuring element (3x3 pixels) that contains its origin. $\check{B}$ is the transposed set ($\check{B} = \{-x : x \in B\}$) and μ is an homothetic parameter. The morphological opening and closing are given, respectively, by

$$\begin{aligned}\gamma_{\mu B}(f)(x) &= \delta_{\mu \check{B}}(\varepsilon_{\mu B}(f))(x) \\ \varphi_{\mu B}(f)(x) &= \varepsilon_{\mu \check{B}}(\delta_{\mu B}(f))(x)\end{aligned}\quad (1)$$

where the morphological erosion $\varepsilon_{\mu B}$ and dilation $\delta_{\mu B}$ are expressed as: $\varepsilon_{\mu B}(f)(x) = \wedge\{f(y) : y \in \mu \check{B}_x\}$ and $\delta_{\mu B}(f)(x) = \vee\{f(y) : y \in \mu \check{B}_x\}$. $\wedge$ is the inf operator and $\vee$ is the sup operator. Another class of filters is composed by the opening and closing by reconstruction. These filters are built using the geodesic dilation and erosion. Where the geodesic dilation and the geodesic erosion of size one are given by $\delta_f^1(g) = f \wedge \delta(g)$ with $g \leq f$ and $\varepsilon_f^1(g) = f \vee \varepsilon(g)$ with $g \geq f$, respectively. When filters by reconstruction are built, the basic geodesic transformations, the geodesic dilation, and the geodesic erosion of size 1 are iterated until idempotence is reached: $R(f, g) = \lim_{n \rightarrow \infty} \delta_f^n(g)$ and $R^*(f, g) = \lim_{n \rightarrow \infty} \varepsilon_f^n(\delta_\mu(f))$. When the function g is equal to the erosion or the dilation of the original function, we obtain the opening and the closing by reconstruction:

$$\tilde{\gamma}_\mu(f) = R(f, \varepsilon_\mu(f)) \quad \tilde{\varphi}_\mu(f) = R^*(f, \delta_\mu(f)) \quad (2)$$

3 Color Model Based on Opponent Colors

3.1 Luminance definition

Different approaches exist to extend the morphological operators from gray level to color; marginal, reduced and vectorial processing. From these approaches, it is preferable to use a purely vectorial approach to process the image all at once using the information of three channels. But one must define a well suited vectorial lattice structure such that a vectorial ordering relation is introduced. An interesting vectorial ordering relation is given by the lexicographic order that enables us to have a total ordering. Then, one codifies the three bytes of the (R,G,B) vector in a 24-bit representation by establishing the importance of the color. One gives to each byte, corresponding to a color channel, the most, the middle or the least significant byte in the 24 bit representation. This approach induces an important dissymmetry

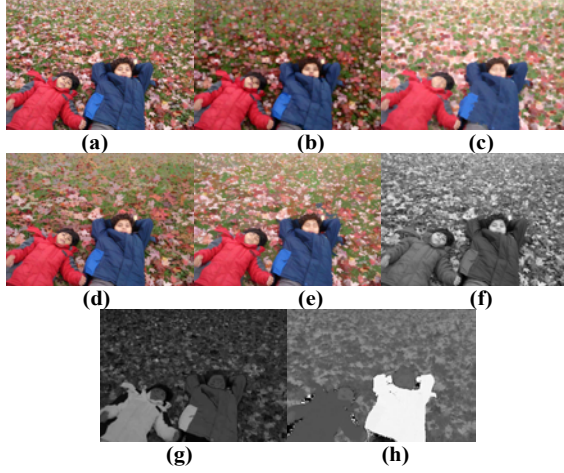


Figure 1. a) Original image; b) and c) Morphological opening and closing size 5; d) and e) Opening and closing by reconstruction size 5; f), g), and h) Luminance, saturation and hue, respectively.

between the different components of the image. It is preferable to mix the bits of the three bytes corresponding to the components of the image [3]. Thus, one first takes the first bit of one component, followed by the first bit of the second component and finally the first bit of the last component. The process is repeated until the 8 bits of the three channels are codified. Nevertheless, different combinations can be taken created. Here, one proposes the choice given by the human visual perception. Consider the following order in the bit mixing where the green channel has the greatest importance and the blue component the smallest. In other words, a 24-bit image is built from the 3 color components $G_7R_7B_7 \cdots G_0R_0B_0$ where G_j, R_j, B_j are the bit codification of the (R,G,B) color with $R_j, G_j, B_j \in \{0, 1\}$. Under this ordering the color components are weighted closely to the luminance Y of the European standard for television given by $L = 0.299R + 0.587G + 0.144B$. In fact, the bit mixing is weighted by $Y = 0.2857R + 0.5714G + 0.1429B$ (4/7 for the green, 2/7 for the red and 1/7 for the blue), which is closely linked to the cone perception in the eye's retina. Moreover, this bijective mapping ensures that the one-dimensional representation contains all the information of the three color components. For a point p of the image, the bit mixing codification is given in this case by $I(p) = \sum_{j=0}^7 2^{3j+2}G_j + \sum_{j=0}^7 2^{3j+1}R_j + \sum_{j=0}^7 2^{3j}B_j$.

In Fig. 1 some examples of morphological filters using the bit mixing approach are illustrated. Figures 1 (b) and (c) show the morphological opening and closing, respectively, of the original image in Fig. 1 (a), whereas in Figs. 1(d) and

(e) the respective opening and closing by reconstruction, are illustrated.

3.2 Saturation and hue definitions in terms of opponent colors

Many color models in the literature have different luminance and saturation definitions. In Serra [20], a very interesting study has shown the main drawbacks of luminance and saturation definitions of the perceptual spaces such as the HSI, HSV and HSL. In particular, the main conclusion of this last study shows that the luminance and saturation must be defined by means of a norm L_1 , L_2 , or the semi-norm $Max - Min$. In Serra [18], the author shows that the norm L_1 presents several advantages from a practical point of view; however, we show here that the semi-norm $Max - Min$ has other interesting characteristics. First, in the chromatic plane the $Max - Min$ becomes a norm. Second, this definition is equivalent to a linear combination of the form:

$$|R - G| + |G - B| + |B - R|/3 =$$

$$(2/3)(\max(R, G, B) - \min(R, G, B))$$

Now, let us introduce our saturation definition in terms of opponent colors, $(2B - (R + G))$ and $(R - G)$, that is closely a representation of $Max - Min$ saturation. In fact, take the expressions $O_1 = (R - G)$ and $O_2 = 2B - (R + G)$ and compute the norm L_1 : $c = \frac{|O_1| + |O_2|}{2} = \frac{|R - G| + |2B - (R + G)|}{2}$, for $R > G > B$ to get $|R - G| + |2B - (R + G)| = (R - G) + (R - B) + (G - B)$, and then $2R - 2B = \max(R, G, B) - \min(R, G, B)$. Similarly, for the other combinations where the blue component is not the middle element, for example: $B > R > G$ one has $R - G + 2B - (R + G)$, thus $2B - 2G = \max(R, G, B) - \min(R, G, B)$. In fact, when B is the median element in either $R > B > G$ or $G > B > R$ configurations, two possible expressions can be computed. For example, for $R > B > G$ one obtains: $|R - G| + |2B - (R + G)| = |R - G| + |(R - B) - (B - R)|$. If $(R - B) > (B - R)$, then one has $2R - 2B = 2(\max(R, G, B) - \text{Med}(R, G, B))$ otherwise, the saturation is given by $2B - 2G = 2(\text{Med}(R, G, B) - \min(R, G, B))$. Nevertheless, it is clear that for blue values close to the Max value or to the Min value, the saturation $\frac{|O_1| + |O_2|}{2}$ is equal to the semi-norm $\max(RGB) - \min(R, G, B)$. Finally, the hue definition is given in terms of degrees in the perceptual spaces. Unlike all color spaces where the origin (zero degrees) is always assigned to the red color, here the blue color is taken as the origin. This decision was made by recalling that the blue color is less sensitive in human vision perception. However, this will enable us to be coherent with the selection of the

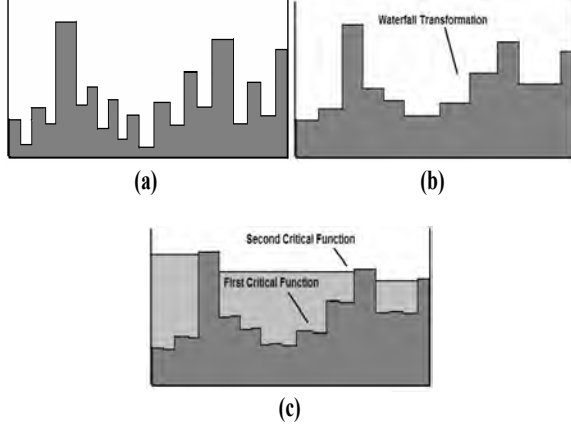


Figure 2. a) Original function; b) waterfall Transformation containing 3 minima; c) First and second critical functions $\Psi_1(f)$ and $\Psi_2(f)$ with 10 and 3 minima, respectively

opponent colors to define the saturation. In our case, the hue is specified as follows:

$$h = \arctan(\sqrt{3} \frac{O_1}{O_2}) = \arctan(\sqrt{3} \frac{(R - G)}{(2B - (R + G))}) \quad (3)$$

The images in Figs 1 (f)-(h) show the luminance Y , the saturation c , and the hue h , respectively. Linked to this color space the following color distance [16] between two colors $k_1 = (Y_1, c_1, h_1)$ and $k_2 = (Y_2, c_2, h_2)$ will be used:

$$d(k_1, k_2) = \sqrt{dy^2 + c_1^2 + c_2^2 - (2c_1c_2 * \cos(dh))} \quad (4)$$

where, $dy = Y_1 - Y_2$ and $dh = h_1 - h_2$.

4 Color Image Segmentation

Image segmentation consists in partitioning the image into different meaningful regions with homogeneous characteristics. In most cases, the segmentation of color images demonstrates to be more useful than the segmentation of monochrome images, because color image reveals more features than monochrome image. A natural question that arises is whether one must focus on the traditional tool for segmenting images called the watershed transform [13]. Here, one illustrates some drawbacks of this transformation and propounds a segmentation method based on a merging process using the notion of critical functions.

4.1 Watershed, waterfall and critical functions

Image segmentation based on the use of the watershed transformation has proved to be an efficient method [13]. Its main drawback consists in the over-segmentation produced by the watershed transformation if applied directly on the images to be segmented. A solution to prevent this over-segmentation consists in a prior selection of a set of markers locating the regions to be extracted. Unfortunately, because of the complexity of the regions in a real scene, this marking is often difficult and sometimes impossible. In Beucher's work [2] another approach for eliminating the over-segmentation problem was proposed. This approach is based on a hierarchical segmentation of the image aiming at merging the catchment basins of the watershed image belonging to almost homogeneous regions. Both, the watershed transformation and the image reconstruction $R^*(f, g)$ are used. Consider a function f in a one-dimensional example as illustrated in Fig. 2 (a) containing 10 minima. Among the various minima of the function f , only three are interesting because they mark regions of higher importance in the image. The other minima can be produced by noise. Consider that the function f is bound and let $W(f)$ be its watershed transformation. $W(f)$ is the set of the watershed lines of f . Let us to build a new function g with $W(f)$

$$g(x) = \begin{cases} f(x) & \text{iff } x \in W(f) \\ 1 & \text{iff } x \in W^c(f) \end{cases}$$

This function g is obviously greater than f . Let us now reconstruct f from g . It is easy to see that the minima of the resulting image correspond to the significant markers of the original image f . In fact, the reconstruction $R^*(f, g)$ fills in (partially) each catchment basin with a plateau at a height equal to the minimum height of the watershed line surrounding this catchment basin. Therefore, if there exists an adjacent catchment basin where the corresponding height is lower than the previous one, the waterfall will not be symmetrical and the plateau generated in the basin will not be a minimum. Moreover, the watershed of $R^*(f, g)$ produces the catchment basins associated with these significant markers. Figure 2(b) shows the output function with 3 minima computed by the waterfall transformation. This procedure, the watershed followed by a morphological image reconstruction, is iterated until a set of homogeneous zones is achieved according to the particular image problem. Recently, a fast implementation of the waterfall was proposed in [11]. The method does not require the setting of parameters. To illustrate the waterfall algorithm applied to color images, the gray-level image in Fig. 3(b) was computed by the color distance (defined in Section(3)) from the original image in Fig. 3(a). Before computing the color

distance, the color image was filtering by an alternating sequential filter using opening an closing by reconstruction by taking into account the total order given by the bit-mixing codification. This filtering process removes the noise of the image. Instead of mapping the color image according to a color distance, one can take into account the bit-mixing image associated with the YO_1O_2 color space. Nevertheless, in the present work, the gray-level image computed by the color distance is used. Then, after computing the morphological gradient ($Grad(f) = \delta_{\mu B}(f) - \varepsilon_{\mu B}(f)$) of the image in Fig. 3(b), the waterfall transformation was applied to the $Grad(f)$. The images in Figs. 3(c) and (d) show the watersheds computed on the gradient of the original image and on the waterfall after 3 iterations. Observe the reduction in the number of regions after 3 iterations of the waterfall. To obtain the color image in Fig. 3(e), the mean color was computed from the original image in Fig. 3(a) for each element of the partition (Fig. 3(d)) and it was used to affect each partition element. The number of color connected components (connected regions with the same color) in Fig. 3(e), permits to have a parameter representing the reduction in complexity of the output image. For example, the original image has 148,276 connected components, while the output image has 412. The image in Fig. 3 (f) was obtained after 5 iterations of the waterfall transformation and contains 131 connected components. The main drawbacks of the waterfall algorithm is the fact of not requiring parameters, then the segmentation can not be controlled. Therefore, another algorithm using the notion of critical functions is proposed for obtaining an hierarchical segmentation. The algorithm proposed here is simpler than that proposed by Beucher [2]. Let us introduce the notion of a critical functions of the minima. The name of critical functions is in the sense of regional minima (or maxima) modifications. First, let us define an operator that increases the gray-level of the minima of a function f ($Min(f)$) given by the following operator:

$$h(f)(x) = \begin{cases} f(x) + 1 & \text{if } x \in Min(f) \\ f(x) & \text{otherwise} \end{cases}$$

Now, we define an operator that increases the gray-level of the minima if and only if the minima will not disappear.

$$\Psi^1(f)(x) = \begin{cases} f(x) + 1 & \text{if } x \in Min(f) \cap Min(h) \\ f(x) & \text{otherwise} \end{cases}$$

In the second step one has $\Psi^2(f)(x) = \Psi^1\{\Psi^1(f)\}(x)$. At the n th step when the stability is reached, the first critical function of the minima is obtained; $\Psi_1(f)(x) = \Psi^n(f)(x) = \Psi^1\{\Psi^n(f)\}(x)$. To obtain the second critical function one must increase the minima by one and then apply again the critical function $\Psi_2(f) = \Psi^n h \Psi_1(f)$. Figure

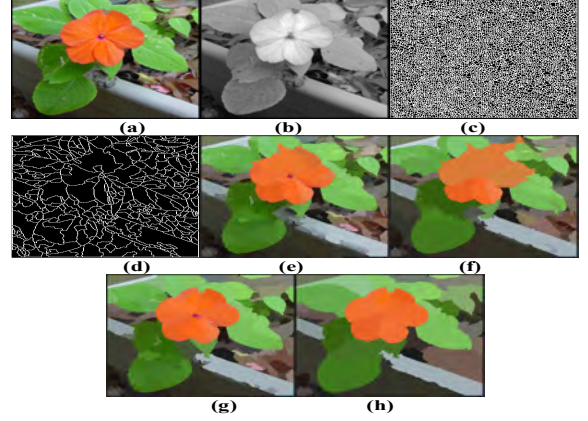


Figure 3. a) Original image; b) Distance image; c) and d) Watersheds of the original function and of the waterfall after 3 iterations; e) and f) Output images computed by the waterfall transformation with 3 and 4 iterations, respectively; g) and h) Output images computed by the critical functions for levels 3 and 4 of the hierarchy, respectively.

2 (c) shows the first and the second critical function $\Psi_1(f)$. Although the sizes of the minima of the second critical function and the output image computed by the waterfall transformation are different, both algorithms detect in this case the same minima. Let us now introduce the hierarchy generated by the critical functions. First, we remember that a pyramid of operators is a family $\{\Gamma_\lambda\}$ depending on a positive parameter λ , with $\mu \geq \lambda \geq 0$, where there exists $\nu \geq 0$, such that $\Gamma_\mu(f) = \Gamma_\nu \Gamma_\lambda(f)$. Then, a composition of critical functions defines a pyramid which implies that each level resumes the past.

Property 1 Let $\{\Psi_i(f)\}$ be a family of critical functions. The family of critical functions $\{\Psi_i(f)\}$, given by $\Psi_i(f) = \Psi^n h \Psi_{i-1}(f)$ with $\Psi_1(f) = \Psi(f)$, defines a pyramid of additive law. This means, for each $\mu \geq \lambda \geq 0$, there exists $\nu \geq 0$, such that $\Psi_\mu(f) = \Psi_\nu h \Psi_\lambda(f)$, with $\mu = \nu + \lambda$, where: $\Psi_i(f) \leq \dots \leq \Psi_2(f) \leq \Psi_1(f)$

By duality, the critical function of the maxima is defined by $\Psi_i^*(f) = [\Psi_i(f^c)]^c$. Consider the minima of a critical function at level n of the hierarchy, given by $\Psi_n(f)$. The minima for each element of this family are the zones in which after the arithmetical addition of one gray level, one of the following situations may occur: a) the minima disappear and b) the minima are merged with adjacent minima and they remain minima at an upper gray level. Figures 3 (g) and (h) illustrate two levels of the hierarchy (Ψ_3 and Ψ_4). The critical functions were computed on the gradient of the im-

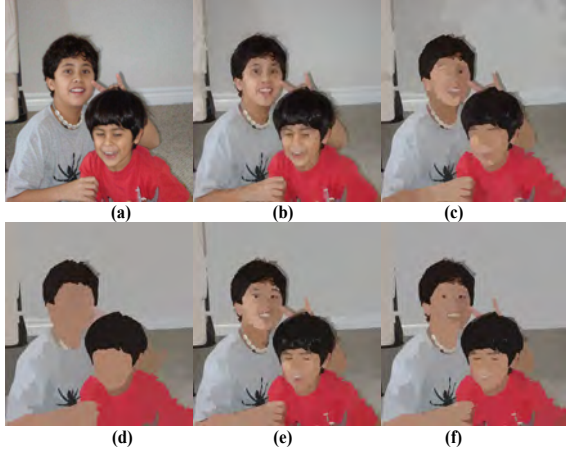


Figure 4. a) Original image; b), c) and d) Three hierarchy levels (Ψ_2 , Ψ_3 , Ψ_4 , respectively) of the critical functions; e) and f) Two levels of idempotent critical functions (Ψ_{m_1} and Ψ_{m_2} ,)

age in Fig. 3 (b). Once the critical functions are obtained, the watershed is applied to the critical function to illustrate the segmented image. The output images have 179 and 39 color connected components, respectively. The critical output images are easily obtained by using hierarchical queues (this data structure has been shown to be extremely efficient in mathematical morphology). Unlike the waterfall transformation, one can introduce neighborhood relation criteria to decide the minima that will be attenuated by the operator. This will be shown in the next section.

4.2 Idempotent critical functions based on color distance

Similarly to the waterfall transformation, the critical functions do not have a manner to control the fusion process described above. Thus, if these transformations are iterated until the stability is achieved, the output image will be composed of only one region (the whole image). It is clear that $\Psi_i(f) \neq \Psi_{i-1}(f)$ at all level i of the hierarchy, and $\Psi_i(f) = \Psi_{i-1}(f)$ when there are no more regions to be removed. Then, to compute a *good* partition of the image, one requires to control the merging process in such a way that at a given level of the hierarchy $\Psi_i(f) = \Psi_{i-1}(f)$, with $\Psi_i(f)$ containing the principal regions of the image. These regions correspond to different levels of the hierarchy. A way of computing an idempotent critical function consists in introducing a distance criterion that permits to take the decision of merging the regions (minima). Figure 4 illustrates the output images computed with and without distance criterion. The images in Figs. 4(b)-(d) were computed

with 3, 4, and 5 critical functions, respectively, without using a distance criterion. The number of regions (color connected components) in the original image is 408,456, while that in the output images in Figs. 4(b)-(d) are 3220, 456 and 65, respectively. Observe, that several important regions of the children's faces (mouths, noses, eyes), particularly in Fig. 4(d), have been removed. By introducing a gray-level distance between the minima to be merged, the images in Figs. 4(e) and (f) were obtained by applying idempotent critical functions Ψ_{m_n} with distance criteria m_n of 20 and 30, respectively (until stability is reached). The output images contain 505 and 203 regions. Even if some regions of the children's faces were preserved, they would be degraded. Thus, the color distance introduced in Section(3) is applied. In this case the criterion based on the color measure is used to take the decision of whether a minimum can be merged. This means that if a minimum is merged with other minima to form a single one a color distance must be computed between the color of the original image of this minimum and color of the neighboring regions. If the largest color distance is above a given value, then the minima will be preserved, otherwise they will be merged. Images in Figs. 5(b) and (c), show the output images computed by both criteria, gray and color distances, respectively. The image in Fig. 5(b), computed by a gray-level distance contains 116 regions, while the image in Fig. 5(c) has 106 regions. Observe in Fig. 5(b), that zones with color and without color are merged since a gray-level distance was used. Conversely, when a color distance is applied (Fig. 5(c)), this merging process does not occur. Finally, a last example using the color distance criterion is shown in Fig. 5(d) and (e). The image in Fig. 5(d) was computed using a criterion value of 30, whereas the image in Fig. 5(e) was obtained using a criterion value of 35. The number of regions of these output images is 357 and 209, respectively. In particular, compare the image in Fig. 5(e) containing 209 regions, with that in Fig. 4(e) with 203 regions. Even though the number of regions is practically similar, the output image using a color distance preserves better the features than when a gray-level distance was used.

5 Conclusion

In the present paper the notion of critical functions for segmenting color images was investigated. The color image segmentation study was carried out on a new color space based on opponent colors YO_1O_2 . The performance of the critical functions for segmenting images was compared with the waterfall transformation. In fact, critical functions provide better results for segmenting images than the waterfall transformation, since they take into account the fusion of regional minima (or maxima) to transform the image, enabling a good control of the merging process of the regions.

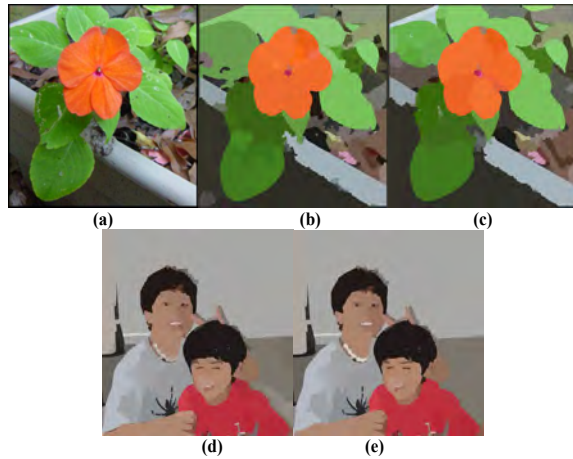


Figure 5. a) Original image; b), c) Idempotent critical functions using gray-level and color distances, respectively; d) and e) Two hierarchy levels of idempotent critical functions

Moreover, contrary to the waterfall transformation that is non-parametric, the use of critical functions permits to easily introduce image segmentation criteria in order to compute an image partition. The performance of both proposals, the color model and critical functions, for image segmentation was illustrated with several examples. Also, some interesting properties of the color model YO_1O_2 and the critical functions were shown.

References

- [1] J. Angulo and J. Serra. Image Color Segmentation Using Bivariate Histograms in Luminance/Saturation/Hue Polar Color Spaces. *Computacion y Sistemas*, 8(4):303–316, 2005.
- [2] S. Beucher. Watershed, Hierarchical Segmentation and Waterfalls Algorithm. In *Mathematical Morphology and Its Applications to Image Processing*, pages 69–76, 1994.
- [3] L. P. Chanussot, J. Total ordering based on space filling curves for multivalued morphology. In *Mathematical Morphology and Its Applications to Image Processing*, pages 51–58, 1998.
- [4] M. C. Espino-Gudiño, V. Rodríguez-Hernández, I. R. Terol-Villalobos, and G. Herrera-Ruiz. Improved afis for color and gray image based on biometric triangulation. *Journal of Computer Science and Technology*, 8(4):228–234, 2007.
- [5] M. C. Espino-Gudiño, I. Santillan, and I. R. Terol-Villalobos. A Morphological Multiscale Contrast Approach for Gray and Color Images Consistent with Human Visual Perception. *Opt. Engineering*, 46(6):416–429, 2007.
- [6] W. R. E. Gonzalez, R. C. *Digital Image Processing*. Prentice Hall, 1992.
- [7] A. Hanbury and J. Serra. Mathematical Morphology in the cielab Space. *J. Vis. Comm. Image Represent.*, 1:21–46, 1990.
- [8] R. W. G. Hunt. *Measuring Color*. Chichester, U. K. Ellis Horwood.
- [9] L. Lucchese, S. K. Mitra, and J. Mukherjee. A new algorithm based on saturation and desaturation in the xy chromaticity diagram for enhancement and re-rendition of color images. In *In Proc. Int. Conf. Image Processing (ICIP 2001)*, volume 2, pages 1077–1080, 2001.
- [10] M. S. K. Lucchese, L. A new class of chromatic filters for color image processing. theory and applications. *IEEE Transaction on Signal Process.*, 62(4):534–548, 2004.
- [11] B. S. Marcotegui, B. Fast implementations of waterfall based on graphs. In *Mathematical Morphology: 40 Years On*, Springer, pages 177–186, 2005.
- [12] J. Martinez-Alcantar and I. Terol-Villalobos. Vehicle License Plate Recognition for Parking Access in Michoacan State. In *In Proc. of IV Congreso Internacional sobre Innovacion y Desarrollo Tecnológico CIINDET 06*, volume 1, pages 1–6, 2006.
- [13] F. Meyer and S. Beucher. Morphological Segmentation. *J. Vis. Comm. Image Represent.*, 1:21–46, 1990.
- [14] N. Moroney. The ciecarn02 color appearance model. In *In Proc. of the Tenth Color Image Conference: Color Science, System and Application*, pages 23–27, 2002.
- [15] S. C. Pei, Y. C. Zeng, and C. H. Chang. Virtual Restoration of Ancient Chinese Paintings Using Color Contrast Enhancement and Lacuna Texture Synthesis. *IEEE Trans. on Image Processing*, 13(3):416–429, 2004.
- [16] V. A. Plataniotis, K. *Color image processing and applications*. Chichester, U. K. Ellis Horwood.
- [17] M. Sarifuddin and R. Missaoui. A New Perceptually Uniform Color Space with Associated Color Similarity Measure for Content-Based Image and Video Retrieval. In *ACM SIGIR Workshop on Multimedia Information Retrieval*, volume 1, pages 1–8, 2005.
- [18] J. Serra. *Image Analysis and Mathematical Morphology, Vol. I*. Academic Press, New York., 1982.
- [19] J. Serra. *Image Analysis and Mathematical Morphology, Vol. II, Theoretical advances*. Academic Press, New York., 1988.
- [20] J. Serra. *Espaces Couleur et Traitement d’Images*. Ecole des Mines de Paris, Internal Note N-34/02/MM, Fontainebleau, France, 2002.
- [21] A. R. Smith. Color gamut transform pairs. *Computer Graphics*, 12(3):12–19, January 1978.
- [22] P. Soille. *Morphological Image Analysis: Principles and Applications*. Springer-Verlag, Heidelberg Berlin, 2nd edition, 2003.

A Statistical Analysis of Visual Cues for Estimating Dense Range Maps

Sergio A. Rosales-Morales and Luz A. Torres-Méndez

Robotics and Advanced Manufacturing Group, CINVESTAV Unidad Saltillo

Ramos Arizpe, Coahuila, Mexico.

sergio.rosales; abril.torres@cinvestav.edu.mx

Abstract

A method for recovering dense range maps from sparse range maps by using statistical analysis of visual cues is presented. The proposed technique is based on constructing a 3D map of a real environment, which in turn, requires visual information to densely cover the environment to be placed. Moreover, the method relies only on the information coming from intensity images taken at the scene in question and compared to existing work, use a small, but representative, set of visual cues to estimate their geometry. The steps for implementing the proposed technique require obtaining an initial (sparse) geometric information from stereo vision. A set of visual characteristics with relevant geometric information is extracted by statistically analyzing small patches from data. These characteristics help to assign confidence values to the sparse range map and apply a range synthesis algorithm based on a Markov field model to estimate a complete dense range map. Preliminary experimental results validate the proposed method.

1 Introduction

One of the main goals of computer vision is to recover the geometric structure of objects from images. Surface depth recovery is essential in multiple applications involving robotics and computer vision. Several important applications (e.g. virtual exploration of remote and hazardous places for security and inspection tasks) require a 2.5D map of the environment, having a robot able to build a reliable map is particularly appealing, as these applications depend on the transmission of meaningful visual and geometric information. However, the problem of inferring the underlying structure from visual images lacks of an analytical solution.

Different types of visual cues have been used, the so-called *shape from X* techniques extract depth information from intensity images by using cues such as shading, texture, retinal disparity and motion. These models are tra-

ditionally based on physical principles of light interaction. However, due to the highly under-constrained characteristic of the inverse problem of these principles, many assumptions about the type of surface and albedo need to be made, which may not be all suitable for the complex real scenes. Dense stereo vision gained popularity in the early 1990's due to the large amount of range data that it could provide [9, 4]. In mobile robotics, a common setup is the use of one or two cameras mounted on the robot to acquire depth information as the robot moves through the environment. Over the past decade, researchers have developed very good stereo vision systems (see [14] for a review). Although these systems work well in many environments, the cameras must be precisely calibrate for reasonably accurate results. Also, the results are limited by the baseline distance between the two cameras. The depth estimates tend to be inaccurate when the distances considered are large. The depth maps generated by stereo under normal scene conditions (i.e., no special textures or structured lighting) suffer from problems inherent in window-based correlation. These problems manifest as imprecisely localized surfaces in 3D space and as hallucinated surfaces that in fact do not exist. Recently, there has been much progress on using learning approaches in stereo vision. One of the top-performing methods for stereo vision is the work by Zhang and Seitz [22], who iteratively estimate the global parameters of a MRF stereo from the previous disparity estimates, without having to rely on ground-truth data. In a more recent work, Saxena *et al.* [13] incorporate monocular cues from a single image into a stereo system for modeling depths and relationships between depths at different points in the image using a hierarchical, multi-scale MRF. A training set, comprising a large set of stereo pairs and their corresponding ground-truth depth maps, is used to model the posterior distribution of the depths given the monocular image features and the disparities.

Saxena *et al.* [12] applied supervised learning to the problem of estimating depth from single monocular cues on images of unconstrained outdoor and indoor environments. This task is difficult since requires a significant amount of

prior knowledge about the global structure of the image. Michels *et al.* [10] used supervised learning to estimate distances to obstacles in order to autonomously drive a remote controlled car. Methods such as shape from shading [21] rely on purely photometric properties, assuming uniform color and texture. Hoiem *et al.* [6] also considered monocular 3D reconstruction, but focused on generating visually pleasing graphical images by classifying the scene as sky, ground, or vertical planes, rather than accurate metric depth maps.

Man-made environments exhibit strong statistical regularities¹ that are exploitable by biological and machine vision systems. Recent emphasis on explicit probability models for images may be due to the growing appreciation for the variability exhibited by the images and the realization that exact mathematical models may not be feasible. The key to a statistical approach is to have probability models that capture the essential variability and yet are tractable. Earliest, and still widely used, probabilistic models for images were based on Markov Random Field (MRF) models [20]. MRFs, in particular, define a class of statistical models which enable to describe both the local and global properties of structures in images. Potetz and Lee [11] measure the correlations between linear properties of range images and linear properties of image intensity data, to explore the structure of the correlations that could usefully underlie 3D judgments from intensity data in images of natural scenes (e.g., shape from shading). Those few studies have uncovered meaningful and exploitable statistical trends in real scenes which may be useful for designing new algorithms in surface inference, and also for understanding how humans perceive depth in the real scenes. A greater understanding of how real images are formed could lead to substantial insight into how depth information may be inferred from single images. In spite of the recent studies lead to an agreement of that the 3D structure can be estimated directly from the same image using the statistical analysis, it has been surprisingly difficult to establish the precise quantitative link.

In particular, we investigate the problem of inferring a dense and complete depth map from the information that stereo images can give. Using stereo vision techniques, one can obtain a depth map in an economical way, but they are usually incomplete or with limited resolution. Other more recent approaches recover the geometric structure through several images using statistical learning techniques. We can start saying that natural images, including images taken from natural scenes as well as those taken from man-made scenes, are statistically redundant. This is the reason why to humans is easier to interpret the geometric information

of a scene through one image. However, this is a hard task for a computer as it is necessary to have *a priori* knowledge about the scene.

Thus, the challenge becomes one of trying to infer from the sparse depth map obtained from stereo, an overall concept of shape and size of the structures within the scene.

The proposed method for recovering the depth information uses the initial geometric information (sparse) obtained by a stereo vision algorithm and, through a statistical analysis of small patches from the stereo images, we obtain the set of characteristics with relevant information about the geometry of the scene. These characteristics help to assign a confidence value to the depth map and thus be able to synthesize in a more optimal way a dense depth map using a statistical learning model based on Markov random fields.

2 Our framework

This research work focuses on recovering the depth information of a man-made indoor scene (e.g. an office, a room) by incorporating visual cues to an initial sparse depth map obtained from stereo. Man-made indoor environments have inherent geometric and photometric characteristics that can be exploited to help in the reconstruction. The methodology consists on designing a statistical learning model for depth estimation, which involves the following steps. First, we apply a statistical analysis on an image database. Second, we obtain a pair of stereo images from the indoor scene to model and use a stereo algorithm to compute an initial sparse depth map. Third, we apply the filters obtained in the first step to the stereo images in order to obtain the regions with relevant geometric information (the internal representation). With the internal representation, we can assign confidence values according to the ternary values obtained. These values will indicate the *filling* order of the missing range values. And finally, we use the non-parametric range synthesis method to estimate the missing range values and obtain a dense depth map. In the following sections, the statistical learning model is explained in more detail.

2.1 Statistical analysis on the image database

Given that our method is based on a statistical analysis, the type of images to analyze in the database must contain characteristics and properties similar to the scenes of interest, as we focus on man-made scenes, we should have images containing those types of images. However, we start our experiments using a public available image database [16] that contains scenes of natural images. As these database contains important changes in depth in their

¹Real scenes are constrained by many regularities in the environment, such as the natural geometry of objects and their arrangements in space, natural distributions of light, and regularities in the observer's position.

scenes, this turns out to be the main characteristic to be considered so that our method can be functional.

The statistical analysis of small patches implemented is based on part of the Feldman and Yunes algorithm [18]. This algorithm consists in extract characteristics of interest from an image through the observation of a image database and obtain an internal representation that concentrates the relevant information in a form of a ternary variable. To generate the internal representation we follow three steps. First, we reduce (in scale) the images in the database (see Figure 1a). Then, each image is divided in patches of same



(a) Examples of images (already scaled) in the database.



(b) The ternary images of images in (a).

Figure 1.

size (e.g. 13x13 pixels), with these patches we make a new database which is decomposed in its principal components by applying PCA to extract the most representative information, which is usually contained, in the first five eigenvectors. In Figure 2 the eigenvectors are depicted. These



Figure 2. The five eigenvectors

eigenvectors are the filters that are used to highlight certain characteristics on the stereo images. The last step consists on applying a threshold in order to map the images onto a ternary variable where we assign -1 value to very low values, 1 to high values and 0 otherwise. This way, we can obtain an internal representation.

$$\xi_I : G \rightarrow \{-1, 0, 1\}^k$$

where k represents the number of filters (eigenvectors). G is the set of pixels of the scaled image. ξ_I depends only on $W_I(s, \delta)$. The function

$$\varphi : W \rightarrow \varphi(W) \in \{-1, 0, 1\}^k$$

is defined according to the homeostasis principle: the central configuration (or average) $(0, \dots, 0)$ corresponds to a "typical" region of the image, and the other configuration

are formed from a deviation of this average. Figure 1b shows an example of the internal representation of images shown in Figure 1a.

2.2 Obtaining the initial sparse depth map from stereo and the internal representation

In order to compute the disparity between the stereo images, we implemented the method based on the Shirai's algorithm [19] using the epipolar geometry and the Harris corner detector [17] as constraints. Given the human eyes position and the way we move them, the images we receive in each eye are practically the same, with a difference in the relative position of objects. These relative differences in the position in each image (disparity) has a direct relationship with the distance (depth) to which objects are found among them and from the observer. The underlying structure recovered from stereo vision is the position of the objects in the scene.

2.3 Non-parametric MRF Model for Range synthesis

After obtaining the sparse depth map and the internal representation, we can apply the range synthesis method proposed in [15]. In general, the method estimates dense depth maps using intensity and partial range information. The Markov Random Field (MRF) model is trained using the (local) relationships between the observed range data and the variations in the intensity images and then used to compute the unknown range values. The Markovianity condition describes the local characteristics of the pixel values (in intensity and range, called from now on *voxels*). The range value at a voxel depends only on neighboring voxels which have direct interactions on each other. Due to space limitations we describe the non-parametric method and skip the details of the basis of MRF, the reader is referred to [15] for further details.

To compute the MAP estimate for a depth value R_i of the augmented voxel V_i , one first needs to construct an approximation to the conditional probability distribution $P(f_i | f_{N_i})$ and then sample from it. For each new depth value $R_i \in \mathbb{R}$ to estimate, the samples, which correspond to the neighborhood system for the voxel location i , $\mathcal{N}_i$, are queried and the distribution of R_i is constructed as a histogram of all possible values that occurred in the samples. $\mathcal{N}_i$ is a subset of the real infinite set of augmented voxels, denoted by $\mathcal{N}_{real}$.

Based on the MRF model, we assume that the depth value R_i depends only of its immediate neighbors in intensity and range, i.e. of N_i . If we define a set

$$\Gamma(R_i) = \{N^* \subset \mathcal{N}_{real} : \|N_i - N^*\| = 0\} \quad (1)$$

containing all occurrences of N_i in $\mathcal{N}_{real}$, then the conditional probability distribution of R_i can be estimated with a histogram based on the depth values of voxels representing each N^* in $\Gamma(R_i)$.

Unfortunately, we are only given $\mathbf{V}$, i.e., a finite sample from $\mathcal{N}_{real}$. Thus, there might not be any neighborhood containing exactly the same characteristics in intensity and range as N_i in $\mathbf{V}$. Thus, we must use a heuristic which let us find a plausible $\Gamma'(R_i) \approx \Gamma(R_i)$ to sample from.

Let $\mathcal{A}_p$ be a *local* neighborhood system for the augmented voxel p , that comprises nearby neighborhoods within a radius r ,

$$\mathcal{A}_p = \{A_q \in \mathcal{N} \mid \text{distance}(p, q) \leq r\} \quad (2)$$

where R_q is a known depth value. In other words, the local neighborhood system contains all the neighborhoods which center voxels have already assigned labels (depth values), located at a maximum distance r from the location of voxel p . This set will conform the training/sample data for that particular voxel p .

In the non-parametric approach, a depth value R_p from the augmented voxel V_p with neighborhood N_p , is synthesized by first selecting the most similar neighborhood (N_{best}) to N_p , i.e., the closest match to the region being filled in,

$$N_{best} = \underset{A_q \in \mathcal{A}_p}{\operatorname{argmin}} \|N_p - A_q\|, \quad (3)$$

Second, all the neighborhoods A_q in $\mathcal{A}_p$ that are similar (within a threshold ϵ) to this N_{best} are included in $\Gamma'(R_p)$, as follows

$$\|N_p - A_q\| < (1 + \epsilon) \|N_p - N_{best}\| \quad (4)$$

The similarity measure $\|\cdot\|$ between two generic neighborhoods N_a and N_b is described over the partial data in the two neighborhoods and is calculated as follows,

$$\|N_a - N_b\| = \sum_{\vec{v} \in N_a, N_b} G(\sigma, \vec{v} - \vec{v}_0) \cdot D, \quad (5)$$

$$D = \sqrt{(I_{\vec{v}}^a - I_{\vec{v}}^b)^2 + (R_{\vec{v}}^a - R_{\vec{v}}^b)^2}, \quad (6)$$

where $\vec{v}_0$ represents the augmented voxel located at the center of the neighborhoods N_a and N_b , $\vec{v}$ is a neighboring voxel of $\vec{v}_0$. I^a and R^a are the intensity and range values of the neighboring augmented voxels of the depth value $R_p \in \vec{v}_0$ to synthesize, and I^b and R^b are the intensity and range values to be compared with and in which, the center voxel $\vec{v}_0$ has already assigned a depth value. G is a 2-D Gaussian kernel applied to each neighborhood, such that those voxels near the center are given more weight than those at the edge of the window.

We can now construct a histogram from the depth values R_p in the center of each neighborhood in $\Gamma'(R_p)$, and randomly sample from it. R_q is then used to specify R_p . For each successive augmented voxel this approximates the maximum *a posteriori* estimate.

2.4 Computing the priority values to establish the filling order

Critical to the quality of the reconstruction is the order in which a voxel, whose range value is to be synthesized, is selected. A suitable ordering should be based on the amount of available information in the voxel's neighborhood, such that voxels containing the maximum number of neighboring augmented voxels are synthesized first. It was observed that the reconstruction across depth discontinuities is often problematic as there is comparatively little constraint for probabilistic inference at these locations. Such regions are often identified by the internal representation described above. Thus, we can take advantage of the internal representation in the reconstruction sequence by assigning high confidence to voxels with ternary variable 1, so that they can be synthesized first, low confidence to voxels with ternary value of 0 and -1 so that they are deferred as much as possible.

Summarizing, the reconstruction sequence of the depth values of those voxels is based on essentially two factors: 1) the number of neighboring augmented voxels (i.e. locations with already assigned range and intensity) and, 2) the value of the ternary variable on that voxel. Priority values are computed based on these factors and are assigned to each voxel for reconstruction, such that as we reconstruct, the voxel with the maximum priority value is selected. If more than one voxel shares the same priority value, then the selection is done randomly.

2.4.1 Computing the Priority Values

The reconstruction sequence depends entirely on the priority values that are assigned to each voxel. The priority is biased toward those voxels that are surrounded by high-confidence voxels.

The region to be synthesized, i.e., the *target* region is indicated by $\Omega = \{\omega_i \mid i \in \mathcal{A}\}$, where $\omega_i = R(x_i, y_i)$ is the unknown depth value at location (x_i, y_i) and $\mathcal{A} \subset Z_m$ is the set of subscripts for the unknown range data. The input intensity (I) and the known range values (R_k) together form the *source* region, and is indicated by ζ . This region ζ is used to calculate the statistics between the intensity and the input range for reconstruction, as it was described in Section 2.3. Let V_p be an augmented voxel with unknown range located at Ω and N_p be its neighborhood, which is a $n \times n$ square window centered at V_p .

For all voxels $V_p \in \Omega$, their priority value is computed as follows:

$$P(V_p) = \frac{\sum_{i \in N_p} C(V_i)F(V_i)}{|N_p| - 1}, \quad (7)$$

where $|N_p|$ is the total number of voxels (augmented or not) in N_p . At the beginning, the confidence of each voxel $C(V_i)$ has assigned a 1 if its ternary variable is 1, a 0.8 if its ternary value is 0 and 0.2 if it is -1 . $F(V_i)$ is 1 if its intensity and range values are filled and 0 if the range value is unknown. The priority $P(V_p)$ may be thought of as a measurement of the amount of reliable information surrounding the voxel V_p . Thus, as we reconstruct, those voxels whose neighborhood has more of their voxels already filled and high confidence value, are synthesized first, with additional preference given to voxels that were synthesized early on.

Once all priority values of each voxel on Ω have been computed, we find the voxel with the highest priority. We then use our MRF model to synthesize its depth value. After a voxel has been synthesized, the priority of its neighboring voxels is updated.

3 Experimental Results

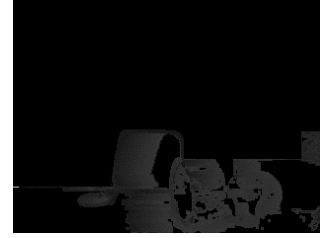
We present some experimental results of the proposed method. First, we applied the Shirai's algorithm to obtain a disparity map of the input stereo images adding as a restriction the points detected by the Harris algorithm, in order to reduce the searching area of corresponding pixels. Figures 3a-b, show one of the stereo images and the range map obtained after applying the stereo algorithm, respectively. In this range map, it is relatively easy to see that there are regions where the algorithms were not capable of finding corresponding points.

The statistical analysis to obtain the internal representation was done on 50 images from the database of van Hateren [16]. These images were scaled by 50% of the original size as a part of the characteristic extraction process. The scaled images were divided in patches of 13×13 pixels to form a new database of 1000 elements (some of them are shown in Figure 1a). We use the 5 eigenvectors depicted in Figure 2 to filter the stereo images. Figures 4a-b show one of the filtering result of the left stereo image and the internal representation after applying a threshold of 0.7 y 0.3, respectively. It can be observed that the internal representation enhances (white pixels) those regions where exist changes in depth, which represent objects located at different distances.

Now, we can use the internal representation as a basis to assign high confidence values to those pixels in the stereo images. We choose to assign a 1 (the highest confidence) to those pixels that coincide with the 1 values (white), 0.8



(a) The left stereo image



(b) The sparse depth map

Figure 3. The input intensity and range data.

to pixels that coincide with the -1 (gray) and 0.2 to the rest pixels. With this values assigned to the sparse range map, we can compute priority values at each pixel having a missing range value to set the filling order and use the non-parametric (NP) MRF model to estimate the dense range map. Figure 5 shows the synthesized depth map after running the NP-MRF algorithm.

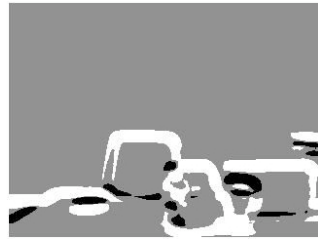
While it seems sufficient to infer depth values by learning the local relationships among the nodes in a *local* neighborhood system around the voxel whose depth value is to be inferred, yet, there are some issues. In particular, when the local information does not capture correctly the global, smooth variations in depth, the reconstructions may lead to piecewise constant (i.e., fronto-parallel) surfaces along surfaces like walls. Consider imaging a wall in which thick stripes of range data are missing. If we only observe the local neighborhoods on each side of the missing range, we will tend to get estimates of constant depth, because there were no observations of the wall sloping away at the desired depths.

4 Discussion and Future Work

There exists some implementation issues, for example, the searching space depends on the parameter d , i.e., the distance from the voxel to be synthesized p to the farthest neighboring voxel, which neighborhood is going to be compared to. Thus, if d is equal to the size of the image, then N_p is compared with all possible neighborhoods in the image, which would be a time consuming process. The exhaustive search takes $O(n^2)$ for each voxel to be synthesized, where n is the total number of known augmented voxels



(a) One of the resulting filtered images.



(b) The internal representation.

Figure 4. Experimental results.

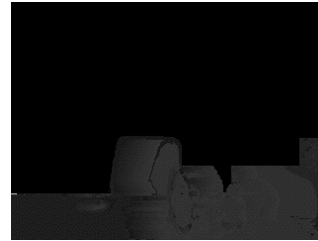
whose neighborhoods are being compared to. Therefore, d should be of a reasonable size, not too big nor too small, so that appropriate candidates, which are similar enough in their neighborhoods, can be found. However, if the searching space must be large for whatever reason, the searching time can be reduced by indexing the n augmented voxels using a kd -tree structure, so that the new computational time is $O(\log n)$.

Neighborhoods should be as small as possible to minimize processing but should be large enough that features are not missed. Additionally, if neighborhoods are too large, small features are more easily missed in the processing phase. In other words, ideal neighborhoods would include enough voxels to easily distinguish variations in surface structure but be no larger.

There is also the question of how "similarity" between two neighborhoods is measured. This is partly determined by the model through neighborhood size. However, it is also determined by how we measure the classification probability. We determine a good fit probabilistically by comparing the set of statistics obtainable from the available data. The consequence of this, however, is that when the MRF model is undertrained, the statistics will not change rapidly as the similarity between the neighborhood of the voxel to synthesize and the neighborhood from the training data diminishes. Therefore neighborhoods which are not all that similar could be given a high similarity measure. On the other hand, if the MRF model is overtrained, then the statistics obtainable from the training data would have a high entropy, and again dissimilar neighborhoods could be given a high similarity measure.



(a) Mapa de prioridad



(b) The synthesized range map

Figure 5. Experimental results.

References

- [1] Y. Boykov, O. Veksler, and R. Zabih. Fast approximate energy minimization via graph cuts. *IEEE Trans. on PAMI*, 23(11):1222–1239, 2001.
- [2] P.F. Felzenszwalb and D.P. Huttenlocher. Efficient belief propagation for early vision. *IJCV*, 70(1):41–54, 2006.
- [3] A.W. Fitzgibbon and A. Zisserman. Automatic 3d model acquisition and generation of new images from video sequences. In *Proc. of European Signal Processing Conf.*, pages 1261–1269, 1998.
- [4] P. Fua. A parallel stereo algorithm that produces dense depth maps and preserves image feature. *Machine Vision and Applications*, 6:35–49, 1993.
- [5] J.M. Hammersley and P. Clifford. Markov field on finite graphs and lattices. In *Unpublished*, 1971.
- [6] D. Hoiem, A.A. Efros, and M. Herbert. Geometric context from a single image. In *Proc. of ICCV*, 2005.
- [7] C.Q. Howe and D. Purves. Range image statistics can explain the anomalous perception of length,. In *Proc. Natl. Acad. Sci. U.S.A.*, volume 99, pages 13184–13188, 2002.
- [8] J. Sun, N. Zheng and H. Shum. Stereo matching using belief propagation. *IEEE Trans. on PAMI*, 25(7):787–800, 2003.

- [9] J.J. Little and W.E. Gillett. Direct evidence for occlusion in stereo and motion. *Image and Vision Computing*, 8:328–340, November 1990.
- [10] J. Michels, A. Saxena, and A.Y. Ng. High speed obstacle avoidance using monocular vision and reinforcement learning. In *Proceedings of the 21th International Conference on Machine Learning (ICML)*, 2005.
- [11] Potetz, B., Lee, T.S. Scaling Laws in Natural Scenes and the Inference of 3D Shape. In *Proceedings of the Advances in Neural Information Processing Systems (NIPS)*, MIT Press. 2006.
- [12] Ashutosh Saxena, SungH. Chung, and Andrew Y. Ng. Learning depth from single monocular images. In *Proceedings of the Advances in Neural Information Processing Systems (NIPS)*, MIT Press. 2006.
- [13] Ashutosh Saxena, Jamie Schulte and Andrew Y. Ng. Depth estimation using monocular and stereo cues. In *Proceedings of International Joint Conference on Artificial Intelligence (IJCAI)*. 2007.
- [14] D. Scharstein and R. Szeliski. A taxonomy and evaluation of dense two-frame stereo correspondence algorithms. *International Journal of Computer Vision*, 47(7):7–42. 2002.
- [15] L.A. Torres-Méndez and G. Dudek. Statistics of Visual and Partial Depth Data for Mobile Robot Environment Modeling. *LNAI 4293: MICAI*, Springer. A. Gelbukh and C.A. Reyes-García (Eds.), pp. 715–725, 2006.
- [16] J.H. van Hateren database: <http://hlab.phys.rug.nl/mlib/index.html>.
- [17] C. Harris and M.J. Stephens. A Combined Corner and Edge Detector. In *Alvey Conference*, pages 147–152, 1988.
- [18] T. Feldman and L. Younes. Homeostatic image perception: An artificial system. In *Computer Vision and Image Understanding* 102, pages 70–80, 2006.
- [19] Reinhard Klette, Karsten Schlens, and Andreas Koschan. *Computer Vision, Three-dimensional Data from Images*. Springer-Singapore, 1998.
- [20] G. Winkler. *Image analysis, random fields, and dynamic Monte Carlo methods: a mathematical introduction*. Springer-Verlag, Berlin, 1995.
- [21] R. Zhang, P.-S. Tsai, J.E. Cryer and M. Shah. Shape from shading: A survey. *IEEE Transactions on PAMI*, 21(8):690–706, 1999.
- [22] L. Zhang and S.M. Seitz. Parameter Estimation for MRF Stereo, In *CVPR, Vol. 2*, pages 288–295, 2005.

A Feature Extraction Method Based on Morphological Operators for Automatic Classification of Leukocytes

Pilar Gómez-Gil¹, Manuel Ramírez-Cortés¹, Jesús González-Bernal¹, Ángel García Pedrero¹,
César I. Prieto-Castro¹, Daniel Valencia¹, Rubén Lobato², José E. Alonso²

¹*Instituto Nacional de Astrofísica, Óptica y Electrónica*

²*Instituto Mexicano del Seguro Social*

{pgomez, jmram, jagonzalez, agarciapedrero}@inaoep.mx

Abstract

In this paper we present preliminary results obtained from the application of morphological operator pecstrum, for the extraction of discriminating characteristics in leukocytes and similar artificial images. Experts have identified six categories of leukocytes, very similar in shape and size, which makes them extremely difficult to distinguish automatically or even by non-expert humans. A feature vector based on a 7-component pecstrum, normalized area, and nucleus - cytoplasm area ratio, was tested using 4 kinds of recognizers: Euclidean distance, k-nearest Neighbor, Back Propagation Neural Net and Support Vector Machine. Using 36 patterns for training and 18 for testing, recognition of 87% was obtained in the best case, which is encouraging, given the complexity of the problem. The amount of samples used at this point for experiments is not statistically representative, however these results are promising and more experiments will be carried out.

1. Introduction

The use of differential counting and assessment of blood white cells morphology from bone marrow is an important source of data for clinical cytology in a wide range of pathologies in the medical fields of oncology and hematology (see, for example [1-3]). In order to accomplish these activities, an important attribute to consider in leukocytes analysis is the age of the cell, which is strongly reflected in a morphological evolution of its nucleus and cytoplasm.

Cytology experts have identified six categories of this type of cells, according to their maturity stage, that ordered from youngest to oldest are named: Myeloblast, promyelocyte, myelocyte, metamyelocyte, band, and polymorphonuclear leukocytes (PMN) [4].

Some typical examples of these categories are shown in figure 1, where nucleus and cytoplasm of each cell are in different colors. Myeloblast is the earliest recognizable myeloid cell, and normally it has a large round to oval nucleus. Promyelocyte is slightly larger than a myeloblast. Its nucleus, although similar to a myeloblast, shows slight chromatin condensation and less prominent nucleoli with respect to the size of the cytoplasm. Myelocytes are slightly smaller than promyelocytes and have eccentric round-oval nuclei, often flattened along one side. In a typical image of a myelocyte, the proportion of the area occupied by the nucleus with respect to the cytoplasm is about 50-50 %. Metamyelocytes (10-18 μ) are slightly smaller than myelocytes. They have kidney shaped indented nuclei and relatively dense chromatin, especially along the nuclear membrane. Bands are usually characterized by a U-shaped or deeply indented nucleus. Opposite sides or lobes are of roughly equal size or diameter, and there is no nuclear constriction greater than one half of the lobe diameter. Polymorphonuclear (PMN) leukocytes or segmented neutrophils are characterized by definite lobation with thin thread-like filaments of chromatin joining the 2-5 lobes.

From the previous description of typical forms of leukocytes, we can notice that definition of characteristics to discriminate among them is not straightforward. It is required to use a feature extractor able to consider shapes and at the same time, be tolerant enough to variation among patterns due to translations and rotation of the images. An additional problem associated to the creation of automatic leukocyte recognizers is the lack of public databases of leukocytes images to use for experimentation [5]. The scarce published research that we have found related to this kind of problem (for example [14]), is normally carried out using private databases which in many cases present a unbalanced number of images with respect to classes, or have been digitized under very different conditions.

In this research we explore the use of mathematical morphology as a tool to create a feature extractor able to discriminate leukocytes. To automatically deal with this classification could be very useful in the construction of other automatic systems, as classification of different kinds of leukemia, counting of cells etc. The paper is organized as follows: Section 2 describes theory associated to Morphology and its operator spectrum, Section 3 describes the feature vector. Section 4 describes the experiments developed to test the performance of the proposed feature vector, and analyzes the results. Section 5 discusses conclusions and perspectives.

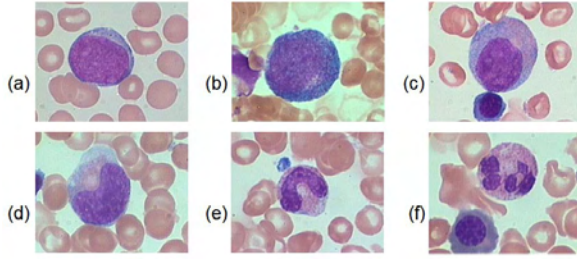


Figure 1. Maturity stages of white blood cells.
(a) Myeloblast. (b) Promyelocyte.
(c) Myelocyte. (d) Metamyelocyte. (e) Band,
(f) Polymorphonuclear leukocytes (PMN).

2. Morphology and pecstrum

Mathematical morphology aims to quantitatively describe operations effective for the shape of objects in an image [6, 7]. Operations are described by combinations of a basic set of numerical manipulations between an image A and a small object B , called a structuring element, which can be seen as a probe that scans the image and modifies it according to some specified rule. The shape and size of B , typically much smaller than the image A , together with the specific rule, define the characteristics of the performed process.

A very interesting morphological operator is the pattern spectrum or pecstrum. This operator decomposes the target image in morphological components, according to the shape and size of the structuring element, providing a quantitative analysis of the morphological content of the image [8, 9]. Pecstrum has been used in the last years with different purposes: analysis of partial discharges in high voltage systems [10], automatic recognition of automotive plates [11], texture analysis in several applications, such as images of debris particles in polymers and composite materials [12, 13], and cytology of bone

marrow images for the counting of white blood cells based on morphological granulometries [14].

Binary mathematical morphology is based on two basic operators: Dilation, and erosion. Both are defined in terms of the interaction of the original image A to be processed, and the structuring element B . Next both basic operators are defined.

2.1 Morphological dilation

It is defined as the set union of the objects A obtained after the translation of the original image for each coordinate pixel b in the structuring element B :

$$A \oplus B = \bigcup_{b \in B} T_b(A) \quad (1)$$

Binary dilation can be interpreted as the combination of two sets by using the vector additions of set elements, called the *Minkowski Addition*. This operation is expressed as:

$$A \oplus B = \{r \mid r = a + b \quad \forall a \in A \text{ and } b \in B\} \quad (2)$$

2.2 Morphological erosion.

Erosion is the morphological dual of the dilation. It is defined in terms of the Minkowski subtraction as:

$$A \ominus B = \{r \mid (r + b) \in A \quad \forall b \in B\} \quad (3)$$

This definition can be expressed in terms of set intersections as:

$$A \ominus B = \bigcap_{b \in B} T_{-b}(A) \quad (4)$$

2.3 Opening filter.

An important operator, which is the backbone of the pattern spectrum, is the *opening* morphological filter, defined as an erosion operation followed by a dilation using the same structuring element. The opening operator is defined as:

$$A \circ B = (A \ominus B) \oplus B \quad (5)$$

Dilation tries to undo erosion operation. However, some details closely related with the shape and size of the structuring element will vanish. Furthermore, an object disappearing as consequence of erosion can not be recovered.

2.4 Closing filter.

Closing morphological operator, which is the dual of the *opening*, is defined as dilation followed by an erosion operation using the same structuring element:

$$A \bullet B = (A \oplus B) \ominus B \quad (6)$$

Opening and closing filters have been used as discriminators for filtering, segmentation, edge detection, differential counting, or numerical analysis of shapes.

2.5 Pecstrum.

Pattern spectrum of a compact binary image $A \subseteq R^2$, relative to a convex binary pattern $B \subseteq R^2$ called the *structuring element*, is defined as the differential size distribution function:

$$P_x(n, B) = \frac{dM(A \circ nB)}{dn}, \quad n \geq 0 \quad (7)$$

where M represents the area measured in the intermediate operations, and nB is the n -times dilated structuring element. *Pecstrum* has the property of invariance to translation and rotation when B is an isotropic structuring element. Scale is determined by the size of the structuring element. The discrete version is given by:

$$P(n, B) = \frac{M[A \circ nB] - M[A \circ (n+1)B]}{M[A]} \quad (8)$$

Figure 2 shows the obtained *pecstrum* for two images. It was calculated using a structuring element formed by a small circle with diameter of 6 pixels.

3. A feature vector for leukocytes discrimination

A composed feature vector based on the discussed morphological information is proposed as follows:

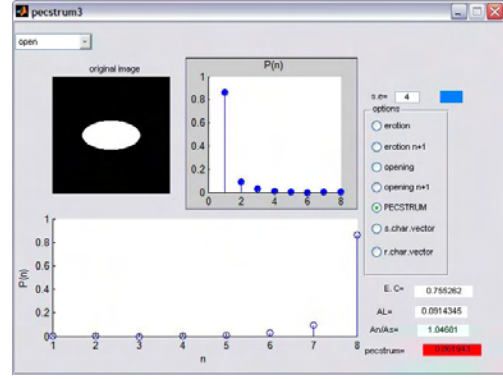
$$\vec{x} = \{A_L : R_{nc} : P(n, B)\} \quad (10)$$

where:

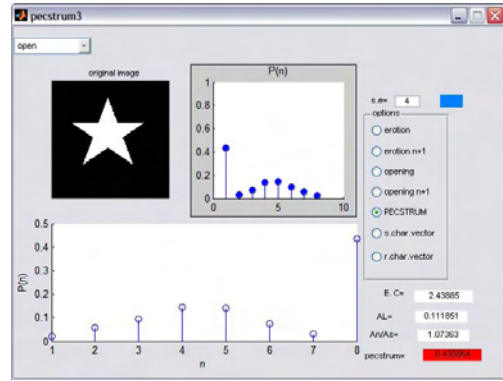
A_L = Leukocyte normalized area

R_{nc} = Nucleus-cytoplasm ratio

$P(n, B)$ = 7 components of Pecstrum of nucleus. If the image generates less than 7 components, the missing values are set to 0.0.



(a) Ellipse



(b) Star

Figure 2. Pattern spectrum of two geometric forms.

This vector contains information that in general human experts could consider to classify a leukocyte:

- shape of the nucleus, represented by the pecstrum $P(n, B)$
- relationship among the area of the cytoplasm and the area of the nucleus, represented by R_{nc}
- size of the cell, represented by A_L

Figure 3 (a) shows an image of a leukocyte, and 3 (b) the results obtained for the computation of the composed feature vector described before. The segmented image is plotted in the up-left part of figure (b). The obtained feature vector is plotted in the lower chart of the figure, which is:

$$\vec{x} = \{A_L : R_{nc} : P(n)\} = \{0.2700, 0.2878, 0.2709, 0.6216, 0.0858, 0.0075, 0.0009, 0.0, 0.0\}$$

Table 1 shows means and variances of the feature characteristics for each class of leukocytes, obtained from the dataset described at section 4.1.

4. Experimental description and results

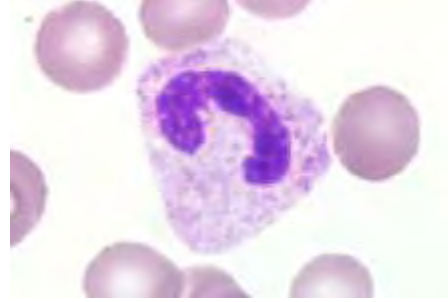
4.1 Data Set.

Input data was obtained from hand-segmented images of white blood cells in different maturity stages, some examples are shown in figure 4. Data set consists of 54 patterns corresponding to 7 artificial and 2 real images for each class. Artificial images were needed due to the lack of a public database to carry out the experiments. An artist generated the artificial segmented images hand-painting them in black and white pixels, using Paint. Figure 4 (a) shows examples of segmented real cells corresponding to each class. Figure 4 (b) shows examples of artificial cells.

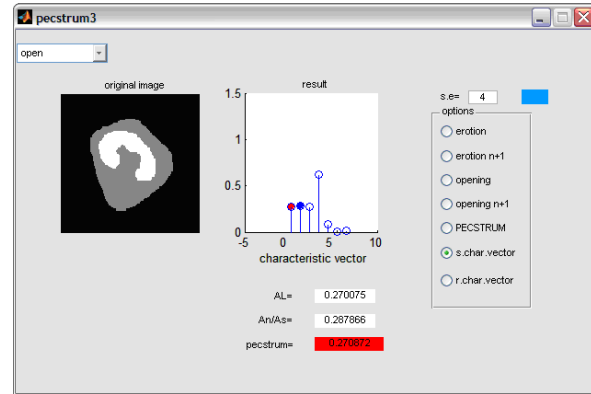
4.2 Pattern classification

To test the performance of the proposed feature vector we used 4 different classifiers based on Euclidean distance, K-nearest neighbor (KNN), Feed-forward Neural Networks (FFNN) and Support Vector Machines (SVM) respectively.

All classifiers were built using Matlab V 7.0. The FFNN was trained using the Matlab function TRAINLM (Levenberg-Marquardt backpropagation). By experimentation, FFNN architectures of (9-15-6) and (9-19-6) were found to fit best the data. SVM was extended to multi-class classification using the method proposed by [15] known as DAGSVM (Directed Acyclic Graph Support Vector Machine). Matlab function svmtrain() was used as a nucleus for the implementation of DAGSVM. The KNN was tested for k=1, 3 and 5.



(a) Original image



(b) Plot of composed feature vector for (a)

Figure 3. An example of a feature vector.

Table 1. Mean and variance of composed feature vectors for the training data.

	A_L	R_n	P(1)	P(2)	P(3)	P(4)	P(5)	P(6)	P(7)
Leukocyte type:									
Myeloblast: Mean	0.1860	0.8347	0.8156	0.0726	0.0371	0.0124	0.0084	0.0074	0.0049
Variance	0.0203	0.0643	0.1005	0.0516	0.0162	0.0089	0.0062	0.0058	0.0031
Promyelocyte: Mean	0.2939	0.5075	0.841	0.0615	0.0361	0.0234	0.0114	0.0073	0.0046
Variance	0.0331	0.0029	0.0642	0.0158	0.0089	0.0183	0.0058	0.0046	0.0028
Myelocyte: Mean	0.2105	0.4984	0.9234	0.0336	0.0198	0.0091	0.0026	0.0026	0.0025
Variance	0.0321	0.0114	0.0254	0.0132	0.0049	0.0016	0.0018	0.0021	0.0003
Metamyelocyte: Mean	0.2048	0.4629	0.5170	0.3760	0.0678	0.0234	0.0070	0.0042	0.0026
Variance	0.0384	0.0136	0.0507	0.0751	0.0249	0.0035	0.0007	0.0008	0.0017
Band: Mean	0.2127	0.2028	0.5190	0.3510	0.1210	0.0088	0	0	0
Variance	0.0226	0.0047	0.1691	0.1063	0.0601	0.0026	0	0	0
PMN: Mean	0.1796	0.1736	0.3893	0.4186	0.1573	0.0346	0	0	0
Variance	0.0198	0.0447	0.0179	0.0099	0.0398	0.0119	0	0	0

Three-fold cross validation was used to obtain the classification results shown in Table 2. Best results were obtained by a FFNN with 19 hidden nodes. A similar work [14] reported a classification rate around 77%. However, a fair comparison is not possible due to differences in the databases used.

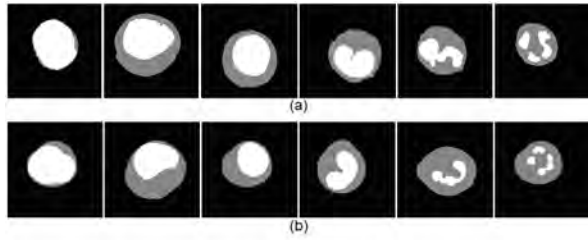


Figure 4. Examples of patterns for each class, from left to right: Myeloblast, promyelocyte, myelocyte, metamyelocyte, band, and polymorphonuclear leukocytes (PMN). (a) Real images. (b) Artificial images.

Table 2. Three-fold cross validation

Classifier	Classification Rate
Euclidean Distance	77.7%
K-NN with K= 1	70.4%
K-NN with K= 3	72.2%
K-NN with K= 5	70.4%
FFNN with 19 hidden nodes	87.6%
FFNN with 14 hidden nodes	84.9%
DAGSVM	71.6%

5. Conclusions and perspectives.

A composed feature vector based on morphological information for classification of hand-segmented leukocytes images was presented. This feature vector was constructed with the morphological *pecstrum* as the central part, and some geometric considerations.

The composed feature vector shows very good attributes to reflect the evolution in time of the white blood cells. The proposed feature extraction was tested using 4 different classifiers based on Euclidean distance, K-nearest neighbor (KNN), Feed-forward Neural Networks (FFNN) and Support Vector Machines (SVM) respectively. Three-fold cross validation was used to get the classification rates for each classifier. The best result was obtained by a FFNN

with 19 hidden nodes, getting 87.6% of corrected classification.

The reported work is part of a ongoing project on the cytology of human bone marrow images. The tools described in this paper will be used as the base for a second part in the project, oriented to the differential counting of white blood cells, and its use in the diagnosis and follow up of several pathologies, under the supervision of medical experts.

6. References

- [1] G. Bogdanovic, D. Jakimov, B. Stojiljkovic, V. Jurisic "The cell growth, morphology and immunocytochemistry of novel cell line established from a bone marrow of the patient with therapy-related myelodysplastic syndrome", *Medical Oncology*, Vol. 24, No. 4, 2007.
- [2] Navarro, M.A.Ruiz, A. Cabello, R. Collado, R. Ferrer, J. Hueso, J. Martínez, A. Miguel, M.T. Orero, P. Pérez, A. Nolasco, F. Carbonell, "Classification and scoring systems in myelodysplastic syndromes: A retrospective analysis of 311 patients", *Leukemia Research*, Vol. 30 , No. 8 , pp. 971 - 977, 2006.
- [3] T.C. Bortolheiro, "Morphological classification of myelodysplastic syndromes", *Rev. Bras. Hematol. Hemoter*, Vol. 28, No.3, pp. 194-197, 2006.
- [4] Bennet J., "Classification of acute leukemias and the myelodysplastic syndromes", *International Journal of Hematology*, vol. 72, pp.131-133, 2000.
- [5] Mayumi-Ushizima, D. and Costa-Rosatelli, M. "E-Learning in Medical Diagnosis," *XVIII Brazilian Symposium on Computer Graphics and Image Processing SIBGRAPI* 2005. Natal – RN- Brazil
- [6] J. Goutsias, L. Vincent, D. S. Bloomberg , "Mathematical morphology & its applications to image & signal processing". *Computational imaging & vision series*, Vol 18, Lavoisier Libraire, 2000.
- [7] Ledda A., J. Quintelier, P. Samyn, P. D. Baets, W. Philips, "Quantitative image analysis with mathematical morphology", *Proceedings of ProRISC 2003*, Veldhoven, The Netherlands: 399-406, 2000.
- [8] Ledda and W. Philips, "Majority Ordering and the Morphological Pattern Spectrum", *ACIVS, Proceedings Lecture Notes in Computer Science 3708*: 356-363, 2005.
- [9] P. Maragos, "Pattern spectrum and multiscale shape representation," *IEEE Transactions on Pattern Analysis and Machine Intelligence*, vol. 11, pp. 701-716, 1989.
- [10] L. Yunpeng, L. Fangcheng, L. Chengrong, "Pattern recognition of partial discharge based on its pattern

spectrum”, *International Symposium on Electrical Insulating Materials, Conference Proceedings*, June 5-9, Kitakyushu, Japan, 2005.

[11] Valderrabano A., D. Baez, J.M. Ramírez, “Pattern Recognition in Automotive Plates”, *Proceedings of the Midwest Symposium on Circuits and Systems*, University of Notre Dame, pp. 56-59, august 10-12, 1998.

[12] Asano A., “Texture analysis using morphology pattern spectrum and optimization of structuring element”, *Proc. 10th International Conference on Image Analysis and Processing*. 1999.

[13] Ledda A., P. Samyn, J. Quintelier, P. De Baets, and W. Philips, “Polymer Analysis with Mathematical Morphology”, *IEEE Benelux Signal Processing Symposium, Conference Proceedings*, Hilvarenbeek, The Netherlands, pp. 87-92, 2004.

[14] N. Theera-Umpon, P.D. Gader, “Counting white blood cells using morphological granulometries”, *Journal of Electronic Imaging*, Vol. 9, No. 2, pp. 170-177, 2000.

[15] Platt, J., Cristianini, N., Shawe-Taylor, J. “Large Margin AGS for Multiclass Classification.” *Advances in Neural Information Processing Systems*, 12 ed. S.A. Solla, T.K. Leen and K. R. Muller, MIT Press, 2000

In Search of an Optimal Set of Images for Uncalibrated Photometric Stereo

Felipe Hernández-Rodríguez and Mario Castelán

Centro de Investigación y de Estudios Avanzados del I.P.N.

Grupo de Robótica y Manufactura Avanzada

Carretera Saltillo-Monterrey km 13.5 Ramos Arizpe 25900, Coahuila, Mexico.

{felipe.hernandez, mario.castelan}@cinvestav.edu.mx

Abstract

In this paper, we draw on the ideas of previous work done by Drbohlav and Chantler [5], where a theoretical study on the optimal lighting configuration for Photometric Stereo was developed. They demonstrated that the known n light source azimuth angles should be equally spaced over $360/n$ degrees and the n light source elevation angles should be constant. In this paper, our aim is to develop a method to find a set of images that best satisfies this lighting configuration, assuming that light source directions are unknown. To this end, we estimate lighting coefficients using a harmonic spherical illumination model and a reference surface. Once the set of light source directions is estimated, we select those images that best satisfy the optimal lighting configuration. Experiments on face datasets show that the selected images are the most suitable for the purposes of calculation of height maps through surface integration.

1 Introduction

Regardless of representation (depth, height, derivatives), 3D shape provides useful information for solving important issues in computer vision and visualization such as changes in illumination and pose. Range-scanners [8, 3, 4], stereoscopic cameras [13] and structured light sensors [2, 12] offer ways in which surface models can be acquired. A common feature between these methods is the use of special devices to achieve surface reconstructions.

The Photometric Stereo Method (PSM) [14] computes local surface orientations and reflectance at each pixel using still images captured under different illumination directions with a fixed camera view. In general, the intensity at a pixel observing a certain surface patch will depend on the orientation of the patch with respect to the viewing and illumination directions. Ideally, PSM is more appealing than the above methods, since only the direction of the light source needs to be manipulated in order to grab differently illumi-

nated images. In practice, however, knowledge of the light source direction implies accurate calibration and very special illumination settings. The most common example is the wired illumination mesh used in [7], where a light source was located on each vertex of the mesh.

On the other hand, the uncalibrated photometric stereo variant¹ does not assume any knowledge of the light source directions. It only requires a $k \geq 7$ [7] number of images containing enough information to concentrate illumination variations in a linear model, i.e., using techniques such as Principal Component Analysis (PCA) or Singular Value Decomposition (SVD) on the k -column matrix of produced intensities. In [7] it has also been shown that the first three basis vectors obtained through the diagonalization of this matrix approximates the surface normals of the scene. The accuracy of this approximation is intimately related to the lighting configuration.

Although uncalibrated PSM seems appealing since no information about the light source directions is required, in practice, an optimal set of images is needed to guarantee an appropriate surface integration from the obtained surface normals. In order to address this problem, in this paper, we draw on the ideas of previous work done by Drbohlav and Chantler [5], where a theoretical study on the optimal lighting configuration for PSM was developed. They demonstrated that the known n light source azimuth angles should be equally spaced over $360/n$ degrees and the n light source elevation angles should be constant.

In this paper, our aim is to develop a method to find a set of images that best satisfies this lighting configuration, assuming that light source directions are unknown. To this end, we estimate lighting coefficients using a harmonic spherical illumination model and a reference surface. Once the set of light source directions is estimated, we select those images that best satisfy the optimal lighting configuration. Experiments on face datasets show that the selected

¹In this work, the term “uncalibrated” means that the light source directions are unknown. We do not consider other image formation aspects such as light source intensity, inter-reflections, albedo, BRDFs, etc.

images are the most suitable for the purposes of calculation of height maps through surface integration.

The organization of the paper is as follows: in Section 2, we present preliminary concepts related to the proposed method, such as surface representation, lighting recovery and PSM; Section 3 presents the iterative algorithm used to find the optimal set of images for uncalibrated PSM; Section 4 presents results on real images from both well known databases and some acquired with a simple lighting setting; finally, conclusions and future work are given in Section 5.

2 Preliminary concepts

In this section, we will introduce preliminary concepts related to the proposed method. In subsection 2.1, we revise vectors, angles and surface orientation. Subsection 2.2 describes harmonic spherical reflectance and light source direction estimation concepts. Finally, the photometric stereo problem is explained in subsection 2.3.

2.1 Vectors, angles and surface orientation

Information about a surface that is intermediate between a full 3D representation and a 2D projection onto a plane is often referred to as a 2.5D surface representation [10]. Surface orientation is one of the most important 2.5D representations. For every visible point on a surface, there exists a corresponding orientation which is usually represented by either surface normal, surface gradient or the azimuth and zenith angles of the surface normal vector.

In contrast to height data, directional information cannot be used to generate novel views in a straightforward way. However, given the illumination direction and the surface albedo properties, then surface normal directions play the central role in the surface radiance generation process. This is of particular interest in image-based analysis since light-source effects are responsible for more variability in the appearance of facial images than changes in shape or identity [11].

Let $E(x, y)$ be an image over a defined domain $\Omega \in \mathbb{R}^2$. Let $Z(x, y)$ be its corresponding height surface. The *surface gradient* is based on the directional partial derivatives of the height function Z ,

$$p = \frac{\partial Z(x, y)}{\partial x} \quad \text{and} \quad q = \frac{\partial Z(x, y)}{\partial y}. \quad (1)$$

The set of first partial derivatives of a surface is also known as the gradient space. This is a 2D representation of the orientation of visible points on the surface.

The *surface normal* is a vector perpendicular to the plane tangent to a point of the surface. The relation between surface normal and surface gradient is given by

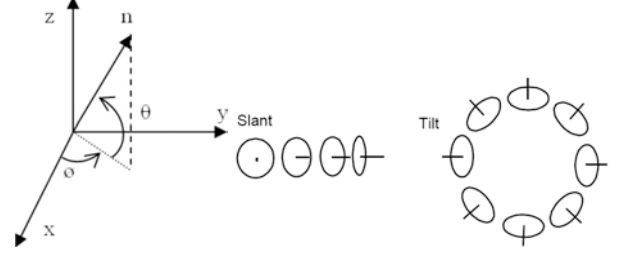


Figure 1. The azimuth (ϕ) and elevation (θ) angles of a surface normal (left) and the visual interpretation of the slant and the tilt (middle and right).

$$(n_x, n_y, n_z) = \frac{(p, q, -1)}{\sqrt{p^2 + q^2 + 1}}. \quad (2)$$

Directional information can also be expressed using the elevation and azimuth angles of the surface normals. In terms of the slope parameters, the zenith angle is $\theta = \arctan \sqrt{p^2 + q^2}$ and the azimuth angle is $\phi = \arctan \frac{q}{p}$. Here we use the four quadrant arc-tangent function and therefore $-\pi \leq \phi \leq \pi$ and $0 \leq \theta \leq \pi/2$. The elevation angle is related to inclination whereas the azimuth angle is related to orientation. We may also refer to these two angles as slant and tilt. Note how the light source vector can also be expressed by the azimuth and elevation angles (see Figure 1).

2.2 Harmonic spherical reflectance and light source direction estimation

It has been shown that, under Lambertian assumptions, known albedo $\rho(x, y)$, and ignoring inter-reflections and cast shadows, an object illuminated by an arbitrary configuration of light sources at infinity, reflectance can be modeled in terms of spherical harmonics as the following image irradiance equation,

$$\mathbf{E}(x, y) \approx \rho \mathbf{l}^T \mathbf{Y}(\mathbf{n}(x, y)) \quad (3)$$

where $\mathbf{l} = (l_0, \dots, l_{K-1})$ denote the harmonic coefficients of lighting and $Y_i(\mathbf{n}(x, y))$ include the spherical harmonic functions evaluated at the surface normal $\mathbf{n}(x, y)$, i.e.,

$$\mathbf{Y}(\mathbf{n}) = (1, n_x, n_y, n_z)^T. \quad (4)$$

Reflectance can be modeled using a first order harmonic approximation with four harmonic lighting coefficients, capturing around 85.7% of the energy in the image [1]. In practice, however, as only surface normals with

$n_z > 0$ are visible possible, around 95% of the energy can be captured.

Note how under this approximation, the lighting coefficients $\mathbf{l} = (l_0, l_1, l_2, l_3)$ contain both ambient illumination (l_0) and direction of the light source (l_1, l_2, l_3). Our aim is to estimate these coefficients from a single image. This means neither $\rho(x, y)$ nor $\mathbf{n}(x, y)$ are known. We follow Kemalmechar and Basri [9], by using a reference surface model to approximate $\mathbf{l}$.

Let $\mathbf{n}_{ref}(x, y)$ be the surface normals obtained from the reference surface model, and ρ_{ref} be the reference albedo calculated from $\mathbf{n}_{ref}(x, y)$ and the image intensity $\mathbf{E}(x, y)$. We have to solve

$$\min_{(x,y) \in \Omega} \mathbf{l} \sum (E(x, y) - \rho_{ref}(x, y)(l_0 + \hat{\mathbf{l}}^T \mathbf{n}_{ref}(x, y)))^2, \quad (5)$$

where $\hat{\mathbf{l}} = (l_1, l_2, l_3)^T$. As this is a highly over-constrained linear least square optimization with only four unknowns (the components of $\mathbf{l}$), it can be solved simply using the pseudo-inverse.

2.3 Photometric Stereo

Let $\tilde{\mathbf{n}}(x, y) = (\tilde{n}_x, \tilde{n}_y, \tilde{n}_z)$ be the surface normal at the position (x, y) scaled by its corresponding albedo $\rho(x, y)$, i.e. $\tilde{\mathbf{n}}(x, y) = \rho(x, y)\mathbf{n}(x, y)$. Let $\mathbf{N}$ be a three column matrix containing all the values of $\tilde{n}_x$, $\tilde{n}_y$ and $\tilde{n}_z$. Let $\tilde{\mathbf{l}} = (1, l_2, l_3)$ be the light source direction. A column containing all the intensity values for a determined light source direction $\tilde{\mathbf{l}}_i$ is

$$\mathbf{E}_k = \max(0, \mathbf{N}\tilde{\mathbf{l}}_k^T), k = 1, 2, \dots, K. \quad (6)$$

When $\tilde{\mathbf{l}}_K$ is known, we have a system of linear equations that can be solved by calculating the inverse ($K = 3$) or pseudoinverse ($K > 3$) of the matrix whose rows contain the light source direction vectors $\tilde{\mathbf{l}}_k$. This is the classical photometric stereo setting, when at least three light source directions are given.

The uncalibrated photometric stereo variant does not assume any knowledge of the light source directions. It only requires a $k \geq 7$ [7] number of images containing enough information to concentrate illumination variations in a linear model, i.e., using techniques such as Principal Component Analysis (PCA) or Singular Value Decomposition (SVD) on the K -column matrix of intensities $\mathbf{E}$.

The SVD of $\mathbf{E}$ produces a diagonal matrix $\mathbf{S}$, of the same dimension as $\mathbf{E}$ and with nonnegative diagonal elements, and unitary matrices $\mathbf{U}$ and $\mathbf{V}$ so that

$$\mathbf{E} = \mathbf{U}\mathbf{S}\mathbf{V}^T. \quad (7)$$

It has been shown that the first three basis vectors obtained through such techniques approximate the columns of the matrix $\mathbf{N}$. This approximation is intimately related to the lighting configuration, as we shall see in Section 4.

3 Searching a set of images with optimal lighting configuration

In this section we explain an iterative method to select a set of K images that best complies with the optimal lighting configuration derived in [5]. The iterative method can be written as follows:

1. Take a set $\mathbf{E}$, containing $M > 7$ images of a still scene illuminated by a moving light source.
2. Set $K \geq 7$ to be the number of images of the subset of $\mathbf{E}$ observing an optimal lighting configuration.
3. Build the matrix $\tilde{\mathbf{E}}$, whose k_{th} column contains the intensity values of each image in $\mathbf{E}$. This matrix will be fixed during the whole iterative process.
4. Set $\tilde{\mathbf{E}}' = \tilde{\mathbf{E}}$
5. Perform the Singular Value Decomposition of $\tilde{\mathbf{E}}'$, using Equation 7. The first three left singular vectors (columns of the matrix $\mathbf{U}$) represent an orthogonal basis that spans the set of all the linear combinations of the images in $\tilde{\mathbf{E}}'$. This orthogonal basis also approximate the surface normals of the still scene. Set the first three left singular vectors as the columns of $\mathbf{N}$.
6. Enforce integrability and obtain the surface $\mathbf{Z}$ from the normal field $\mathbf{N}$. This can be done using the Frankot and Chellappa global integration method in [6]. As suggested in [7], we used the Discrete Cosine Transform to avoid the bass-relief ambiguity.
7. Set $\mathbf{Z}$ as the reference surface and estimate the harmonic spherical lighting coefficients $\tilde{\mathbf{l}}$ for each of the columns in $\tilde{\mathbf{E}}$. Use Equation 5 for this task. Store the lighting coefficients as the rows of a lighting matrix $\tilde{\mathbf{L}}$.
8. Project the azimuth and elevation angles of the rows of $\tilde{\mathbf{L}}$ on the (x, y) -plane. Select the K projections that best satisfy the following constraints:
 - The projection of the azimuth and elevation angles on the (x, y) -plane should lie close to the n -vertex of a regular polygon. The center of the polygon is the projection closest to the origin. In other words, the light source azimuth angles should be equally spaced over $360/n$ degrees.
 - The elevation angles should be near the boundaries of $\pm 15^\circ$.

9. Record the K indexes of the columns of $\tilde{\mathbf{E}}$ corresponding to the chosen projections and build the matrix $\mathbf{T}$.
10. Set $\tilde{\mathbf{E}}' = \mathbf{T}$ and return to step 5.
11. Stop until no change occurs between the recorded indexes from the previous iteration and the current indexes.

With this method, we select images whose light source azimuth angles are uniformly distributed. The elevation angle is preferred to be close to the $\pm 10^\circ$. Using a harmonic spherical approximation of the lighting coefficients from images of the Yale B database, we found the $\pm 10^\circ$ variation in elevation to be the most adequate to concentrate relevant intensity information. This is explained in more detail in the next section.

4 Experiments

For experimental validation, we worked with a number of real world face images. These images are drawn from the Yale B database [7]. In the images, the faces are in the frontal pose and were illuminated by a point light source varying from 0 to 2π radians azimuth and from 0 to $\pi/2$ radians elevation. There is a total of 64 images per subject, per pose. These images have been acquired using a special meshed sphere, whose vertex correspond to a specific light source azimuth and elevation angles. Since the light source direction is known, we can easily find the optimal subset within the database that best satisfies the constraints explained in the former section.

We illustrate this idea with Figure 2. Here, for each subject, we calculated the set of surface normals through the SVD of the $K = 7$ images that best satisfy the set of optimality constraints. With this normal field, we integrate a reference surface. Using this surface, light coefficients estimation is achieved for each of the 64 images. This procedure is repeated so as to estimate all the lighting coefficients for the remaining subjects. The average projection of the azimuth and elevation angles for the ten subjects is shown as a cloud point in Figure 2. There are some interesting features to note from the figure. Firstly, the point distribution appears to have a concentric shape. Markers near the origin mean a decrease in elevation, while markers far from the origin indicate elevation angles close to $\pi/2$ radians. Secondly, the dark circles represent the optimal lighting configuration, which means that their corresponding images are the most appropriate set for the purposes of surface reconstruction from uncalibrated PSM. The stars represent a random configuration near the optimal one. Thirdly, the optimal set of images is located within the ranges of $\pm 10^\circ$ elevation (± 0.2 in the diagram). Finally, the estimated distribution presents a reasonable accordance with the ground-truth illumination setting presented in [7], which suggests

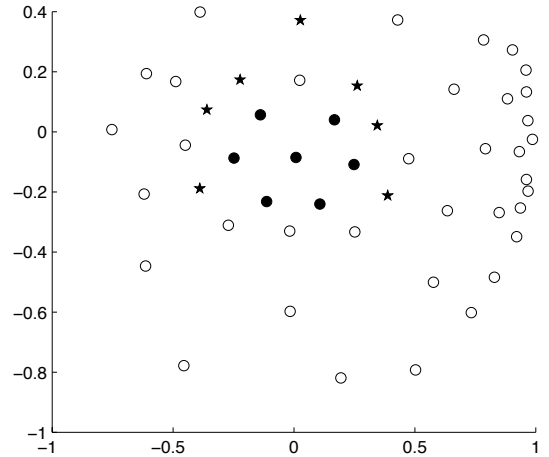


Figure 2. Projection of the azimuth and elevation angles of the light source direction, for the average of the ten Yale B database subjects. Dark circles represent the optimal lighting configuration. Stars represent a random configuration near the optimal one.

accuracy in the lighting coefficients estimation method of subsection 2.2.

We take this analysis further in Figure 3, where we show, respectively in the upper and lower rows, the subsets of $K = 7$ images corresponding to the dark circles and stars of Figure 2. Although an increase of the light source elevation angle is slightly noticeable in the second row, there is no evidence to visually determine the direction of the light source for each image. This is important since it justifies the use of special illumination estimation methods (i.e., the one explained in subsection 2.2) before the incapability of the human visual system to perform such tasks.

It is important to note that small changes in elevation can cause noticeable changes in the calculated orthogonal basis and therefore in the integrated surface. This effect is clear in Figure 4. The figure shows, from left to right, orthogonal basis w.r.t. x , orthogonal basis w.r.t. y , orthogonal basis w.r.t. z and surface integration using these basis. The first row shows results obtained from the optimal lighting configuration. Similarly, the second row shows results obtained from a random configuration near the optimal. Although there is no visible difference between both sets of basis, the quality of the reconstructed surface is greatly improved when the optimal set of images (first row) is used. This effect is complementarily shown in Figure 5. In the figure, we present the surface integration results from the optimal subset of images (upper row) and from the random



Figure 3. Input image analysis. The subsets of $K = 7$ images corresponding to the dark circles and stars of Figure 2 are shown in the upper and lower row of this figure, respectively.



Figure 4. Analysis of the obtained basis and integrated surfaces. The first row shows the orthogonal basis and surface integration obtained from the optimal lighting configuration (dark circles in Figure 2). Similarly, the second row shows the orthogonal basis and surface integration obtained from a random configuration near the optimal (stars in Figure 2).

subset near the optimal (lower row) for other subjects of the Yale B database. Again, quality improvement in surface integration is considerably visible in the upper row.

We finally present results on data acquired under non-controlled light. We present two cases where 50 snapshots of the subjects were gathered using only a desk lamp arbitrarily moved by a human. The images were taken afternoon, in an open windows area, when sunlight was not too bright to cancel out the effects of the artificial light. The initial reference surface was obtained using the whole set of 50 images. We then applied SVD to obtain surface normals and integrated the reference surface. We commence the experimental analysis with Figure 7, where the point distribution from azimuth and elevation angles, for the last iteration, is shown, from left to right for subject (a) and subject (b). The dark circles represent the optimal set of 7 images. Note that the final distribution is very different from the es-



Figure 5. Surface integration analysis. The figure shows the surface integration results from the optimal subset of images (upper row) and from the random subset near the optimal (lower row) for other subjects of the Yale B database.



Figure 6. Experiments with non-controlled light. The rows in the figure show the best set of images for surface reconstruction, after the last iteration of the algorithm.

timated using the Yale B database. This reflects the nature of the non-controlled lighting setting and the randomness of the moving light trajectory.

The selected series of images after the last iteration, for both cases is shown row-wise in Figure 6. Here, we can notice uncontrolled lighting conditions (unlike the Yale B database), since the different subjects present no similar illuminations. For example, subject (b), shown in the lower row seems to have the light source closer to his face than the subject (a). This effect can also be seen in Figure 8. The figure is divided into two panels. The upper and lower panels respectively show results on subject (a) and subject (b). The first row of each panel presents the orthogonal basis obtained after the last iteration of the algorithm. From the visual analysis of the basis we can notice that errors due to movements of subject (b) are propagated as instabilities in the surrounding areas of the mouth, nose and eyes. For the second row, each panel presents two surfaces. Each surface corresponds, from left to right, to the integrated surface from the first (using the whole set of 50 images, initial reference surface) and last (using the selected 7 images) iteration

of the algorithm. Note how, for subject (a), the first iteration surface seems to be asymmetrically elongated while the last iteration surface preserves symmetry better. This suggests that the snapshots obtained from subject (a) present more adequate lighting conditions than those obtained from subject (b), where we can see serious instabilities in the first iteration surface, which seem to be overcome in the last iteration.

5. Conclusions

We have drawn on the ideas of previous theoretical work by Drbohlav and Chantler [5], where a study on the optimal lighting configuration for PSM was developed. We have proposed an iterative method to select an optimal subset of images for the purposes of surface integration from a PSM database. Our method assumes no knowledge on the light source direction or illumination conditions and have been tested in real imagery. Experiments suggest that it is possible to obtain accurate PSM surface integrations by selecting images whose lighting azimuth angles observe a gaussian distribution as well as lighting elevation close to $\pm 15^\circ$. Future work includes the design of optimal lighting trajectories driven by a robotic arm which are consistent with the constraints considered in this paper.

References

- [1] R. Basri and D. W. Jacobs. Lambertian reflectance and linear subspaces. *IEEE Transactions on Pattern Analysis and Machine Intelligence*, 25(2):218–233, 2003.
- [2] J. Batlle, E. Mouaddib, and J. Salvi. Recent progress in coded structured light as a technique to solve the correspondence problem: a survey. *Pattern Recognition*, 31:963–982, 1998.
- [3] P. Besl. Active optical range imaging sensors. In *Advances in Machine Vision*, pages 1–63, 1989.
- [4] F. Blais. A review of 20 years of range sensor development. In *Proc. Videometrics VII, SPIE*, pages 62–76, 2003.
- [5] O. Drbohlav and M. Chantler. On optimal light configurations in photometric stereo. In *Proc. ICCV 2005*, pages 1707–1712, Oct 2005.
- [6] R. Frankot and R. Chellappa. A method for enforcing integrability in shape from shading algorithms. *IEEE Transactions on Pattern Analysis and Machine Intelligence*, 10:438–451, 1988.
- [7] A. Georgiades, D. Belhumeur, and D. Kriegman. From few to many: Illumination cone models for face recognition under variable lighting and pose. *IEEE Transactions on Pattern Analysis and Machine Intelligence*, pages 634–660, 2001.
- [8] R. Jarvis. A perspective on range finding techniques for computer vision. *IEEE Transactions on Pattern Analysis and Machine Intelligence*, 5(2):122–139, 1983.
- [9] I. Kemelmacher and R. Basri. Molding face shapes by example. In *Proc. European Conference in Computer Vision*, 2006.
- [10] D. Marr. *Vision: A Computational Investigation into the Human Representation and Processing of the Visual information*. Freeman, 1982.
- [11] Y. Moses, Y. Adini, and S. Ullman. Face recognition: the problem of compensating for changes in illumination direction. In *Proc. European Conference on Computer Vision*, pages 286–296, 1994.
- [12] D. Scharstein and R. Szeliski. High-accuracy stereo depth maps using structured light. In *IEEE Proceedings on International Conference on Computer Vision and Pattern Recognition*, 2003.
- [13] M. Starks. Stereoscopic imaging technology: Review of patents and literature. *International Journal of Virtual Reality*, 1(2):2–25, 1995.
- [14] R. J. Woodham. Photometric method for determining surface orientation from multiple images. *Optical Engineering*, 19(1):139–144, 1980.

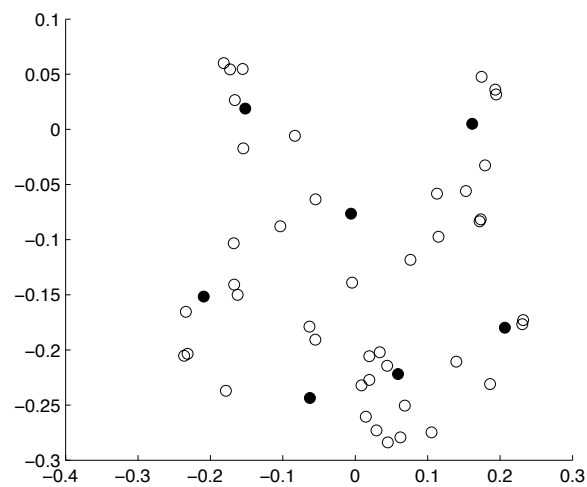


Figure 7. Experiments with non-controlled lighting. Final point distribution is shown respectively in the upper and lower diagrams, for subject (a) and subject (b). Dark circles represent the optimal set of images.

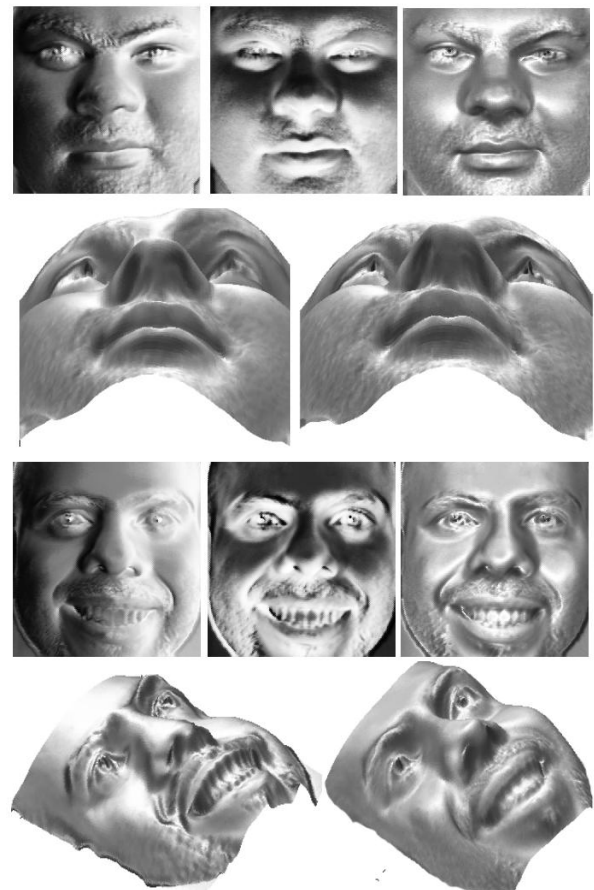


Figure 8. Experiments with non-controlled light. The figure is divided into two panels. The upper and lower panels respectively show results on subject (a) and subject (b).

Robotics, Planning, and Scheduling

Geometric Tools for Perspective Taking for Human-Robot Interaction

Luis Felipe Marin-Urias, E. Akin Sisbot, Rachid Alami
LAAS-CNRS ; Université de Toulouse ;
7 avenue du Colonel Roche, F-31077 Toulouse, France

Abstract

Psychological studies on human interactions made surface the notion Perspective Taking. This notion, referring to reason on other persons' points of view, eases the communication between interacting individuals. For a robot interacting with people, we believe that Perspective Taking can play an important role that will allow the robot to understand, reason and act according to human's point of view thus allowing an efficient communication and cooperation.

In this paper, we propose a group of algorithms to evaluate and generate robot configurations that are not only collision free but also obeying HRI constraints by reasoning explicitly on human's perspective.

1. Introduction

One of the challenges of human-robot interaction (HRI), is the environment sharing and spatial placement between the robot and the human. This, added to the problem of understanding human intentions to improve the robot's cognitive capabilities in order to result a better interaction. For these purposes, robot has to adopt different human behaviors, like perspective taking.

The notion of *perspective taking* comes from psychological studies on human to human interactions. It refers essentially to the fact of reasoning from other persons point of view. It is also interpreted as taking its own perspective from a different place on the space by applying what is called a *mental rotation* to perceive the environment from different place. These sets of actions are used by humans in their everyday lives, and are intended to ease communication between individuals and to help to have shorter and faster interactions.

Perspective taking can be used by the robot to generate configurations to approach human or to compute a geometric way to place itself where it can perceive an object referred by a person. The introduction of perspective taking concepts on robot's spatial reasoning for placement is called here *perspective placement*.

In this paper, we introduce a group of algorithms that introduces perspective taking to the geometrical reasoning of the robot. In section 2, we present the foundations of perspective taking as well state of the art approaches using this notion. Section 3 introduces PerSpective Placement (PSP), a system that generates and evaluates robot configurations according to human's perspective. Section 4 and 5 illustrates simulation and real world results of this system applied in various scenarios. Finally, the section 6 concludes this paper and gives perspectives on perspective taking in robotics.

2. Related work

Although human-robot interaction is a very active research field, there is no extensive amount of research on perspective taking on robotics.

Teversky et al. [1] [2] show the effectiveness of changing perspectives between persons in face-to-face communication scenarios for spatial descriptions. They conclude that the perspective switching is more effective than trying to avoid it.

Akerman [3], on the other hand, explains how a person can estimate his relative position in the world by 3D mental transformation of an object or of an entire place. As mental rotation and perspective taking are important for interacting with humans, they are often taken into account in 1) computer graphics for simulating human-like view [4][5], 2) in virtual reality for home or car design [6] [7], 3) in human training [8] and 4) in interface design for human-computer interaction [9].

Mental rotation has also found its place in robotics, precisely in mobile robot simulators [10], where a virtual environment simulates sensor data for the robot. With this, researchers can test different robot programs without moving real robots. Next best view problems are also applications where mental rotation is calculated in order to obtain a position to construct 3D models of objects [11] [12] or to obtain an automatic surface acquisition [13].

Perspective taking has begun to be used in HRI research field in the last years. Richarz et al.[14] use the area in front

of a human in order to obtain pointing places so that the robot can interact with him. Trafton et al. show in [15] and [16] a robot system that uses geometrical reasoning in perspective taking to take decisions about human reasoning. They prove how human-human interaction helps if it is applied to human-robot interaction. [17] and [18] describe in a 3D simulated environment using perspective taking to help a robot with the learning process, the authors establish some principles in perspective taking for HRI, and treat the problem on high level layer, but don't take into account robot motion to change its own perspective.

Nevertheless, to our knowledge mental rotation and perspective taking have not been used for a mobile robot who is in a close interaction with humans. In this article we present the importance of these characteristics in finding robot configurations to place itself.

3. PerSpective Placement (PSP) for HRI

As in every robot motion planning, the robot has to find a continuous path between its actual configuration (starting point) and its final configuration (destination point). Perspective Placement (PSP) is concerned by this last part of motion planning.

We define perspective placement as a set of robot tasks to find, validate and evaluate robot configurations in a human shared environment, performing perspective taking and mental rotation in order to interact and ease communication with humans.

3.1. PSP for Humans

To interact with human, the robot has to find how to place itself in a configuration where it has direct contact with this person. This constraint helps to reduce search space to find such a point. This search can be subdivided in two phases: first, the robot has to find positions that belong to the human field of view represented as a semicircle area in front of human (with preference on the attentional field of view) and; second, it has to validate positions obtained for visual contact, preventing big visual obstructions from blocking robot perception.

The area in human attentional field of view, called here as "Interaction Area", is defined as the zone in front of the person by the angle and by the radius:

$$\alpha_{view} \mid 0^\circ \leq \alpha_{view} \leq 180^\circ$$

$$Rad \mid Rad_{min} \leq Rad \leq Rad_{max}$$

α and Rad are defined by the characteristics of the interaction (e.g. proximity needed for just visual interaction or handing an object), by robot sensor capabilities and by human preferences.

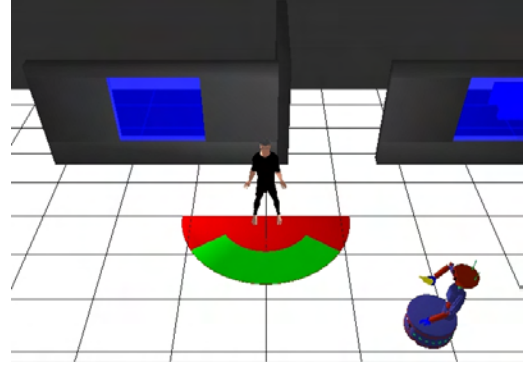


Figure 1. The interaction area is marked as the band in front of the human.

Once defined this area, illustrated in figure 1, a set of points around the person are generated and will be chosen in order to ensure these properties:

- **Collision Free:** Robot in this position must not be in collision neither with objects and persons nor with itself.
- **Sensor Oriented:** Selected sensors must be oriented towards the human in order to perceive it.
- **Without Visual Obstructions:** In sensor's acquisition, human has to be perceived depending on the interaction task.
- **Human-Like Positioning:** Based on user studies [19] of robot-human spatial placement, robot has to find spatial formation with the person in a human acceptable way.
- **Minimal Cost:** Robot should find a position that minimizes the cost based on human-robot distance and on human's comfort [20] [21].

To determine what is perceived by a camera, we use 2D perspective projection of the 3D environment. This projection is obtained from the sensors's position when the robot is placed in desired configuration. The projection obtained is a matrix $MatP$ where the value of the position (x, y) represents one point in the object's projection image in sensor's field of view. In the figure 2, 2D projection is illustrated.

We define "Projection rate" Pr as the projection percentage of an element El (object, human or obstacle) on the environment represented in $MatP$. Pr is obtained by: $Pr(El) = \sum MatP(x, y) \mid (x, y) \in El$.

The projection rate of an element that is not projected Pr_{hidden} can be obtained with: $Pr_{hidden}(El) = Pr_{desired}(El) - Pr_{visible}(El)$ where $Pr_{visible}$ is the projection rate that considers visual obstructions (only visible

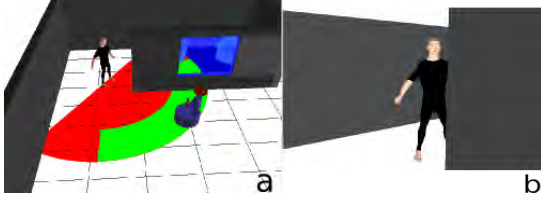


Figure 2. a) Robot positioned in the interaction area. b) Computed robot perception (2D projection)

projection). On the other hand, $Pr_{desired}$ is the relative projection obtained without considering objects in the environment (as it should look without visual obstacles). In figure 3 we can observe the difference between desired and visible relative projections.

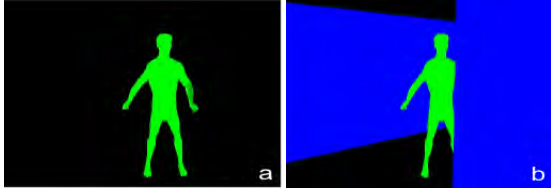


Figure 3. Relative projections, here the person is the objective and differs from other elements on the environment. a) Desired relative projection b) Visible relative projection

Objective's Ob visibility percentage, $Watch$ is determined by:

$$Watch(Ob) = \frac{Pr_{visible}(Ob)}{Pr_{desired}(Ob)}$$

Finally, to know if a point is candidate for perspective placement by: $Watch(Ob) \geq \mu$ where μ is a threshold that corresponds to a desired percentage.

In the other hand, costs for each point are, as we have introduced before, based partially on the direct distance from robot's actual position $Cost_{distance}$, and the total cost is calculated as:

$$Cost_{total}(x, y) = (Cost_{distance}(x, y) * Gain_{distance}) + ((Cost_{frontal}(x, y) + Cost_{inarea}(x, y)) * Gain_{frontal}) + (Cost_{HAMP}(x, y) * Gain_{HAMP})$$

Where $Cost_{frontal}$ is the distance from the front of the human gaze orientation, to incite a *face-to-face* human like position. $Cost_{inarea}(x, y)$ is a fixed cost depending if the

point is in the interaction area (cost is higher otherwise). $Cost_{HAMP}$ is the cost of the point calculated by the Human Aware Motion Planner (explained in detail on [20] and [21]) based on human safety and comfort. $Gain_x$ is the weight given to this criterion. An example of calculated costs is shown in figure 4 for the scenario of figure 1.

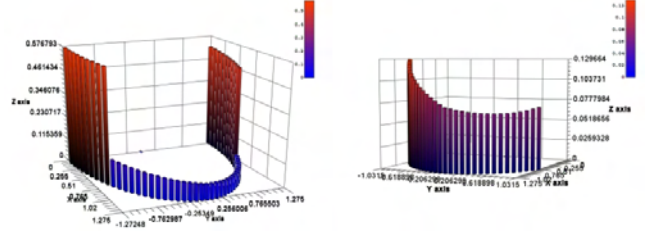


Figure 4. Computed costs of the points. a) Points in all around the field of view area, those points out of the interaction area have the highest cost. b) Points on the interaction area, lower cost are due to robot proximity and that are closer to front.

Maximal and minimal ranges of interaction area are predefined based on Hall's interpersonal distances mentioned in [19] depending on desired interaction task.

3.2. PSP for Known Objects

Objects in the environment like tables, desks, sofas, chairs, etc. All considered as fixed obstacles and are called here as "known objects". The robot knows object's model and its position in the environment.

Methodology of perspective placement for a known object, is similar to PSP for humans. Here the object's perspective is not taken into account, so the robot can place itself all around object's position. Nevertheless, robot has to take into account its own perspective to look for the best placement configuration.

In this case "Interaction area" is called "Approach area" (obviously because there is no interaction with unanimated objects). Maximal and minimal ranges are defined based only on collision avoidance and robot sensor limits. For a given point, costs are calculated with:

$$Cost_{total}(x, y) = (Cost_{distance}(x, y) * Gain_{distance}) + (Cost_{HAMP}(x, y) * Gain_{HAMP})$$

An example of costs function for a known object is illustrated in figure 5. Configuration search starts at the least expensive point that contains all the properties mentioned above.

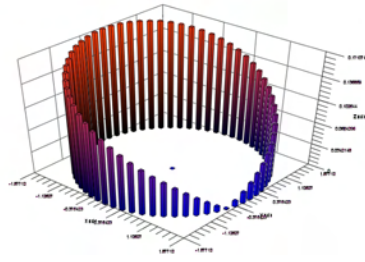


Figure 5. Costs of the points in modeling area around an object.

3.3. PSP for Non-visible or Unknown Objects

Sometimes there are manipulable objects in the environment that are small. But they are not always at the same place and they can be hidden by another object. The object referenced by the human can be unperceivable to the robot (e.g. too far from robot sensors to create a model or know the exact position and orientation of the referenced object).

Here, our approach to this problem is to create a simple geometrical object, called “search sphere”, that could be set on possible places like the one referenced by human or on surfaces of known objects (e.g. on the top of a table). Discovering the referred object can be interpreted as seeing the search sphere.

Due to the search sphere’s shape and position, its approaching area and its costs functions are placed around the sphere as it is in for known objects. Maximal and minimal approaching ranges are defined by the specifications of the task.

4. Simulation Results

Our perspective taking system is implemented in C and integrated and tested within the Move3D [23] software platform developed at LAAS-CNRS.

Fig. 6 illustrates two similar situations where the robot has to interact with a person, in the first scenario (fig. 6-a) there is no obstacles between human and robot, and on the second one (fig. 6-b) we can see an obstacle between them but it doesn’t cause a visual obstruction for the robot, the interaction zone is also further from human.

In the scenario illustrated in figure 7, a desk prevents the robot to see the human that is seated behind this visual obstruction. Robot finds a position where it can see the person whom it wants to interact.

Figure 8 gives examples of how PSP can find a configuration for known object (a furniture) in two different situations. In the first case with a person is blocking the obstacle

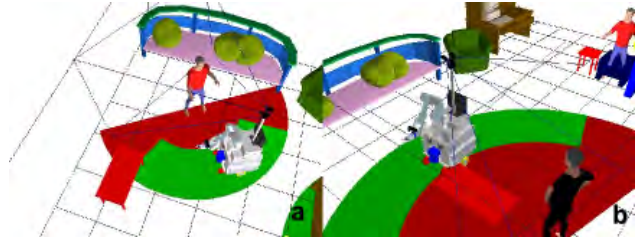


Figure 6. Simple case: Results of PSP for Humans. Robot have found a point near the human where it can see him

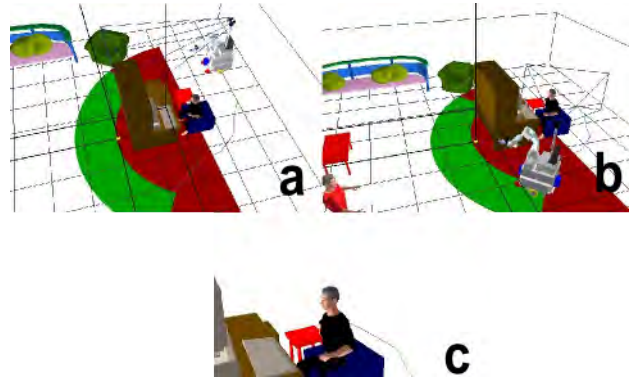


Figure 7. Results of PSP for Humans with visual obstacles. a) Initial scenario with a human is partially hidden by a desk. b) Robot final configuration where it can see de human. c) Robot’s perspective in its computed configuration

and causing a visual obstruction, here the robot finds a position on the right side of the person, by avoiding it in order to perceive the table. Note that the robot computes a final configuration where it avoids not only collisions with human or obstacles but also with visual obstacles.

On the second situation an extra person is occluding the table from the position chosen on the first case, here the robot computes a configuration on the other side of table where none of the persons is present on the environment are visually blocking this table.

5. Robot Implementation and results

The PSP system is implemented into OpenGenom [24] with the help of other modules of supervision, task planning, human detection and tracking, robot self localization, path and motion planning, etc. that forms the same architecture as described in detail on [25].

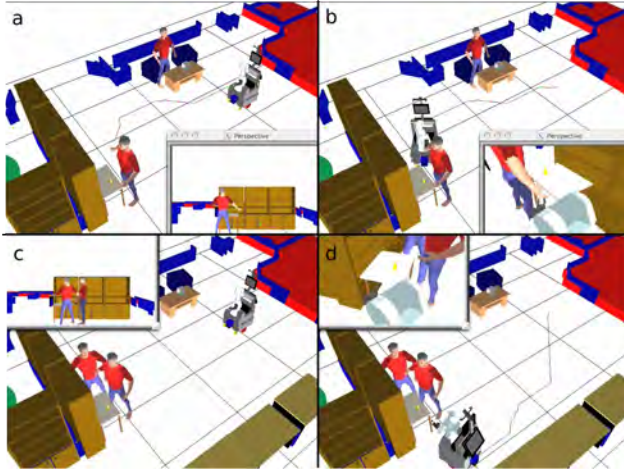


Figure 8. Approaching to the table, a) Initial position, b) Final configuration looking at the table c), d) initial and final configuration with two persons hiding the table.

The whole system has been carried into our robot *Jido* equipped of three Pentium IV processors, one Laptop with a Intel core-duo processor, two 2D SICK laser scanner (front and rear), a six DOF Mitsubishi PA-10 manipulator, 1 tilt and pan stereo camera, 1 color stereo cameras on the end axis of the manipulator, 8 sonar sensors, and three finger tactile hand.

In figure 9, a scenario is shown with a person interacting with the robot. The person indicates the robot to go to pick up an object on the table and bring it back to him (“bring me the yellow bottle”). The robot makes the respective plan with the sequence of different tasks. PSP module finds a valid configuration to execute each task that implies navigation motion of the robot.

The first configuration is placed to see the table where the bottle is located, maximum and minimum distances are set according to arm capabilities for grasping the bottle. Second configuration is found to get close to human and give him the bottle.

Another example is illustrated in 10, where an obstacle (a table) prevents the robot from several positions close to the human to bring him the bottle.

6. Conclusions and future work

In this work, we have shown how using perspective taking concepts added to motion planning techniques can be useful on human-robot interaction. Furthermore, we have presented different algorithms that take into account human and robot perspectives to determine different robot configurations, where the robot can perceive its objective (human or



Figure 9. First PSP for known objects, approaching to the table, a) Initial position, b) Final configuration looking at the table. Second, PSP for humans, c)-d) Initial and final configuration looking at human

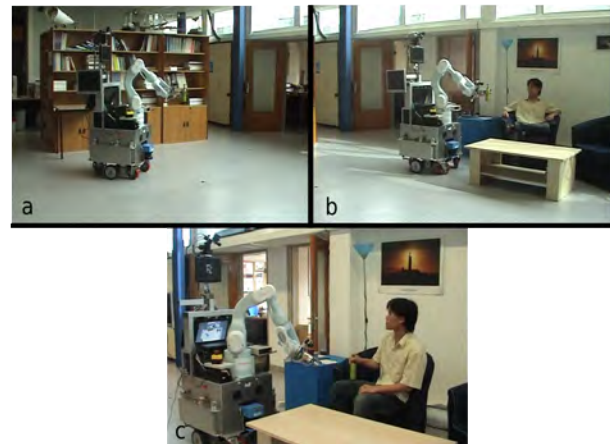


Figure 10. PSP to hand an object to a human sat in front of a table a) Initial position, b)-c) Final configuration looking at the human. The robot is perceiving human by cameras on top. The robot finds its final configuration to complete the task.

objects) without any collision and visual obstructions. This allows robot to plan its trajectory to a position where both, the human and the robot, can see each other and interact.

A possible extension of this approach is a fusion of a task planner with these geometrical tools. This could increase the precision for obtaining viable tasks. Another extension of PSP, could be on the close human-robot interaction where the robot has to reason about manipulating objects, deciding

positions on a shared space with human.

Also we plan to integrate these systems to obtain a general solver which will produce friendly and socially acceptable robot positions.

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References

- [1] B. Tversky, P. Lee, and S. Main Waring, "Why do Speakers Mix perspectives", *Spatial Cogn. Computat.*, vol. 1, 1999, pp 312-399.
- [2] Lee, P. U. and Tversky, B. "Costs of Switching Perspectives in Route and Survey Description", *Proceedings of the twenty-third Annual Conference of the Cognitive Science Society*, Edinburgh, Scotland 2001.
- [3] Edith K. Ackerman, "Perspective-Taking and Object Construction: Two keys to learning", *Constructionism in Practice: Designing, Thinking, and Learning in a Digital World*, Mahwah, New Jersey, 1996.
- [4] John P. Costella, "A Beginner's Guide to the Human Field of View", School of Physics, The University of Melbourne, November 1995.
- [5] Hansong Zhang, Dinesh Manocha, Tom Hudson and Kenny Hoff, "Visibility Culling Using Hierarchical Occlusion Map", *Proceedings of SIGGRAPH*, 1997.
- [6] Antonio Gomes de Sá and Joachim Rix, "Virtual Prototyping - The Integration of Design and Virtual Reality", *CAD Tools and Algorithms for Product Design*, 1998, 128-150.
- [7] Pascal Mueller, Peter Wonka, Simon Haegler, Andreas Ulmer, Luc Van Gool, "Procedural Modeling of Buildings", *ACM Transactions on Graphics*, volume 25, number 3, pages 614-623. 2006.
- [8] M. Alejandra Menchaca-Brandan and Andrew M. Liu and Charles M. Oman and Alan Natapoff, "Influence of perspective-taking and mental rotation abilities in space teleoperation", *HRI '07: Proceeding of the ACM/IEEE international conference on Human-robot interaction*, 2007, Arlington, Virginia, USA.
- [9] S.W. Hsu and T.Y. Li, "Third-Person Interactive Control of Humanoid with Real-Time Motion Planning Algorithm", *Proc. of IEEE International Conf. on Intelligent Robots and Systems (IROS'06)*, 2006, Beijing, China.
- [10] Josh Faust, Cheryl Simon, William D. Smart, "A Video Game-based Mobile Robot Simulation Environment", *Proc. of IEEE International Conf. on Intelligent Robots and Systems (IROS'06)*, 2006, Beijing, China.
- [11] Andrea Bottino and Aldo Laurentini, "What's NEXT? An interactive next best view approach", *Journ. of Pattern Recognition*, 2006, pages 126-132.
- [12] Y.F. Li, B. He and Paul Bao, "Automatic view planning with self-termination in 3D object reconstructions", *Journ. on Sensors and Actuators 2005*, Pages 335-344.
- [13] Richard Pito, "A Solution to the Next Best View Problem for Automated Surface Acquisition", *IEEE Transactions on Pattern Analysis and Machine Intelligence*, 1999.
- [14] Richarz, J. Martin, C. Scheidig, A. Gross, H.M., "There You Go! - Estimating Pointing Gestures In Monocular Images For Mobile Robot Instruction", *Robot and Human Interactive Communication ROMAN 2006.*, Univ. of Hertfordshire, Hatfield, UK, .
- [15] J. Gregory Trafton, Nicholas L. Cassimatis, Magdalena D. Bugajska, Derek P. Brock, Farilee Mintz, Alan C. Schultz, "Enabling effective human-robot interaction using perspective-taking in robots", *IEEE Transactions on Systems, Man, and Cybernetics, Part A*, 2005, 460-470.
- [16] Trafton, J.G.; Schultz, A.C.; Bugajska, M.; Mintz, F., "Perspective-taking with robots: experiments and models", *Robot and Human Interactive Communication ROMAN 2005*, Page(s): 580 - 584.
- [17] Cynthia Breazeal, Matt Berlin, Andrew Brooks, Jesse Gray, Andrea L. Thomaz, "Using perspective taking to learn from ambiguous demonstrations", *Robotics and Autonomous Systems*, 2006, pgs: 385-393.
- [18] Matt Berlin, Jesse Gray, Andrea L. Thomaz, Cynthia Breazeal, "Perspective Taking: An Organizing Principle for Learning in Human-Robot Interaction.", *AAAI*, 2006, Boston, Mt.
- [19] Helge Huettenrauch, Kerstin Severson Eklundth, Anders Green, Elin A. Topp, *Investigating spatial relationships in human-robot interaction*, Proc in (IEEE/RSJ) International Conference on Intelligent Robots and Systems, 2006. Beijing, China
- [20] E. Akin Sisbot and Luis F. Marin and Rachid Alami and Thierry Siméon, *A mobile robot that performs human acceptable motion*, Proc in (IEEE/RSJ) Interna-

tional Conference on Intelligent Robots and Systems, 2006. Bejjin, China

- [21] E. Akin Sisbot and Luis F. Marin-Urias and Rachid Alami and Thierry Siméon, *A Human Aware mobile robot motion planner* IEEE transactions on Robotics, vol. 23 No. 5. Oct.2007.
- [22] E. Akin Sisbot and Luis F. Marin-Urias and Rachid Alami, *Spatial reasoning for human robot interaction* Proc in (IEEE/RSJ) International Conference on Intelligent Robots and Systems, 2007. San Diego, CA. USA.
- [23] T. Siméon, JP. Laumond, F. Lamiriaux, *Move3D: a generic platform for motion planning*, 4th International Symposium on Assembly and Task Planning, Japan,2001.
- [24] S. Fleury and M. Herrb and R. Chatila *Genom: a tool for the specification and the implementation of operating modules in a distributed robot architecture* IEEE/RSJ International Conference on Intelligent Robots and Systems, IROS 1997, Grenoble, FR.
- [25] E. Akin Sisbot and Maxime Ransan and Aurelie Clodic and Rachid Alami, *Supervision and Motion Planning for a Mobile Manipulator Interacting with Humans* Proc in (ACM/IEEE) International Conference on Human Robot Interaction, 2008. Amsterdam, HL.

Exploring unknown environments with multiple robots

Abraham Sánchez L., Alfredo Toriz P.
Computer Science Department - BUAP
14 South and San Claudio, Pue. México
asanchez@cs.buap.mx
alfredot@hotmail.com

Maria A. Osorio L.
Chemical Engineering School - BUAP
18 South and San Claudio, Pue. México
aosorio@cs.buap.mx

Abstract

There are numerous studies to find efficient ways for exploring and generating maps for unknown environments. Unfortunately, most of the studies deal with single robots. In this paper, we propose a cooperative exploration strategy for mobile robots. The extension of the SRT method to the case of multiple robots is essentially a parallelization of the basic method. To this simple extension, the following functionalities were added to it: cooperation to increase the efficiency, coordination to avoid conflicts and communication to cooperate and to coordinate. We present simulation results to show the performance of this proposed technique.

1. Introduction

Generating maps is one of fundamental tasks of mobile robots. Many successful robotic systems use maps of the environment to perform their tasks. Exploration is the task of guiding a mobile robot during mapping such that it covers the environment with its sensors. Efficient exploration strategies are also relevant for surface inspection, mine sweeping, or surveillance.

Although most mobile robotic systems use a single robot that only operates in its environment, a number of researchers have considered the advantages and disadvantages of the potential use of a group of robots that cooperate for the accomplishment of a required task [2], [3], [1]. In exploration, it aims at significantly reducing the time required to complete the task. The multi-robot teams can explore unknown environments more quickly than a single robot, but a central question is how to coordinate the behavior of multiple robots, specially in an environment whose structure is not known.

A pioneering work in the multi-robot case was presented by Yamauchi [11], the robots merge the acquired information in a global grid map of the environment, from which

the frontier is extracted and used to plan the individual robot motions. In [1], the authors proposed an arbitration mechanism to negotiate robots targets by optimizing a utility function which takes into account the information gain of a particular region. A novel approach to coordinate multi-robot exploration under global uncertainty about the robot's relative start locations was presented in [8]. The key technical innovation of this approach is an adapted particle filter that allows pairs of robots to efficiently and sequentially estimate their relative locations during the exploration process. In [7], the algorithm deploys robots one-at-a-time, with each robot making use of information gathered by the previous robots to determine the next deployment location. The paper proposed by Simmons et al. [10], addresses the problem of exploration and mapping of an unknown environment by multiple robots. The mapping algorithm is an on-line approach to likelihood maximization that uses hill climbing to find maps that are maximally consistent with sensor data and odometry.

This paper presents a strategy to explore an unknown environment by multiple robots. The strategy is a parallelization of the SRT method, which was presented in [9]. The SRT method is an exploration strategy based on the random generation of robot configurations within the local safe area detected by the sensors. A data structure called Sensor-based Random Tree is created, which represents a roadmap of the explored area with an associated safe region. Each node of the SRT consists of a free configuration with the associated Local Safe Region (LSR) as reconstructed by the perception system; the Safe Region (SR) is the union of all the LSRs, while an arc between two nodes represents a collision-free path between the two configurations. The LSR is an estimate of the free space surrounding the robot at a given configuration; in general its shape will depend on the sensor characteristics but may also reflect different attitudes towards perception. An interesting alternative option to explore unknown environments was presented in [4], this strategy deals with non-holonomic constraints and is named SRT-Radial [5].

To this simple extension, the following functionalities were added to Multi-SRT: *cooperation* to increase the efficiency, *coordination* to avoid conflicts and *communication* to cooperate and to coordinate. Each robot of the team constructs a SRT, taking into account the presence of others robots by means of a suitable redefinition of its local frontier and planning its motion towards areas that have not been explored by itself, and by the rest of the team. In addition to this mechanism of local cooperation, an algorithm of simple coordination that guarantees a safe collective motion has been also considered. Once a robot has completed its SRT, it can support others robots in their expansion, this action is considered a form of global cooperation. An important characteristic of this technique is the decentralization, which can be implemented with a limited or unlimited communication range.

2. Exploration with SRT-Radial

The authors in [9] presented two techniques in their work. The first, where the LSR is a ball, performs a conservative attitude particularly suited to noisy or low-resolution sensors, and results in an exploration strategy called SRT-Ball. The second technique is confident, and the corresponding strategy is called STR-Star; in this case, the LSR shape reminds of a star. (see [9] and [4] for more details). Espinoza et al., presented in [4] an interesting extension of the SRT method, the SRT-Radial strategy proposed in this works takes advantage of the information reported by the sensors in all directions, to generate and validate configurations candidate through reduced spaces [5].

The two strategies (SRT-Star and SRT-Radial) were compared through simulations. We used the same environment to prove the efficiency of SRT-Radial over SRT-Star. Figure 1 above presents the explored regions and the safe region obtained with the SRT-Star strategy for an environment composed of two obstacles (Time = 438.86 secs and 109 nodes.). Figure 1 down shows the explored regions and the safe region with the SRT-Radial strategy (Time = 97.05 secs and 104 nodes.). The final number of nodes in the tree and the running time are much smaller with SRT-Radial than with SRT-Star.

In SRT-Star, the local safe region is a region with star form because of the union of several cones with different radii each one. The radius of the cone i can be the minimum range between the distance of the robot to the closest obstacles or the measurable maximum rank of the sensors. Therefore, SRT-Star approach must identify first, the correspondent cone of the exploration direction. On the opposite, SRT-Radial considers that the local safe region in absence of obstacles has the ideal form of a circumference, a reason that makes unnecessary the identification of the cone. In the presence of obstacles, the form of the local safe re-

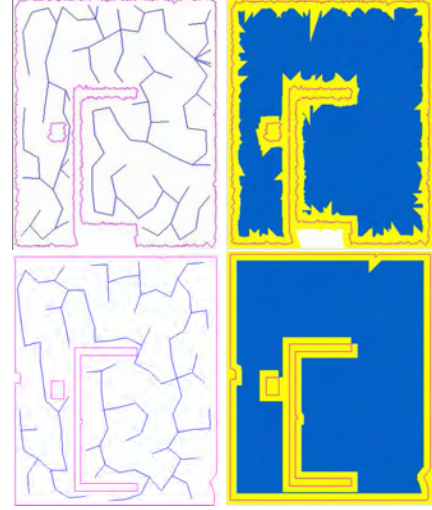


Figure 1. SRT-Star vs. SRT-Radial.

gion is deformed, and for different exploration directions, the radii lengths vary (see [5] for more details).

3. Cooperative SRT-Radial

The method of cooperative exploration based on SRT-Radial is presented under some suppositions: i) All the robots are identical, ii) The workspace is $\mathbb{R}^2$ or a connected subset of it, iii) Each robot is a polygon that can move in any direction and it also knows its configuration, iv) the robots are equipped with a sensory system, which provides the local safe region, v) the robots know its ID number and each robot can broadcast within a communication range the information stored in its memory at any time.

```

BUILD MULTI-SRT( $q_{init}$ )
1   $\mathcal{T}.init(q_{init})$ 
2  BUILD_SRT( $q_{init}, \mathcal{T}$ );
3  SUPPORT_OTHERS( $q_{init}$ );

```

Figure 2. The Multi-SRT algorithm.

The exploration algorithm for each robot is shown in Figure 2. First, the procedure BUILD_SRT is executed, i.e., each robot builds its own SRT, $\mathcal{T}$ is rooted at its initial configuration q_{init} . This procedure comes to an end when the robot can not further expand $\mathcal{T}$. Later, the robot executes the SUPPORT_OTHERS procedure, this action contributes to the expansion of the SRTs that have been built by others robots. When this procedure finishes, the robot returns to the root of its own tree and finishes its exploration. The algorithm has been inspired of the work presented in [6].

```

BUILD_SRT( $q_{init}, \mathcal{T}$ )
1   $q_{act} = q_{init};$ 
2  do
3    BUILD_AND_WAIT_GPR();
4     $S(q_{act}) \leftarrow \text{PERCEIVE}(q_{act});$ 
5     $\text{ADD}(\mathcal{T}, (q_{act}, S(q_{act})));$ 
6     $\mathcal{G} \leftarrow \text{BUILD\_GER}();$ 
7     $\mathcal{F}(q_{act}) \leftarrow \text{LOCAL\_FRONTIER}(q_{act}, S(q_{act}), \mathcal{T}, \bigcup \mathcal{T}_i);$ 
8     $q_{target} \leftarrow \text{RANDOM\_PLANNER}(q_{act}, \mathcal{F}(q_{act}));$ 
9    if  $|\mathcal{G}| > 1$ 
10      $(\mathcal{G}_f, \mathcal{G}_u) \leftarrow \text{CHECK\_FEASIBILITY}(\mathcal{G});$ 
11     if  $\mathcal{G}_u \neq \emptyset$ 
12        $q_{target} \leftarrow \text{COORDINATE}(\mathcal{G}_f, \mathcal{G}_u);$ 
13      $q_{act} \leftarrow \text{MOVE\_TO}(q_{target});$ 
14     exit  $\leftarrow (q_{act} = q_{init}) \text{ and } (\mathcal{F}(q_{act}) = \emptyset);$ 
15  while exit = 0

```

Figure 3. The BUILD_SRT procedure.

We can notice that in the algorithm described in Figure 3, perception and planning procedures are only emphasized. Let us remember that a parallel communication thread is run in each robot, we can consider two possible scenes, one where each robot broadcasts all its knowledge (also one can consider the derived from others robots) and therefore the important information for the cooperation and coordination with the others robots affects the communication range; the second scene tries to reduce the bandwidth consumption, i.e., the communication is established when it is necessary. The procedure BUILD_SRT is shown in Figure 3.

In each iteration of the BUILD_SRT, the robot uses all available information (partially collected by itself and partially gained through the communication with other robots) to identify the group of engaged robots (GER), i.e. the other robots in the team with which cooperation and coordination are adequate. This is achieved by the construction of the first group of pre-engaged robots (GPR), or robots that are candidates to be members of the GER, and are synchronized with them (BUILD_AND_WAIT_GPR). Then, the robot collects data through its sensory systems, builds the current LSR (PERCEIVE) and updates their own tree $\mathcal{T}$. The current GER can now be built (BUILD_GER). At this point the robot processes its local frontier (the portion of its current LSR limit leads to areas that are still unexplored) on the basis of $\mathcal{T}$ as well as any other tree $\mathcal{T}_i$ gained through communication and stored in its memory (LOCAL_FRONTIER).

If the local frontier is not empty, the robot generates a random configuration contained in the current LSR and headed towards the local frontier, if not, the target configuration is fixed to the node father with a backward movement (RANDOM_PLANNER). If the GER is composed only by

the same robot, the robot moves directly to its target. Otherwise, the paths advanced by the robot in the GER are checked for mutual collisions, and classified in feasible and unfeasible paths (CHECK_FEASIBILITY). If the subset $\mathcal{G}_u$ of robots with unfeasible paths is vacuum, a coordination stage takes place, perhaps, confirming or modifying the current target of the robot (COORDINATE). In particular, the movement of the robot can be banned by simply readjusting the target to the current configuration. Then, the function MOVE.TO transfers the robot to the target (when this is different from q_{act}). The loop is repeated until the condition in the output line 15 is verified: the robot is unable to expand the tree $\mathcal{T}$ (no local frontiers remaining) and therefore it has to move back to the root of its SRT.

The SUPPORT_OTHERS procedure is detailed in Figure 4, it can be divided into two major phases, which are repeated over and over again. In the first phase, the robot picks another robot to support it in its exploration, or, more precisely, another tree that helps it to expand (there may be more than one robot acting on a single tree). In the second phase, the selected tree is reached and the robot tries to expand it, tying subtrees constructed by the procedure BUILD_SRT. The main cycle is repeated until the robot has received confirmation that all the other robots have completed their exploration. In the first phase, the robot takes in a set $\mathcal{I}$, the trees belonging to $\bigcup \mathcal{T}_i$ that may require support for expansion.

4. Simulation results

This section presents simulations results for car-like robots. The strategies were implemented in Visual C++ 6.0,

```

SUPPORT_OTHERS( $q_{act}$ )
1  do
2    for  $i = 1$  to  $n$ 
3      if  $\text{OPEN\_FRONTIER}(\mathcal{T}_i) \geq \bar{F} \cdot \text{ACTIVE\_ROBOTS}(\mathcal{T}_i)$ 
4         $\text{ADD}(\mathcal{T}, \mathcal{T}_i)$ ;
5       $\mathcal{T}_s \leftarrow \text{SELECT}(\mathcal{T})$ ;
6      if  $\mathcal{T}_s \neq \text{NULL}$ 
7         $q_{act} \leftarrow \text{TRANSFER\_TO}(\mathcal{T}_s.\text{root})$ ;
8         $\text{BUILD\_SRT}(q_{act}, \mathcal{T}_s)$ ;
9         $q_{act} \leftarrow \text{TRANSFER\_TO}(\mathcal{T}.\text{root})$ ;
10 while  $\text{EXPLORATION\_RUNNING}()$ 

```

Figure 4. The SUPPORT_OTHERS procedure.

taking advantage of the MSL library's structure¹. The library GPC² developed by Alan Murta was used to simulate the sensor's perception systems. The tests were performed on an Intel © Pentium D processor-based PC running at 2.80 GHz with 1 GB RAM. Figure 5 illustrates the environments used for the simulation part. A series of experiments revealed that the approach works efficiently exploring environments almost in its totality. One can consider two possible initial deployments of the robots. In the first, the robots are initially scattered in the environment; and in the second, the exploration is started with the robots grouped in a cluster. Since the Multi-SRT approach is randomized, the results were averaged over 20 simulation runs.

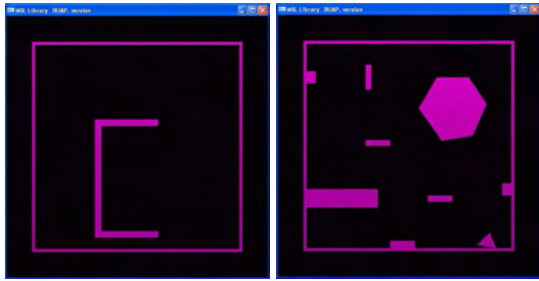


Figure 5. Environments used for the tests of the Multi-SRT.

Exploration time for teams of different cardinality are shown in Figures 6 and 7, both in the case of limited and unlimited communication range. In theory, when the number of robots increases, the exploration time would quickly have to decrease. This affirmation is fulfilled in the case of the scattered start; note however that, in the case of the clustered start, there are examples where this affirmation is not verified. We consider that an increment of the number of

evenly deployed robots corresponds to a decrement of the individual areas they must cover. In the case of a limited communication range, when the robots are far apart at the start, they can exchange very little information during the exploration process. The total travelled distance increases with the number of robots because more robots try to support the others in their expansion.

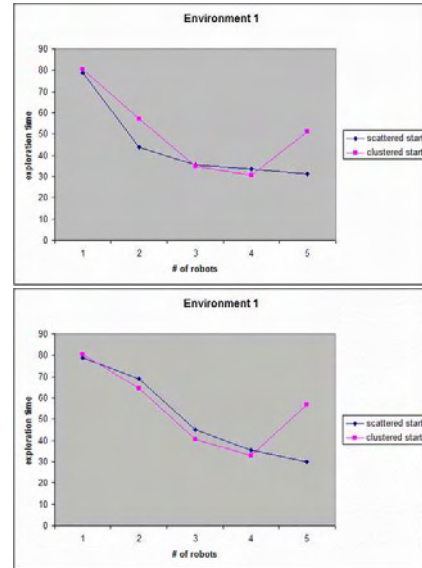


Figure 6. Environment 1 exploration with scattered and clustered start. To the left with unlimited communication range and in the right with limited communication range.

Figures 8 and 9 show the Multi-SRT and the explored region for the environment 2 with a team of 10 robots in the case of unlimited communication range. We can see the difference when the robots are evenly distributed at the start or are clustered. At the end, the environment has been com-

¹<http://msl.cs.uiuc.edu/msl/>

²<http://www.cs.man.ac.uk/~toby/alan/software/>

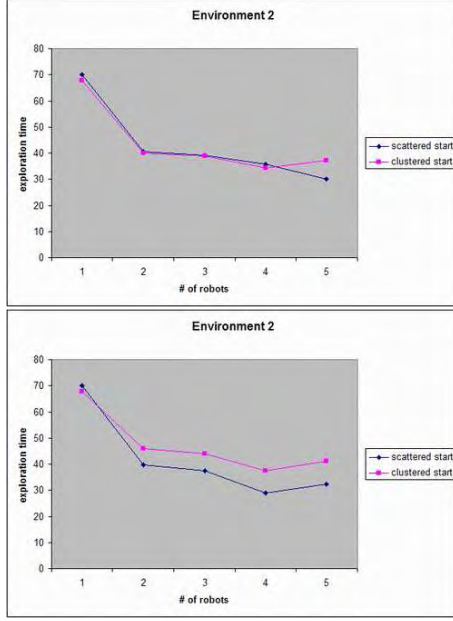


Figure 7. Environment 2 exploration with scattered and clustered start. To the left with unlimited communication range and in the right with limited communication range.

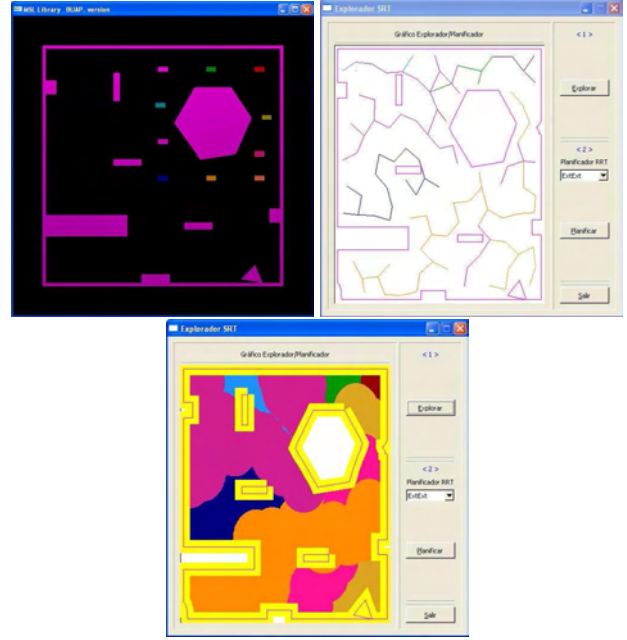


Figure 8. Environment 2: The Multi-SRT and explored regions with cluttered start.

pletely explored and the SRTs have been built. In these figures, we can observe that each robot built its own SRT and when one of them finished, this entered the support phase. If we observed the placed figure in the center of both figures 8 and 9, one can note that the trees are united, this indicates that the support phase took place. Table 1 shows the simulation results for the environment 2, we present teams of different cardinality versus exploration time for unlimited communication range case.

The local coordination procedure implemented in our work guarantees that the collective motion of the robots is feasible from the collision viewpoint. The approach does not need a central supervision. The selection of exploration actions by each robot is spontaneous and it is possible on the basis of the available information.

Single robots can not produce accurate maps like multi-robots. The only advantage of using a single robot is the minimization of the repeated coverage. However, even though repeated coverage among the robots decreases the mission's efficiency, some amount of repeated coverage is a desirable situation for better efficiency. Additionally, this better efficiency can be achieved by coordination among robots.



Figure 9. Environment 2: The Multi-SRT and explored regions with scattered start.

Table 1. Environment 2 exploration with scattered and clustered start.

# of robots	scattered start time (secs)	clustered start time (secs)
1	70.195	67.872
2	40.586	40.023
3	39.265	38.725
4	35.624	34.378
5	30.201	37.281
7	27.113	27.389
10	18.732	19.360

5. Conclusions and future work

Exploration of unknown environments is an important topic in mobile robot research due to its wide real-world applications, such as search and rescue, hazardous material handling, military actions, etc. The obvious advantage of multi-robot exploration is the concurrency, which can greatly reduce the time needed for the mission. Coordination among multiple robots is necessary to achieve certain efficiency in the robot exploration. Other desirable characteristics such as reliability, robustness also require the co-operation among multiple robots.

We have presented an interesting approach for cooperative exploration based on the SRT-Radial. The Multi-SRT considers two decentralized mechanisms of cooperation at different levels. The first simply consists in making an appropriate definition of the local frontier that allows each robot to plan its motion towards the areas apparently unexplored for the rest of the team. The second allows a robot that has finished with its individual exploration phase, to support others robots in their exploration task.

The integration of a localization module into the exploration process based on SLAM techniques will be an interesting topic for a future research. We can consider an extension of the Multi-SRT exploration method, where the robots constantly maintain a distributed network structure, i.e., where they are constantly in contact with each other. This communicative exploration idea is designed to take the constraints of wireless networking, namely the limited range of the transceivers, into account. Besides, the extension to heterogeneous robots and experimental validation will be topics for future works.

References

[1] W. Burgard, M. Moors, C. Stachniss, F. E. Schneider, Coordinated multi-robot exploration, *IEEE*

Transactions on Robotics, Vol. 21, No. 3, pp. 376-386, 2005.

- [2] Y. Cao, A. Fukunaga, A. Kahng, Cooperative mobile robotics: Antecedents and directions, *Autonomous Robots*, Vol. 4, pp. 1-23, 1997.
- [3] G. Dudek, M. Jenkin, E. Milios, D. Wilkes, A taxonomy for multi-agent robotics, *Autonomous Robots*, Vol. 3, pp. 375-397, 1996.
- [4] J. Espinoza León, Estrategias para la exploración de ambientes desconocidos en robótica móvil, *Master Thesis*, FCC-BUAP (in spanish), 2006.
- [5] J. Espinoza L., A. Sánchez L., M. Osorio L., Exploring unknown environments with mobile robots using SRT-Radial, *IEEE Int. Conf. on Intelligent Robots and Systems*, pp. 2089-2094, 2007.
- [6] A. Franchi, L. Freda, G. Oriolo, M. Vendittelli, A randomized strategy for cooperative robot exploration, *IEEE Int. Conf. on Robotics and Automation*, pp. 768-774, 2007.
- [7] A. Howard, M. Mataric, S. Sukthane, An incremental deployment algorithm for mobile robot teams, *IEEE Int. Conf. on Intelligent Robots and Systems*, pp. 2849-2854, 2002.
- [8] J. Ko, B. Stewart, D. Fox, K. Konolige, B. Limketkai, A practical, decision-theoretic approach to multi-robot mapping and exploration, *IEEE Int. Conf. on Intelligent Robots and systems*, pp. 3232-3238, 2003.
- [9] G. Oriolo, M. Vendittelli, L. Freda, G. Troso, The SRT method: Randomized strategies for exploration, *IEEE Int. Conf. on Robotics and Automation*, pp. 4688-4694, 2004.
- [10] R. Simmons, D. Apfelbaum, W. Burgard, D. Fox, M. Moors, S. Thrun, H. Younes, Coordination for multi-robot exploration and mapping, *17th Conf. of the American Association for Artificial Intelligence*, pp. 852-858, 2000.
- [11] B. Yamauchi, Frontier-based exploration using multiple robots, *2nd Int. Conf. on Autonomous Agents*, pp. 47-53, 1998.

Reducing Computational Complexity in Markov Decision Processes using Abstract Actions

Ma.de Guadalupe García-Hernández¹, José Ruiz-Pinales¹, Alberto Reyes-Ballesteros², Eva Onaindía³, J. Gabriel Aviña-Cervantes¹

¹ *Universidad de Guanajuato, Comunidad de Palo Blanco s/n, Salamanca, Guanajuato, México*
{garciag, pinales, avina}@salamanca.ugto.mx

² *Instituto de Investigaciones Eléctricas, Reforma 113, Temixco, Morelos, México,*
areyes@iie.org.mx

³ *Universidad Politécnica de Valencia, Departamento de Sistemas Informáticos y de Computación, Camino de Vera s/n, Valencia, España, onaindia@dsic.upv.es*

Abstract

In this paper we present a new approach for the solution of Markov decision processes based on the use of an abstraction technique over the action space, which results in a set of abstract actions. Markovian processes have successfully solved many probabilistic problems such as: process control, decision analysis and economy. But for problems with continuous or high dimensionality domains, high computational complexity arises because the search space grows exponentially with the number of variables. In order to reduce computational complexity, our approach avoids the use of the whole domain actions during value iteration, calculating instead over the abstract actions that really operate on each state, as a state function. Our experimental results on a robot path planning task show an important reduction of computational complexity.

1. Introduction

It is possible to find more realistic policies through decision-theoretic planning [6], considering that actions may have different effects in the world (no determinism), pondering the weight of alternative plans for achieving the problem goals, and considering their costs and rewards (extended goals). In addition, it must be assumed as partially observable [4], it means that the system state is partially accessible during execution time; therefore, it produces indistinguishable states. In a process control problem, many variables change dynamically due to the operation of devices (valves, equipment switches, etc.) or the occurrence of

exogenous events (uncontrollable events). If the control system does not consider the possibility of fault, then it will surely not make intelligent actions in the event of a fault occurrence. This process control is a very complex problem, where the uncertainty plays an important role during the search of solutions. In this way, decision-theoretic planning allows the evaluation of the strength and weaknesses of alternative plans. In spite of the addition of new capabilities to a planner, using heuristic search has shown limitations for the case of non integer data and additive graphs by solving real-world problems [5], it has been chosen to solve them by using Bayesian representation and inference [10] and Markov decision processes (MDPs) [3, 17]. The latter have successfully solved decision problems, e. g. process control, decision analysis and economy. However, the computational complexity of MDPs is a significant one for the case of continuous or high dimensionality domains. To cope with this problem, it has been proposed the use of abstraction techniques, also known as grouping techniques [6].

In this paper, we present a new approach consisting in to apply an abstraction technique over the action space during the solution of the MDPs, especially for large problems. We present experimental results indicating the high viability of the new approach. In Section 2, MDPs are described shortly and the problem faced by MDP solution methods with the scalability is explained. In Section 3, is presented an overview about the use of abstraction techniques in the solution of MDPs. In Section 4 we present our approach consisting in to apply an abstraction technique over the action space for solving MDPs. In Section 5 our

experimental results are described. In Section 6 we present the conclusions.

2. Markov Decision Processes

These processes are planning under uncertainty techniques and they were named in honor of Andrei A. Markov. They were introduced originally by Bellman [1] and have been studied intensely [3, 14, 17]. MDPs are a mathematical formalism for modeling a sequential decision problem, whose main goal is to find a reactive strategy or action policy, by maximizing an expected reward [17]. Markov's work supposes that an agent always knows the state where it is placed before executing any action (total observably), and the transition probability from a state depends only on the current state, not on its whole history; always taking into consideration a reward discount factor. Bellman's equation (see equation (1)) is the base for the solution of MDPs through value iteration [18] and one of its characteristics is the careful balance between reward and risk. For each possible state one equation is formulated; the unknown quantity is the utility of the reached state. Therefore, it results a system of equations, but these equations are non-linear because they try to obtain the maximum expected utility given by:

$$U(s) = R(s) + \gamma \max_a \sum_{s'} T(s, a, s') U(s')$$

where $U(s)$ is the expected utility of the current state s , $U(s')$ is the expected utility of the reached state s' , $R(s)$ is the immediate reward of the current state s , $T(s, a, s')$ is the transition model that gives the probability of reaching s' when action a in state s is applied, and γ is the discount factor ($0 < \gamma < 1$). The state transition model can be represented by a dynamic Bayesian network [12]. Otherwise, when the future reward is insignificant, then $\gamma = 0$. In contrast, when the current reward is identical to the expected discounted reward, then $\gamma = 1$. Therefore, Bellman's equation gives the utility of an agent in a specific state, and that utility is the immediate reward of this state plus the discounted utility from the reached state; this state results from applying an action over the previous one. The main idea consists in calculating the expected utility of each state and to use these utilities for the selection of the optimal action at each state. However, the problem is that this computation is feasible only for the case of discrete domains or with a few variables. This is due to the size of transition matrix which grows exponentially with the number of variables [18]. There are two ways to reduce the computational complexity:

by using decomposition (divide and conquer) or abstraction (group similar elements) of the problem. This paper is focused on the second one. In the next section we present an overview about the use of an abstraction technique in the solution of MDPs.

3. Abstraction on MDPs: an overview

While in Robotics it is not always necessary to know the exact localization of an autonomous agent, it is necessary to know its relative position with respect to other objects inside the room. Then, if we use only relative movements between objects, our search space will be essentially reduced. This example shows clearly the need of abstraction applied on probabilistic planning. In this way, there have been many efforts for developing different representation forms and computational methods for decreasing the search space. An abstraction technique achieves this decrement by grouping similar states, resulting in abstract states. Boutilier *et al* [7] used abstract states, but they made by hand the simplification of Bellman's equation. In this way, Reyes *et al* have proposed a simulator for planning of robotic movements (SPRM) [19] using abstract states, but they obtained high polynomial solution time. Recently, Chenggang *et al* [9] used first-order decision diagrams with STRIPS probabilistic operators like Rebel [20]; however they did not get a universal quantification. De Fariás *et al* [11] combined weights of relevant states with the value function, although an error was found with this method. Another abstraction technique achieves the decrement of the search space by grouping similar actions, resulting in abstract actions. Hauskrecht *et al* [13] tested the hierarchy with macro-actions, but he did not get the appropriate coupling; moreover he obtained additional complexity because of the presence of macro-actions. On the other side, Benson [2] proposed the use of an action hierarchy for planning systems. But it was tested only on simple problems and the simplification of the obtained equations had been made by hand. Morales [16] has used a constrained action set for each state during the inference process, through STRIPS type probabilistic operators, called abstract actions (r -actions) of the form:

If $\langle \text{state}(i) \rangle$ then $\langle r\text{-action}(k) \rangle$.

In order to obtain r -actions, it is necessary to evaluate the k -th- r -action of the i -th-state. Consequently r -actions only depend on the current state, reducing in this way the search space. However, the scalability problem still remains as a great challenge during the solution of MDPs [20], because large problems are highly demanding of solution time.

4. Our approach

In this paper we propose a new approach consisting in the application of an abstraction technique over the action space, resulting in r -actions, in the solution of MDPs (see Algorithm 1).

```

RAValueIteration( $S, A, T, R, \gamma, \epsilon$ ):
where  $S = \{s_0, s_1, \dots, s_{n-1}\} = \{s'_0, s'_1, \dots, s'_{n-1}\}$ ,
 $A = \{a_0, a_1, \dots, a_{m-1}\}$ 
 $T' \leftarrow \emptyset$ 
 $t \leftarrow 0$ 
1. for  $s \in S$  do
2.   for  $a \in A$  do
 $T'(a, s) \leftarrow \{T(a, s, s'), s' \mid T(a, s, s') \neq 0\}$ 
3. for each  $s \in S$  do  $U_0(s) \leftarrow 0$ 
4.  $t \leftarrow t + 1$ 
5. for each  $s \in S$  do
 $U_t(s) \leftarrow R(s) + \gamma \max_a \sum_{(p, s') \in T'(a, s)} p U_{t-1}(s')$ 
 $\pi(s) \leftarrow \arg \max_a \sum_{(p, s') \in T'(a, s)} p U_{t-1}(s')$ 
6. if  $\max_s |U_t(s) - U_{t-1}(s)| \geq \epsilon$  then go to 4
7. return  $\pi$ 

```

Algorithm 1. RAValueIteration.

In this way, our algorithm calculates only over the actions (r -actions) that really operate on each state (steps 1 and 2) during value iteration. So, the number operations performed by the algorithm will be reduced.

5. Experimental Results

We first tested the proposed algorithm on a robot path planning task, where we placed a robot in a 4×3 grid-world (see Figure 1), with states $\{s_1, s_2, \dots, s_{11}\}$. The robot had to reach the goal state with reward +1 (s_{11}), while avoiding the obstacle and the penalized state with reward -1 (s_7). The robot had desirable movements (with respect to the grid-world) with 80% probability: up, right, left. But it might do undesirable movements with 10% probability which are perpendicular to desirable movements. A collision with a wall or with an obstacle constrained the robot to stay in the same state. All non terminal states had reward

s8	s9	s10	s11 goal
s5	obstacle	s6	s7
s1 robot	s2	s3	s4

Figure 1. A 4×3 grid-world for robot movements.

-0.04, this value is negative in order to force the robot to get out quickly from those states.

Our algorithm was implemented in *Java* inside the simulator for planning of robotic movements (SPRM [19]) and all the experiments were performed on a 2.66 GHz Pentium D computer with 1GB RAM. In the above case, the resulting optimal action policy was {up, up, right, right, right} and the solution time was almost the same for both algorithms, our algorithm and the previous algorithm in SPRM, given a simple problem with only 11 states. Consequently, we decided to use more complex grids, e.g. with 25 to 400 states, and they were associated with 6, 10, 12 and 26 rewards. We repeated 10 times every combination of values. Figure 2 shows a comparison of the solution time required by the previous algorithm and our algorithm, for different number of states. Some of the results obtained with those combinations are presented in Tables 1, 2 and 3.

Table 1. Results obtained: previous algorithm (PREV) vs. our algorithm (RA).

states	time (ms) PREV	number of operations PREV	time (ms) RA	number of operations RA
25	47	384375	16	148215
100	94	6000000	39	1140000
400	1266	102400000	313	9935360

Table 2. Results obtained: previous algorithm (PREV) vs. our algorithm (RA) for 100 states.

rewards	time (ms) PREV	number of operations PREV	time (ms) RA	number of operations RA
10	94	6000000	39	1140000
26	94	6200000	46	1367720

Table 3. Results obtained: previous algorithm (PREV) vs. our algorithm (RA) for 400 states.

rewards	time (ms) PREV	number of operations PREV	time(ms) RA	number of operations RA
6	1195	96000000	328	11503200
10	1266	102400000	313	9935360
12	2969	242187500	820	26129900

In Table 1, we can see that when the number of states is 25, the solution time for the previous algorithm is 47 ms, while it is 16 ms for our algorithm. When the number of states is 400, the solution time for the previous algorithm is 1266 ms, while it is 313 ms for our algorithm. In both cases, our algorithm is faster than the previous one. From Tables 2 and 3, it can be seen that the solution time increases softly when the number of rewards grows, for a given number of states.

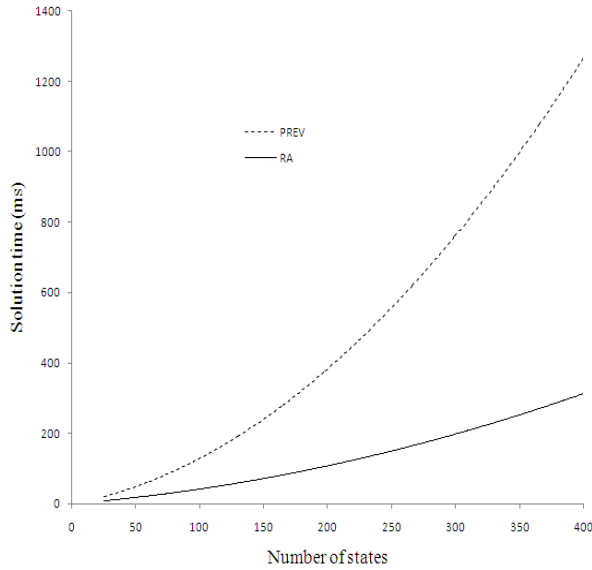


Figure 2. Solution time for previous algorithm (PREV) vs.our algorithm (RA).

Figure 2 shows a comparison of the solution time required by the previous algorithm and our algorithm as a function of the number of states. We can observe that the time required by the previous algorithm increases rapidly when the number of states grows whereas the curve corresponding to our algorithm is always below. On the other hand, in Tables 2 and 3 we can observe that the time is highly dependent of the number of rewards. For instance, for both algorithms, the time increases slowly and then it increases rapidly when the number of rewards is greater than 10. Resuming, when we apply reduction on the search space in MDPs through r -actions, we obtain a considerable decrease in the solution time for large problems.

6. Conclusions

The extension of capacities in a probabilistic planner through heuristic search has presented limitations for non integer values and additive graphs in real-world problems [5]. Nevertheless, the challenge

for solving MDP is the high dimensionality or continuous domains [19] like real-world problems, because they require an exorbitant amount of computational time. That requirement is due to the large growth of the search space with the number of variables. In order to decrease its solution time, our algorithm only calculates the actions that really operate on each state during value iteration. Our experimental results present a considerable reduction in solution time when r -actions were applied for solving MDPs. Then we concluded that our approach is advisable because it considerably reduces the search space and, therefore, it really decreases the computational complexity. In the literature there are two related works: the grid based discretization [4, 8] and the parametric approximations [15]. Unfortunately, the classic grid algorithms grow exponentially with the number of state variables. Otherwise, the parametric approximations, based on hierarchical clustering techniques, also have been used with little success.

10. References

- [1] R.E. Bellman, *Dynamic Programming*, Princeton United Press, Princeton, New Jersey, USA, 1957.
- [2] S.S. Benson, "Learning action models for reactive autonomous agents", *PhD Thesis*, University of Stanford, USA, 1996.
- [3] D.P. Bertsekas, *Dynamic Programming*, Prentice Hall, Englewood Cliffs, NJ, USA, 1987.
- [4] B. Bonet, J. Pearl, "Qualitative MDP and POMDP: An order-of-magnitude approach", *Proceedings of the 18th Conference on Uncertainty in AI*, Edmonton, Canada, 2002, pp. 61-68.
- [5] B. Bonet, H. Geffner, "Learning depth-first search: A unified approach to heuristic search in deterministic and non-deterministic settings and its application to MDP", *Proceedings of ICAPS*, 2006.
- [6] C. Boutilier, T. Dean, S. Hanks, "Decision-theoretic planning: structural assumptions and computational leverage", *Journal of AI Research*, 11, 1999, pp. 1-94.
- [7] C. Boutilier, R. Dearden, M. Goldszmidt, "Stochastic Dynamic Programming with factored representations", *Artificial Intelligence*, 121 (1-2), 2000, pp. 49-107.
- [8] R. Brafman, "A heuristic variable grid solution method of POMDP", *Proceedings of the 14th National Conference on AI*, pp. 727-733, Menlo Park, CA, USA, 1997.

- [9] W. Chenggang, J. Saket, R. Khardon, "First-Order Decision Diagrams for Relational Markov Decision Processes", *Proceedings of IJCAI*, 2007.
- [10] A. Darwiche, M. Goldszmidt, "Action networks: A framework for reasoning about actions and change under understanding", *Proceedings of the Tenth Conference on Uncertainty in AI*, Seattle, WA, USA, 1994 pp. 136-144.
- [11] D.P. De Farias, B. Van Roy, "The Linear Programming approach to approximate Dynamic Programming", *Operations Research*, vol. 51(6), 2003, pp. 850-856.
- [12] T. Dean, K. Kanazawa, "A model for reasoning about persistence and causation", *Computational Intelligence*, vol. 5, 1989, pp. 142-150.
- [13] M. Hauskrecht, B. Kveton, "Linear program approximations for factored continuous-states Markov Decision Processes", *Proceedings of NIPS*, 2003.
- [14] R.A. Howard, *Dynamic Programming and Markov Processes*. MIT Press, Cambridge, MA, USA, 1960.
- [15] C.A. Knoblock, "Automatically generating abstractions for problem solving", *PhD thesis*, School of Computer Science, Carnegie Mellon University, USA, 1991.
- [16] E. Morales, "Scaling up reinforcement learning with a Relational Representation", *Proceedings of the Workshop on Adaptability in Multi Agent Systems*, Sydney, Australia, 2003.
- [17] M.L. Puterman, *Markov Decision Processes*, Wiley Ed., New York, USA, 1994.
- [18] M.L. Puterman, *Markov Decision Processes: A survey*, UBC Commerce, Canada, 2002.
- [19] A. Reyes, P. Ibarguengoytia, L.E. Sucar, E. Morales, "Abstraction and Refinement for Solving Continuous Markov Decision Processes", *Proc. of the 3rd European Workshop on Probabilistic Graphical Models*, Czech Republic, 2006, pp. 263-270.
- [20] M. Van Otterlo, K. Kersting, L. De Raedt, "Bellman goes Relational", *Proceedings of the 21st International Conference on Machine Learning*, Banff, Canada, 2004.

New approaches for the school timetabling problem *

Ana Cerdeira-Pena¹ Luisa Carpente² Antonio Fariña¹ Diego Seco¹

¹Database Laboratory ²Department of Mathematics

University of A Coruña

Campus de Elviña s/n. A Coruña, 15071. Spain

{acerdeira, luisacar, fari, dseco}@udc.es

Abstract

School timetabling is a hard task that educational centers have to perform regularly and which implies a large waste of time and human efforts. For such reason designing techniques for the automatic generation of timetables is still of interest. Even though many contributions exist, the characteristics of the problem vary depending on the school policies, the country (laws), and other particular variables.

The complexity of this problem makes it difficult to find an optimal solution, so approximated techniques are traditionally used in practice. In this paper, we focus in the Spanish school timetabling problem and present several approaches to deal with it. The first technique proposed is based on the random non ascendent method (RNA). Then we provide several genetic algorithms which differ on the policies used for selecting how the next generation is created (including elitism) as well as on the levels of mutation considered. Finally, we study how to combine the two previous approaches. We run experiments both on synthetic and real scenarios in order to compare all the proposals. Even though the RNA and some of the pure genetic algorithms obtain good results in practice, we show that by joining RNA with genetic algorithms we gain stability in the results.

1. Introduction

Timetabling problems arising from educational institutions are related to the task of distributing teachers, periods of time, lessons and available resources (classrooms, labs,...) in such a way that some particular requirements are satisfied. Traditionally, depending on the institution these problems can be classified in three main groups: university, school, and exam timetabling problems.

In this paper, we focus in the school timetabling problem. Particularly, we are interested in all those cases where it is necessary to match fixed periods of time with lessons that are given by certain teachers. Under this context, different constraints have to be considered. Some of them, referred as *hard constraints* have to be satisfied for a solution to be considered valid (for example, a teacher cannot teach two lessons at the same time). Others, referred as *soft constraints*, reflect the preferences given by the teachers or by the policies of the school (for example, teachers usually want to minimize the holes in their schedule, and a school policy could aim at avoiding practising sports after lunch). Therefore, the accomplishment of hard constraints indicates the feasibility of a solution, whereas the soft constraints permit to quantify its goodness.

The timetabling problem we consider (including all the constraints) admits a mathematical programming representation [7], so an exact solution can be obtained by applying well-known techniques in this field [9]. However, it is known to be a NP-complete problem [4, 6, 10]. In practice, the high dimensionality of the problem makes it impossible to find an exact solution, and approximated methods are needed to tackle it. In most cases, they lead to *good quality* solutions in a reasonable amount of time, even at the expense of not guaranteeing that the *best solution* is reached.

We can find in the timetabling literature many works employing these kind of techniques. In [1] they give a survey oriented to the university timetabling problem. A graph coloring-based algorithm is shown in [3] also for the same scope. For the school timetabling field, we found solutions that are based on the use of genetic algorithms [2], but unfortunately they meet neither the requirements nor the structure of our problem. An interesting paper from Schaerf [7] discusses how to adapt both local search and tabu algorithms to this context.

In this work, we study a local search technique known as *Random Non Ascendent Method* (RNA) as well as two variants of genetic algorithms adapted to deal with our problem. The later differ basically in how new populations are ob-

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tained, depending on whether parents having descendance can be chosen again or not. Moreover, we have obtained different variations of both genetic algorithms by applying distinct criteria for mutation and elitist policies. We complete our work with experimental results obtained for different synthetic scenarios. We first compare these techniques separately, and then we also study the effects of combining RNA with the others. Finally, we also test a reduced set of our best algorithms in a real case scenario.

In small scenarios, the three original algorithms manage hard constraints quite effectively, but RNA is usually the best choice for dealing with the soft constraints in most cases, with the unique exception of a variation of a genetic algorithm that uses an elitist policy. In larger scenarios, RNA does not behave so well, and some variants of the genetic algorithms outperform it.

The paper is organized as follows. In Section 2 we define the framework of the school timetabling problem we deal with. Then in Section 3 we describe the techniques developed. Section 4 shows the experimental results obtained for the synthetic scenarios. Section 5 analyze the behavior of some of the best alternatives in a real case. Finally, the conclusions of our work are discussed in Section 6.

2. The school timetabling problem

2.1. Overview

Considering the existence of groups of students, teachers who teach particular subjects to those groups, and a fixed scheduling of periods, the school timetabling problem consists in finding in which period of time a given teacher has to teach a certain subject to a specified group. Additionally, two different types of constraints have to be considered during this process: *hard* and *soft constraints*. The first ones refer to constraints that must be satisfied to obtain a feasible solution of the problem, whereas the second ones permit to measure the goodness of a solution. That is, a good valid timetable must satisfy the *hard constraints*, and it should also fulfill as many *soft constraints* as possible. Common examples of hard constraints are that a teacher can only teach a lesson to one group at the same time, and that a given group cannot receive more than one lesson at the same time (possibly by different teachers). Soft constraints usually depend on school policies or on teacher's preferences (i.e. minimizing the amount of periods with lessons, minimizing the *holes* in the timetable, avoiding some subjects at a specified time, etc).

2.2. Constraints involved

As an optimization problem, the school timetabling problem can be defined by a *mathematical formulation* that describes the feasible regions through a solution space; and an *objective function*, which permits to lead a search process towards an optimal solution. On the one hand, from a theoretical point of view, the constraints that define such space and must be satisfied by a solution are actually the *hard constraints*. On the other hand, each *soft constraint* has usually an assigned weight and contributes to the value of the objective function. Therefore, if such constraint does not hold, the value of the objective function will increase, and consequently, the quality of the solution worsens.

In practice, we are interested in managing both kind of constraints with a function called *cost function* that leads the search process for a solution of the problem. The algorithms we present in this paper use such cost function. Therefore, they are search algorithms that move not only through the solution space, but through a wider search space that includes both feasible and non feasible solutions. As a consequence, during the search of a good solution, they have to be able to distinguish between *feasible* and *non feasible* solutions, as well as considering the goodness of a given solution. It seems that it could worth to design search algorithms that could move only through the feasible space towards a good solution. However, in most problems, it would be very difficult to find an initial feasible solution to start with. This is the reason why we added the *distance to feasibility* to the objective function that permits us to determine the optimality of an obtained solution. Finally, it is important to point out that in order to lead the search process towards feasible regions, the hard constraints are given a higher weight than that of the soft constraints, but both of them are included in the evaluation of the *cost function*.

Before describing the constraints considered by our algorithms, let us define the term *period*, shown as the unit of time in which a lesson is taught. So the weekly schedule is divided into periods, excepting breaks and free hours that are not considered periods. We define a *class* (referred to different concepts in the literature) as the association between a subject (such as Maths, Foreign language, etc.) and one or more *groups* of students that are previously defined by the educational institution. Groups are named depending on the level (1st, 2nd, ... course) and on how all the students in the same educational level are partitioned (A, B, C, ...). For example, as the Mandatory Secondary studies are known as *E.S.O.* in Spain, we will refer as 2A *E.S.O.* to the first partition of students (A) in the second level of *E.S.O.* Therefore, we will have classes such as:

- 1A *E.S.O.*-Maths: Maths in the group 1A *E.S.O.*
- 1A/B/C *E.S.O.*-Computer Science: an association of

students coming from different groups (in this case, from 1A E.S.O., 1B E.S.O. and 1C E.S.O.) who are taught Computer Science subject together.

2.2.1 Hard constraints & Soft constraints

We describe in detail the different constraints we take into account in our algorithms.

Hard constraints. Four constraints are used to measure the feasibility of a solution:

- *Overlaps*: They avoid the possibility of a class been taught by more than one teacher in the same period. It also avoids that classes sharing resources (ie. a classroom, lab, etc.), as well as, classes in which the same group is involved, could be assigned to the same period. The case of a teacher giving more than one lesson at the same time is not considered as our representation of the problem avoids this issue (see Section 3).
- *Simultaneity*: Two classes are defined as simultaneous classes if they are taught by different teachers at the same time. Sometimes a group must be divided into two or more subgroups that are taught different subjects (for example, the case of elective subjects) by different teachers in different classrooms simultaneously. We also consider classes that must be divided into two (or more) subclasses. That is, where the group associated to that class is divided into two or more subgroups that receive the same lesson possibly in a different place and with a different teacher. A good example would be a group of 50 students doing an individual exercise in two different rooms with 25 computers each (and two different teachers). In this case, the class might be divided into as many subclasses (simultaneous or not) as subgroups are needed. In the example above, if only one computers room were available, both subclasses would not occur simultaneously.
- *Unavailability*: It considers periods when a class can not be given or when a teacher cannot teach.
- *Consecutiveness*: This constraint checks whether a distribution of hours for a pair teacher-class is followed. For example, some practical lessons should be taught in two consecutive periods of time.

As the search space of our algorithms avoids some typical non-feasible scenarios, some constraints are not considered *hard*. For example, a class not being given all their lectures, or a lecture not taught at an unchangeable period.

Soft constraints. According to the preferences shown in Spanish schools, we show some interesting constraints that a schedule should try to fulfill:

- *Overuse*: It refers to the number of periods per day in which a teacher gives its lessons, over its specified maximum of periods per day.
- *Underuse*: When teachers have preferences on their minimum number of periods per day, it indicates the number of periods under such minimum.
- *Holes*: We consider the number of *empty* periods between two consecutive periods where a teacher is assigned a class. Breaks and free-time periods are not considered holes.
- *Splits*: They consider the number of periods between two non consecutive assignments to a same class in the same day.
- *Groups*: Assuming a specified maximum of periods per day for an association teacher-class, it considers the number of exceeding periods in such day. This constraint is only considered if a maximum number of consecutive periods (related with *consecutiveness*) for the pair class-teacher is not specified.
- *Undesired*: Assuming that there are periods in which a teacher would prefer not to teach, this constraint indicates the number of such periods where that teacher is assigned a class. This constraint is the *soft* version of the mandatory *unavailability* constraint (referring to teachers).

2.3. Cost function

The constraints described in the previous section are included into a function that associates a *cost* value to a given solution. Such value can be used to compare the goodness of different solutions. This function is defined as follows:

Definition 1 Let S be the search space; $s \in S$, a solution; N , the number of constraints considered and $f_i(s)$ the value of a specific function which scores the number of conflicts of the i^{th} constraint for the solution s . Then the cost function $F_c(s)$ is defined as: $F_c(s) = \sum_{i=1}^N f_i(s)$.

We can see that the lower the value $F_c(s)$ for a given solution s , the more quality of s . Therefore, the school timetabling problem can be shown as the search for the solution $s_i \in S$ such that $F_c(s_i) = \min_{s \in S} F_c(s)$.

Notice that we measure both the distance to feasibility, and the goodness of the solution. Therefore, by giving more weight to the hard constraints than to the soft ones, our algorithms lead the search process towards valid solutions. Additionally, by varying the weights of the soft constraints a given school can adjust the importance of the involved preferences.

3. Algorithms

In this section we first describe the representation of the problem used in our algorithms. Then we describe three different techniques to solve our timetabling problem (the first one is a RNA-based and others are based on genetic algorithms). Finally, we combine them obtaining two new hybrid algorithms. All of them are heuristic-based search methods which try to reach a good quality solution at a reasonable computational cost, even though it is not guaranteed its optimality or feasibility. This approach is commonly used in optimization problems which belong to the *NP-complete* class, as well as, in those ones that can be solved with an exact method, but whose solution is not computationally feasible.

3.1. Previous concepts

3.1.1 School timetabling problem representation

Many different representations have been chosen when tackling this problem, among them [5, 8]. In our case, we used a *teacher-oriented* representation. Hence, timetable is represented as a two-dimensional matrix containing in each cell (i, j) , the class given by the i^{th} teacher at the j^{th} period. As shown before, this representation avoids implicitly the case of a teacher giving more than one class at a same time.

3.1.2 Underlying movements

Although our algorithms represent different approaches, all of them need to traverse the search space to find good solutions. Traversing such space is done through *moves*. We defined two kind of moves: *i) simple-moves*, that are obtained by swapping two distinct values in a given row of the timetable. Obviously, moves including either the swap of identical classes or the swap of empty cells are skipped; and *ii) double-moves* that are the combination of two simple-moves when the first move leads to an unfeasible scenario. Otherwise, it consists just in the first simple-move.

Notice that the search space remains connected and any solution s_i can be reached from any other s_j in a finite number of moves.

3.1.3 Initial solution

Our search algorithms need to start, at least, with an initial solution. Genetic algorithms need more than just one solution as they have to begin with a initial population of individuals. From that starting point, they navigate through the search space by means of moves. An initial solution can be provided a priori, for example, by starting with a previous timetable. However, in most cases, a random initial solution

is generated. Actually a greedy procedure can be used to minimize the conflicts generated by some constraints such as *overlaps* and *unavailabilities*.

3.2. RNA Search

Following the RNA local search technique, we have developed an algorithm which, starting from an initial solution, iteratively moves from a solution to another doing *double-moves*. These are repeated until they make no improvements during a given number of iterations. It keeps track of the current best solution (s_b) at each stage, and by applying those moves generates a new solution s_i , trying to improve the value of $F_c(s_b)$. When $F_c(s_i) \leq F_c(s_b)$, s_i becomes the new best solution.

3.3. Genetic algorithms

Genetic algorithms are included into a branch of the *evolutionary computation* (EC) field called *evolutionary algorithms* (EAs), and base their operation on the evolution mechanism, in particular, on natural selection and inheritance features. Those algorithms do a simultaneous search in different regions of the search space. Starting with a *population*, a set of *individuals* or potential solutions, best candidates are selected based on their *fitness* value which measures the quality of each individual. These will be the *parents* of a new group of individuals obtained by modifying (recombining and mutating) the previous ones, which will compose a new population to be used in the next iteration of the algorithm. In all this process, both genetic operators and selection technique play an important role. The first ones determine how the genetic information is combined or modified. Among them, the most relevant are *crossover* and *mutation*, which allow passing information from parents to offspring, and jumping from a point to another of the search space, guaranteeing the genetic diversity, respectively. The second ones, decide what individuals will become parents of a new population. Many alternatives are known, but we will explain below those involved in our algorithms.

We consider a solution to our school timetable problem, that is, a timetable, as an *individual*, and obviously a set of them, as a *population*. The *fitness* value will be given by our *cost function* evaluation and we will be doing a *mutation* by means of a *simple-move*. We define the *one point crossover* that randomly selects a crossover point within the timetable, in such a way that all rows from the first one to that point are inherited from one parent and the rest ones, from the other. We have developed two main different approaches:

Tournament (GAT) Starting with a population randomly generated, two pairs of individuals are selected and then the best candidate of each one is chosen, according to its *fitness* value. In this way, the parents of the next generation

are obtained. New children result from applying the defined crossover operator and they will mutate at a given probability before being inserted into the population. The process is then repeated, updating the best solution found among the individuals up to now, and it stops when a certain number of cycles is completed. As can be noticed, some individuals can be selected as parents more than once, something that will not happen in the following approach.

Four Children Tournament (GAT4C) The main difference with the previous algorithm is that once two individuals are chosen to form a pair they are discarded, so they will never be chosen again. Hence, the size of a population must be multiple of *four* and also *four* children must be created in each iteration. The rest of the operation follows the same guidelines than the previous algorithm.

3.4. Hybrid algorithms

We can combine the previous techniques to study the performance of two different basic approaches: RNA and genetic algorithms. We check the usefulness of mutation as the way to avoid *local minima*. Thus, we have developed two new algorithms:

Tournament & RNA (GAT & RNA) In this approach, *Tournament* is alternated with *RNA* until a certain number of iterations is reached. First, a *Tournament* phase is performed. It is followed by a *RNA* phase which starts taking the best individual found among all the generations as the initial solution. After finishing the *RNA* stage, a new population of n individuals must be created as the input to a new genetic phase. To do this, during the *RNA* phase a list with the n best timetables is kept. Therefore, we guarantee that the best candidates known up to now are always included at the beginning of every *Tournament* phase, and consequently the best *genetic material*.

Four Children Tournament & RNA (GAT4C & RNA) This technique works exactly as the previous one. The only difference is the use of a *Four Children Tournament* phase instead of the *Tournament* one.

Apart from these techniques we have proposed some variants for our genetic-based algorithms. We include these strategies in our experiments. A brief description of such strategies is presented below:

- v1) To increase dynamically the number of mutations after a given number of iterations without improvements.
- v2) To apply a more elitist selection technique of the best candidates, so reducing its proportion to a given percentage.

- v3) Not to eliminate the loser candidates when we work with a *Four Children Tournament* phase, in such a way, all the individuals could be parents at least once.

4. Experimental results on synthetic data

In this section we summarize our experimental results over two sets of synthetic test cases with different size and configurations. Each collection is composed of a given number of test files. Each file was created depending on: *i*) the number of unavailable and undesirable periods for each teacher; *ii*) the number of unavailable periods for each class; and *iii*) the distribution of classes along the whole week (at most 2 periods per day). We run experiments¹ with our algorithms using in all cases the same configuration of their parameters. Those parameters are: *i*) the weight associated to each *hard constraints* was set to 1000. *ii*) The weight of the *soft constraints* varies: the weight of *overuse*, *splits*, and *groups* is set to 6; that of *underuse* is 4; for *undesired* periods we used the value 3; and finally the weight of *holes* is 1. *iii*) For the genetic algorithms we fixed the population size to 32, and the probability of mutation to 0.4. And *iv*) in the hybrid algorithms, we fixed the maximum number of iteration of the *RNA* phase to 150.

Small test collection The first test collection is composed of 10 files. Each of them has 6 groups, 70 classes and a number of 15 – 16 teachers who teach during 15 periods weekly. Table 1 shows the average value of unsatisfied constraints and the standard deviation from those values for 10 runs of each algorithm over the test data set. Each runs was limited in time to 30 minutes. Results show that most algorithms obtain good results and perform quite similarly, with the exception of a small group that perform very badly: variants $v1$ and $v1+v2$ of the genetic algorithms and variant $v3$. The best choices seem to be *RNA* and *GAT4C*.

Large scale scenario After discarding the algorithms that behave badly in the small scenario, we designed a huge file containing 27 groups, 333 classes and 71 teachers who teach during 15 periods weekly. Again, we run 4 times the different algorithms over the test file, limiting time to 5 hours each. In Table 2, results show that variant $v2$ obtains very good results, especially version *GAT & RNA*, that reduces the variability. *RNA* and variant $v1+v2+v3$ behave also well, but they lead to a high number of unsatisfied *soft constraints*.

5. A case study

In this section we test a selection of our best algorithms with real data obtained from a Secondary school (I.E.S.

¹ We used an Intel Core2Duo E6420@2.13Ghz with 2Gbytes of DDR2-800 memory. The computer ran Windows Xp OS. Prototypes were implemented in Java 1.6.

Table 1. Results for the small synthetic scenario.

Algorithm	constraints			
	Avg(F_c)		Std(F_c)	
	hard	soft	hard	soft
original				
RNA	0.16	0.60	0.09	0.39
GAT	0.22	1.45	0.22	0.32
GAT4C	0.31	2.05	0.29	0.68
GAT & RNA	0.44	2.30	0.50	1.35
GAT4C & RNA	0.38	2.46	0.33	0.93
var: v1				
GAT	6.94	23.57	3.81	3.88
GAT4C	7.07	23.32	3.99	3.84
GAT & RNA	0.34	2.74	0.29	0.96
GAT4C & RNA	0.28	2.74	0.25	1.20
var: v1+v2				
GAT	3.99	15.57	2.86	2.85
GAT4C	5.53	18.90	3.50	2.23
GAT & RNA	0.47	1.90	0.48	0.96
GAT4C & RNA	0.41	1.76	0.43	0.73
var: v2				
GAT	0.26	0.74	0.14	0.45
GAT4C	0.23	0.74	0.16	0.44
GAT & RNA	0.44	1.74	0.46	0.92
GAT4C & RNA	0.33	1.41	0.34	0.84
var: v1+v2+v3				
GAT4C & RNA	0.49	2.06	0.42	0.70
var: v1+v3				
GAT4C & RNA	0.43	4.33	0.42	1.55
var: v2+v3				
GAT4C	0.28	0.75	0.22	0.43
var: v3				
GAT4C	27.62	42.19	7.52	3.97

Menendez Pidal) in A Coruña-Spain. This school has 3 sections with different diplomas. Our data comes from one of them including: commerce, marketing, and international commerce studies. That scenario includes 11 teachers who can select 3 undesirable and 3 unavailable periods respectively. Teachers can be assigned classes from 1 to 5 periods per day. There are 33 classes, and 3 simultaneity relationships between classes.

In the top table in Figure 1 we can see that even though all the chosen algorithms reach good results, we can highlight those of RNA and GAT4C (which is the best choice). The bottom plot in that figure shows the evolution of the cost function in GAT4C with respect to execution time (1 hour). It is remarkable that during the first minutes the curve of the cost function is very steep, and that after 20 minutes

Table 2. Results for the large synthetic scenario.

Algorithm	constraints			
	Avg(F_c)		Std(F_c)	
	hard	soft	hard	soft
original				
RNA	0.88	63.58	0.82	3.77
GAT	2.88	67.21	2.11	8.42
GAT4C	3.88	71.04	2.26	5.85
GAT & RNA	1.50	62.54	1.54	6.50
GAT4C & RNA	1.75	70.33	1.35	3.95
var: v1				
GAT & RNA	1.75	63.75	1.19	4.71
GAT4C & RNA	2.25	69.04	2.12	4.49
var: v2				
GAT	1.94	26.04	1.39	4.17
GAT4C	2.69	34.08	1.53	6.53
GAT & RNA	0.94	30.38	0.98	2.77
GAT4C & RNA	1.94	35.71	1.44	4.12
var: v1+v2+v3				
GAT4C & RNA	1.00	52.67	0.73	5.71
var: v2+v3				
GAT4C	2.75	49.88	1.71	4.64

the quality of the solution obtained is very good. Finally, after 40 minutes the improvements are almost negligible.

6. Conclusions and future work

We have developed different solutions for the secondary school timetabling problem focusing on the Spanish context. They are based on the use of RNA and genetic-based algorithms. A wide study was done by introducing some variations of the genetic-algorithms, as well as by creating hybrid versions. To test the behavior of the different alternatives developed we first applied them to synthetic scenarios (what allowed us to discard some of them). Finally, we studied how the best techniques performed over a large scenario and over a real case.

Our results showed that the RNA method behaves well in most cases. It is usually able to reduce drastically the number of unsatisfied hard constraints, and consequently the cost function. However, it does not manage soft constraints so effectively. This issue is improved by some of the genetic-algorithms variants. More precisely, that called GAT4C performed very well in small scenario. It obtained similar results to those of RNA for the hard constraints and improved values for the others. In our large synthetic scenario the best choice was a hybrid solution (GAT&RNA) using an elitist policy.

Algorithm	constraints			
	Avg(F_c)		Std(F_c)	
	hard	soft	hard	soft
original				
RNA	0.88	5.95	0.74	1.72
GAT	1.05	6.90	1.01	2.92
GAT4C	0.55	6.37	0.72	1.83
GAT & RNA	1.63	8.10	1.07	2.71
var: v2				
GAT4C & RNA	2.20	8.42	1.95	3.69

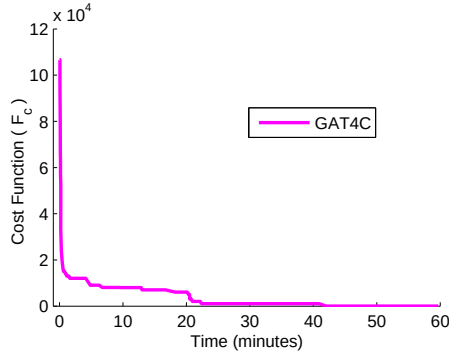


Figure 1. Results for a real case scenario.

As a future work, we are focusing in the generation of better initial solutions. The idea is to create initial timetables using a greedy procedure to avoid infeasibility. We also consider that more intelligent (although maybe heavier) moves can be used to skip bad quality timetables during the search process.

References

- [1] M. W. Carter and G. Laporte. Recent developments in practical course timetabling. pages 3–19, 1998.
- [2] A. Colomi, M. Dorigo, and V. Maniezzo. A genetic algorithm to solve the timetable problem. Technical Report 90-060, 1990.
- [3] P. de Haan, R. Landman, G. Post, and H. Ruizenaar. A four-phase approach to a timetabling problem in secondary schools. In *Practice and Theory of Automated Timetabling (PATAT 2006)*, pages 423–425, 2006.
- [4] D. de Werra. An introduction to timetabling. *European Journal of Operational Research*, 19(2):151–162, February 1985.
- [5] S. Gyri, Z. Petres, and A. R. Vrkonyi-Kczy. Genetic algorithms in timetabling. a new approach.
- [6] A. I. S. Even and A. Shamir. On the complexity of timetable and multicommodity flow problems. *SIAM Journal on Computing*, 5:691–703, 1976.
- [7] A. Schaerf. Local search techniques for large high-school timetabling problems. Technical Report 4, 1999.
- [8] K. A. Smith, D. Abramson, and D. Duke. Hopfield neural networks for timetabling: formulations, methods, and comparative results. *Comput. Ind. Eng.*, 44(2):283–305, 2003.
- [9] A. Tripathy. School timetabling—a case in large binary integer linear programming. *Management Science*, 30:1473–1489, 1984.
- [10] R. Willems. *School timetable construction: algorithms and complexity*. PhD thesis, Technische Universiteit Eindhoven, The Netherlands, 2002.

Uncertainty, Probabilistic Reasoning, and Fuzzy Logic

Bayesian Network Based Time-sequence Simulation for Power System Reliability Assessment

Zhu Yongli*
North China Electric Power
University, Baoding, 071003,
China
yongli_pw@heinfo.net

Huo Limin, Zhang Liguo
Agricultural University of
Hebei, Baoding, 071001,
China

Wang Yan
North China Electric Power
University, Baoding, 071003,
China

Abstract

In this paper, an approximate inference algorithm on Bayesian networks (BN) for reliability assessment of power systems by time-sequence simulation is presented. Based on uniformly distributed pseudo random numbers, the duration time sampling for a component state can be obtained through inverse transform method. According to the logical relationship of the power system to be assessed, the sampling of minimal state cut sets of the system and the system node's sampling are obtained respectively. Thus, the corresponding sample sequence in forward sampling is available. When the amount of sample sequence is large enough, marginal and conditional statistic values are close to marginal and conditional probability of the node. Then the approximate inference results of the BN can be obtained. Numerical simulating results show the effectiveness of the proposed algorithm. It can easily identify the weaker components of a system in reliability, calculate the system reliability indices, and is suitable for a large-scale power system.

Key words: Bayesian networks; Approximate inference; Power systems; Reliability assessment

1. Introduction

The main goal of power system reliability assessment is to provide qualitative analysis and indices in power supply reliability for the system's operation and planning. After the qualitative analysis, it is very necessary to find out the system's weak components. At present, there are only two approaches, analytical approach and Monte-Carlo simulation approach, used to calculate the reliability indices. However, both of them can't find the system's weak components.

Bayesian network (BN) is one of the most effective theoretical models for uncertainty knowledge expression and inference. Based on probability theory, it can make causal inference, diagnosis inference and

explaining inference. Given one or several variables' values, all the other variables' conditional probabilities can be obtained. Therefore, we can make any kinds of hypothesis analysis in the view of whole system and sequentially can identify the system's bottlenecks.

Reference [1] discusses the properties of the modeling framework that make BN particularly well suit for reliability applications. Reference [2] proposes a novel BN-based reliability modeling/analysis approach to handle the ever-increasing complexity of systems, and perform various analyses such as reliability index computation, sensitivity analysis. Reference [3] presents a continuous-time Bayesian network (CTBN) framework for dynamic systems reliability modeling and analysis. References [4-5] discuss BN models of generic multi-states reliability systems and its inference techniques. Reference [6] proposes a method to apply BN to structural system reliability reassessment. Reference [7] shows how to use BN techniques to evaluate the reliability of distributed communication systems. Reference [8] is the first paper to apply BN in reliability evaluation of interconnected power systems. References [9-11] make some study on applying BN to reliability evaluation of power stations or distribution systems.

In the process of applying BN to reliability assessment of a power system, all the above references [1-11] adopt exact inference. However, it is difficult to perform exact inference when the assessed power system is large, so the power system has to be divided into several subsystems. Even so, the exact inference is still hard and even impossible sometime, because each component in a subsystem has many states. Cooper shows that exact inference for any structure BN is NP-hard [12]. Therefore more and more people began to pay their attention to BN approximate inference algorithms.

The main existing BN approximate inference algorithms are the search based ones [13] and

stochastic simulation based ones [14]. However, both the exact inference algorithms and the approximate inference algorithms can't solve the time-related problems with the indices of fault frequency and fault duration time. They also can't express the fault rate and repair rate of power system components [10].

Considering the problems mentioned above, a new Bayesian network inference algorithm for time-sequence simulation, namely time-sequence simulation inference algorithm, is presented in this paper. It adopts the basic idea of time-sequence simulation used in the traditional reliability assessment of a power system and the basic principle of forward sampling algorithm for BN stochastic simulation inference. Adopting this strategy, not only all the power system's reliability indices can be calculated, but also causal and diagnosis inference of BN can be performed, which sets up a foundation for finding out the reliability bottlenecks of the system.

2. The proposed BN based inference algorithm for reliability assessment by time-sequence simulation

The stochastic simulation is a commonly used approximate inference approach for Bayesian networks. It includes the following two steps. First, stochastically sampling for the joint probability distribution of BN is made to obtain enough samples. Secondly, variables' probability values are obtained by frequency calculation according to the samples. The more the samples, the higher the calculation accuracy is. Among the existing sampling methods, forward sampling is the basic. It is a partial sampling method, which can avoid direct sampling in the completed joint probability distribution^[14]. The algorithm includes the following steps.

a. Nodes in the BN are numbered in sequence. It is noticed that the number of a node must be bigger than that of its parent node.

b. According to the number sequence, sampling each node by using random number generator, respectively. The sampling value is 0 or 1. For root node, according to its conditional probability table (CPT), stochastically produce its value. For evidence node, if the stochastic sampling value is inconsistent with the evidence, give up this sampling and restart at the first node. For other nodes, after inputting their parents' value (which have existed after anterior sampling) into their CPT, also produce their values by random number generator.

c. After step b, one sample consistent with the evidence is produced. If the scheduled sampling times are not achieved, go to step b.

d. Use the obtained samples to calculate the BN variables' probabilities.

The time-sequence simulation is usually used in power system reliability assessment to calculate the reliability indices, because it can take time-related problems into consideration. By using uniformly distributed pseudo numbers, it samples the component states based on the known state probability distribution of the component, and then the faulty time and the repair time of the component can be obtained. During the process of the sampling, the system state should be checked as soon as a component's state is changed. If the system is in faulty state, record the fault's duration time (interval) and add 1 to the fault times. When the above simulation is finished, the system reliability indices can be calculated based on the total fault duration time and the fault times.

To perform the BN based power system reliability assessment, the following two steps are necessary. First, convert reliability block diagram into the corresponding BN. Then, calculate reliability indices and find the system's weakness. A BN used for reliability assessment looks like an inverted fault tree. Its top nodes are component variables and its bottom node is the system variable. The procedure to calculate the system node's marginal probability by the forward sampling is similar to the one to calculate the probability indices in reliability assessment by the time-sequence simulation. The only difference is that the former uses state sampling frequency to indicate the probability of the state, while the latter calculates the power availability by the rate of time interval. If the time-sequence simulation can be adopted in the BN inference for power system reliability assessment, the diagnosis inference, causal inference and the system node's marginal probability can be calculated, so the weaker components for the system's reliability can be found out. The following sections of the paper will mainly introduce the principle and procedure to perform the above relating conditional probability computation of the reliability assessment BN by time-sequence simulation.

2.1. Component state simulation

Take two-state component as example, as Fig.1 shows, the component's working process is its state transfers between failure state and normal state over a long time. Here, T_{ik} denotes the duration time of component i in state x_{ik} , and it can be acquired by computer simulation. In the reliability analysis of engineering system, it is generally supposed that the event duration time is exponential distribution.

Thus, T_{ik} can be obtained by inverse transformation method, shown in equation (1)

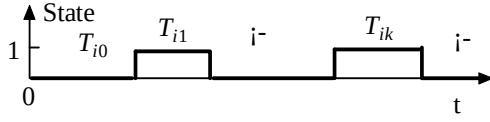


Fig.1 Component working state

$$T_{ik} = \begin{cases} -T_{MTBF} \ln r & x_{ik} = 0, \text{ Normal} \\ -T_{MTTR} \ln r & x_{ik} = 1, \text{ Fault} \end{cases} \quad (1)$$

Here, r is a uniformly distributed random number between 0 and 1. T_{MTBF} and T_{MTTR} are component's average continuous working time and average repair time, respectively. When both T_{MTBF} and T_{MTTR} are exponential distribution, they are reciprocal relations with failure rate and repair rate, respectively.

2.2. Inference procedure of time-sequence simulation

Suppose the number of components in the reliability system is m , including transmission line, transformer, circuit breaker, generator, and so on, and the states of every component can be simulated according to the approach mentioned in 2.1. System state is composed of its components' states. If we assume system state is x_k at the time t_k , then x_k can be denoted by $x_k = \{x_{1k}, x_{2k}, x_{3k}, \dots, x_{mk}\}$, where m is the number of system component. System state will change, only when its component's state changes. It is shown in Fig.2.

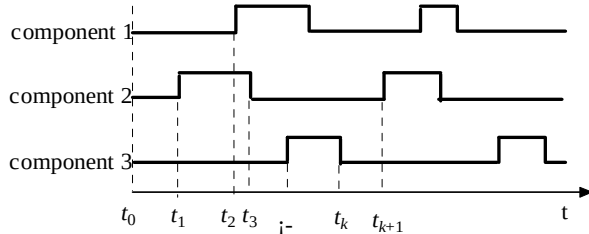


Fig.2 State changes of system

- The simulation process starts with system's normal state, i.e. when $t_0=0$, each component is normal and system state is $x_0 = \{x_{10}, x_{20}, \dots, x_{m0}\}$. According to equation (1), sampling the duration time for each component and then obtain T_{i0} , $i=1, 2, \dots, m$. Assume the duration time of system state x_0 is T_0 , then $T_0 = \min \{T_{i0}\}$.
- Assuming the current simulation time is t_k , the system state at this time is $x_k = \{x_{1k}, x_{2k}, \dots, x_{mk}\}$. Here, x_{ik} denotes the state of component i at t_k . Thus the duration time of system in state x_k is $T_k = \min \{T_{ik}\}$.

- Assume the next time of t_k is $t_{k+1} = t_k + T_k$. At this time, the system state is x_{k+1} , and $T_{i, k+1}$ can be obtained by the equation as follows.

$$T_{i, k+1} = \begin{cases} T_{ik} - T_k & T_{ik} \neq T_k \\ \text{sampling } T_{i, k+1} \text{ by equation (1)} & T_{ik} = T_k \end{cases} \quad (2)$$

- Judge whether the system state x_{k+1} is faulty. If yes, go to step e, otherwise go to step f.
- Add T_k to the total system fault time, and calculate the system's reliability indices and diagnosis inference's conditional probability. At the same time, according to the component state at t_k , add T_k to the corresponding state accumulator. The task of BN diagnosis inference is to compute the fault probability (or normal probability) for each component under the supposition that system is fault. The conditional probability of diagnosis inference equals to the ratio of the accumulate time of component at some state to the system total fault time.
- Accumulate simulation times and repeat step b to e until reliability indices converge or simulation time is over.

The above process is used to compute reliability indices and diagnosis inference. Causal inference can be realized by making a few modifications to it. The task of causal inference is to compute the probability of system at fault (or normal) state when some or several components malfunction.

The process of causal inference is as follows. In step b mentioned above, first check whether the state of evidence component and the evidence are identical. If yes, add T_k to the total time of corresponding state and go to next step. Meanwhile, in step e, the only thing we should do is to accumulate the total system fault time. Otherwise, jump to step f. Conditional probability of causal inference is the ratio of the total system fault time to the accumulate time of evidence component at its evidence state.

3. Simulation analysis

To test the validity of Bayesian network time-sequence inference algorithm presented in this paper, we takes the Bayesian network of some small power system shown in Fig.3[9] as an example to carry out simulation analysis and error analysis. And it is also made a comparative analysis with general BN stochastic simulation inference algorithm.

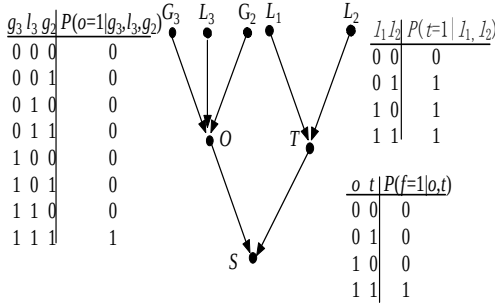


Fig.3 Bayesian network model of a power system

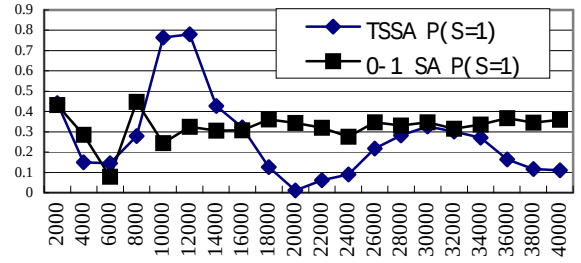
The error of approximate algorithm can be divided into four kinds, i.e. deterministic absolute error, deterministic relative error, probabilistic absolute error and probabilistic relative error [15]. According to custom, we use deterministic relative error to measure the accuracy (other method can obtain the same result). It is defined as follows:

$$\varepsilon = \left| \frac{P(X=x|E=e) - Z}{P(X=x|E=e)} \right| \times 100\% \quad (3)$$

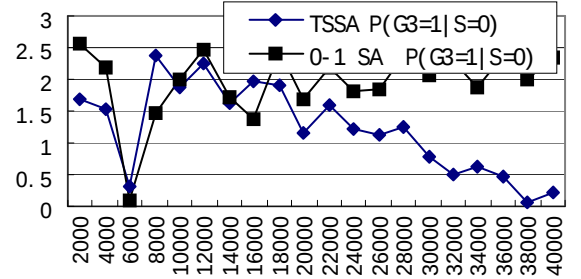
where Z denotes the calculate result of time-sequence simulation algorithm or stochastic simulation algorithm, and $P(X=x|E=e)$ is the calculate result of exact inference algorithm.

We developed the computer application for both Bayesian network time-sequence simulation inference algorithm and stochastic simulation algorithm (adopting forward sampling, and namely, 0-1 simulation algorithm here), with which the Bayesian network shown in Fig.3 are simulated for forty thousand times. The results of the marginal probability, the causal inference's conditional probability and the diagnosis inference's conditional probability calculated by both algorithms are compared in Fig.4, respectively.

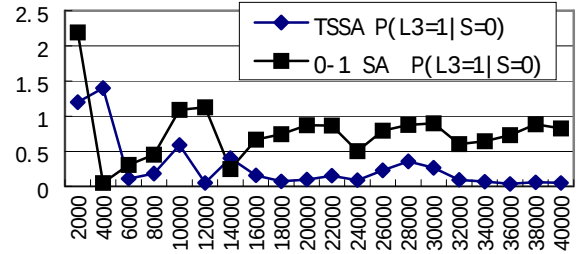
Comparative analysis shows that the convergence speed and the precision of the two algorithms are similar, and the maximal relative error is under 2.5%. In reliability assessment possess of power system, the time-consuming calculation of non-intersection cutest is usually ignored. In that case, the fault probability's relative error of the given power system, shown in Fig.3, is 4.95%. Base on above analysis, time-sequence simulation inference algorithm can meet the demand for the precision of power system reliability assessment.



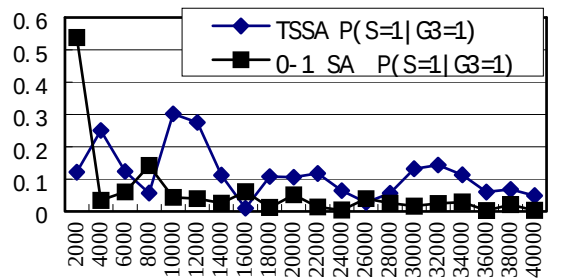
(a) Error of marginal probability



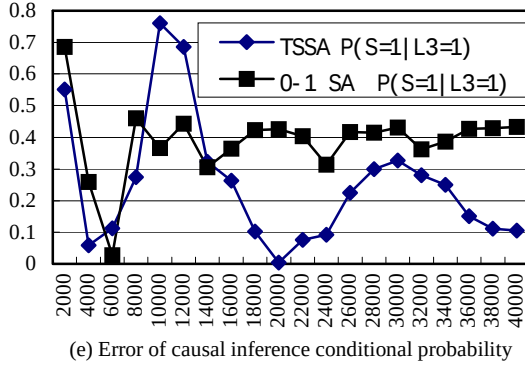
(b) Error of diagnosis inference conditional probability



(c) Error of diagnosis inference conditional probability



(d) Error of causal inference conditional probability



TSSA denotes time-sequence simulation algorithm,
0-1 SA denotes 0-1 simulation algorithm

Fig.4 Comparison of the computed error of time-sequence simulation inference algorithm and forward simulation inference algorithm

4. Case Study

To test the validity of time-sequence simulation inference, here we take the reliability assessment for the main wiring shown in Fig.5 as an example. Reliability model for breaker is shown in Fig.6 (a). Reliability models for generator, transformer and transmission line are identical, as shown in Fig.6 (b). Here, N 、 M 、 S 、 R denote normal state, planned maintenance state, spread fault state and repair state, respectively.

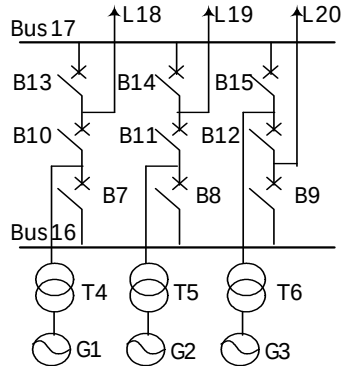


Fig.5 Breaker and a half configuration

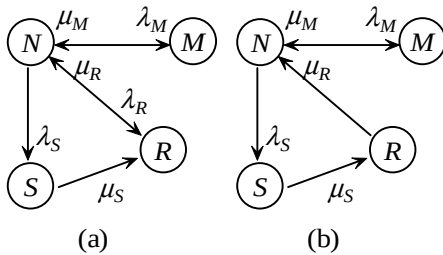


Fig.6 Component reliability model

The method to build BN is as follows [10]. Firstly, find all the minimal paths from generator to all the accessible load ends by bidirectional chain and depth first search technology. Secondly, use matrix operation to find all the first-order and second-order minimum state cut-sets. Lastly, according to the logical relations of component state, minimum state cut-sets, load node and system node, form the corresponding BN. Considering the breakers' resistance to act and the non-random of component planned maintenance, it is need to make some corresponding revision to the corresponding CPT [10].

None of the following software, exact inference software based on junction tree algorithm of this paper, Genie/Smile software developed by University of Pittsburgh and MSBN software developed by Microsoft, can realize exact inference for the given Bayesian network. Here, it is after deleting the minimal cut-sets of SS type whose probability value is under 10^{-11} and merging the son nodes which have the same parental nodes, but different parent states, into multi-states nodes that the precise inference can be realized.

In time-sequence simulation inference algorithm presented in this paper, next event moving forward principle is adopted to simulate the real operating status of the system. For two-state component, its state alters between fault state and normal state. For multi-states component, its state changes according to the order shown in Fig.6 (a). Here, take Fig.6 (a) as an example to illustrate the component's state altering order. If the current state is S-state, the next state is R-state. If the current state is R-state, the next state is N-state. If the current state is N-state, the next state may be R-state, S-state or M-state, which is determined by the value of r (defined in 2.1). If $r \leq \lambda_R / (\lambda_R + \lambda_S + \lambda_M)$, the next state is R-state.

Here λ_S , λ_R , λ_M denote spread fault rate, fault repair rate, and planned maintenance rate, respectively.

If $\lambda_R / (\lambda_R + \lambda_S + \lambda_M) < r \leq (\lambda_R + \lambda_S) / (\lambda_R + \lambda_S + \lambda_M)$, the next state is S-state. Otherwise the next state is M-state.

The calculating time is less than 5 seconds by using either exact inference algorithm or time-sequence simulation algorithm.

The paper adopts the original reliability data of components used in Reference [16], where it is supposed that the capacities of generator and transmission line are 1000MW, the length of transmission line (L) is 100km and the successful

probability to close component is 1. The power supply continuity indices of every load point acquired by both exact inference and time-sequence simulation algorithm and the relative errors are shown in Tab.1. The load-point reliability indices for power supply abundance acquired by time-sequence simulation algorithm are shown in Tab.2 and exact inference don't have the ability to calculate these indices. For malfunction, the EENG (Expected Energy Not Generated) is 1058.91757MWh/a and LOGP (Loss of Generating-capacity Probability) is 0.001198076. Under the abnormal power supply condition that the power supply of the three load points can't be acquired at the same time, the posterior conditional probabilities of every component at each state, which are obtained by BN diagnosis inference, are shown in Tab.3.

Tab.1 Load-point reliability indices

Load Point	Breaker and a half configuration		
	Exact	simulation	Error(%)
W1	0.999184	0.999600	-0.04163
W2	0.999184	0.999602	-0.04183
W3	0.999184	0.999600	-0.04163
W1W2W3	0.997555	0.998802	-0.12501

Tab.2 Load-point reliability indices

Capacity	Probability	Frequency (times/year)
300MW	0.998802	0.876987
200MW	0.00118973	0.808258
100MW	5.95862E-06	0.0475961
0 MW	2.38759E-06	0.021133

Tab.3 Component's posterior probability of 3/2 breaker schemes

C	E (P_N)	S (P_N)	E (P_R)	S (P_R)	E (P_S)	S (P_S)	E (P_M)	S (P_M)	R (%)
G1	0.909762	0.953356	0.0365421	0.0185275	3.84611E-03	0.00228386	0.04985	0.0258328	—
G2	0.909762	0.951236	0.0365421	0.0198399	3.84611E-03	0.00226757	0.04985	0.0266562	—
G3	0.909762	0.951889	0.0365421	0.0195188	3.84628E-03	0.00227965	0.04985	0.0263123	—
T4	0.981369	0.990906	6.88975E-04	0.000379257	1.89569E-05	7.71108E-06	0.0179245	0.00870724	—
T5	0.981369	0.990576	6.88975E-04	0.000336932	1.89569E-05	8.33006E-06	0.0179245	0.00907882	—
T6	0.981369	0.991586	6.88975E-04	0.000263351	1.895697E-05	4.01788E-06	0.0179245	0.00814627	—
B7	0.984976	0.995027	1.43923E-03	0.000430666	9.20148E-05	1.84575E-05	0.0134928	0.00452376	0
B8	0.984976	0.994981	1.43924E-03	0.000408108	9.20148E-05	1.05327E-05	0.0134928	0.00460078	0
B9	0.971769	0.989894	5.95805E-03	0.00139626	2.75861E-03	0.00197511	0.0195148	0.00673453	4.393E-03
B10	0.974561	0.992073	5.74176E-03	0.00108931	2.75861E-03	0.00188321	0.0169391	0.00495479	1.30771
B11	0.974561	0.992207	5.74176E-03	0.00180283	2.75861E-03	0.00183864	0.0169391	0.00415156	1.31017
B12	0.974518	0.990495	5.74593E-03	0.00163022	2.75861E-03	0.0019851	0.0169778	0.00588922	1.30582
B13	0.971768	0.990292	5.95813E-03	0.00133667	2.75861E-03	0.00189815	0.0195154	0.00647269	4.408E-03
B14	0.971768	0.98978	5.95813E-03	0.00159271	2.75861E-03	0.00197462	0.0195154	0.00665297	4.409E-03
B15	0.984937	0.99519	1.43918E-03	0.000411696	1.3187E-04	3.41139E-05	0.0134923	0.00436436	0
Bus16	0.998074	0.998954	2.22258E-04	5.62518E-05	—	—	1.70376E-03	0.000990146	—
Bus17	0.997553	0.998823	4.07785E-04	7.66452E-05	—	—	2.03948E-03	0.00110054	—
L18	0.682297	0.676895	0.316735	0.322685	3.31018E-05	7.22883E-06	9.34654E-04	0.000412618	—
L19	0.682297	0.674265	0.316735	0.325207	3.31018E-05	1.74607E-05	9.34652E-04	0.00051114	—
L20	0.682297	0.669179	0.316735	0.33026	3.31018E-05	1.95978E-05	9.34652E-04	0.000541674	—

Note: E (Exact), S (Simulation), R (Refusing action rate)

Tab.3 shows that the posterior conditional probabilities acquired by two kinds of algorithms are very close and the order of posterior probability is same. In other words, the proposed algorithm can meet the demand for reliability analysis. It also shows that the sum of each state posterior probability for every component is 1(for exact inference algorithm), or approximately 1(for time-sequence simulation algorithm). The component, whose posterior probability of N-state is minimal, is the system's weakness. It is clear that the three transmission lines are the weakest components in Fig. 5. For other states, the bigger the posterior probability value is, the bigger

the side-effect on system normal operation is. In Tab.3, the components with maximal posterior probability of S-state are the generators and breakers, so we should shorten their fault isolation time. The components with maximal posterior probability of R-state are the three transmission lines, so we should shorten their fault repair time or strengthen maintenance. The components with maximal posterior probability of M-state are the three generators, so we should shorten their planned maintenance time. Lastly, it is worth to emphasize that the proposed algorithm can calculate the percentage of system fault times resulting from breakers' rejection in the total fault times, so it can

reveal the effect degree of different breaker rejection ratio for system reliability. Thus, Tab.3 clearly shows that the rejection of B10, B11 and B12 will affect the system reliability more strongly than others and we should strengthen maintenance or use the breakers of good quality.

5. Conclusion

In this paper, an approximate Bayesian network inference algorithm for power system reliability assessment by time-sequence simulation is presented. Being compared with the existing exact inference and approximate inference algorithms for a Bayesian network, it has the distinct advantage in capability to calculate the fault frequency indices and find out weaker components of a system. Moreover, it avoids the defect that exact inference algorithm is hardly to apply in a large-scale network.

The simulation results show that the calculation results of the presented Bayesian network inference algorithm by time-sequence simulation are very close to those of exact inference algorithm. With better applicability and quick computation speed, the presented algorithm can meet the electric utilities' demands on calculating reliability indices and identifying the weaker components of their systems.

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References

- [1] Langseth,H., and Portinale,L., 'Bayesian network in reliability', *Reliability Engineering & System Safety*, 2007, 92,(1), pp. 92-108
- [2] Boudali,H., and DuganA ,J.B., 'A discrete-time Bayesian network reliability modeling and analysis framework', *Reliability Engineering & System Safety* , 2005,87,(3),pp. 337-349
- [3] Boudali,H., and Dugan,J.B., 'A Continuous-Time Bayesian Network Reliability Modeling, and Analysis Framework', *IEEE Transactions on Reliability*,2006, 55(1), pp.86-97
- [4] Zhou,Z.B., Jin,G., Dong,D.D., and Zhou,J.L., 'Reliability analysis of multistate systems based on Bayesian networks', *Proceedings of the 13th Annual IEEE International Symposium and Workshop on Engineering of Computer Based Systems (ECBS'06)*,2006,pp. 344 – 352
- [5] Wilsona, A.G., and Huzurbazar, A.V., 'Bayesian networks for multilevel system reliability ', *Reliability Engineering & System Safety*, 2007, 92,(10), pp.1413-1420
- [6] Mabadevan,S., Zhang,R.X., and Smith,N., 'Bayesian networks for system reliability reassessment', *Structural Safety*, 2001, 23(3),pp.231-251
- [7] Honari,B., Donovan,J., Flood, B., and Murphy,E., 'Use of Bayesian Networks in Estimating the Reliability of Distributed Communication Systems', *Proceedings of the 12th International Conference on Reliability and Quality in Design*,2006, Chicago,IL(US) ,pp.25-29
- [8] David,C.Y., Thanh,C.N., and Peter,H., 'Bayesian Network Model for Reliability Assessment of Power Systems', *IEEE Trans. on Power Systems*, 1999,14,(2), pp.426-432
- [9] Huo,L.M., and Zhu,Y.L., 'Reliability assessment of power system by Bayesian networks. 2002 International conference on power system technology, Kunming, China, pp.876-879
- [10]Huo,L.M., and Zhu,Y.L., 'Bayesian networks application to reliability evaluation of electric distribution systems', *Transactions of china electrotechnical society*, 2004,19,(8), pp.113-118
- [11] Wang,C.S., and Xie,Y.H., 'Applying Bayesian network to distribution system reliability analysis', *2004 IEEE Region 10 Conference on Analog and Digital Techniques in Electrical Engineering (TENCON 2004)*, 2004, pp.562-565
- [12] Cooper,G.,F. 'Computational complexity of probability inference using Bayesian Belief Networks', *Artificial Intelligence*, 1990,15, pp.246-255
- [13] Santos,E.,J., and Shimony,S.E., 'Deterministic approximation of marginal probabilities in bayesnets', *IEEE Transactions on Systems, Man, and Cybernetics*, 1998,28,(4), pp.377-393
- [14] Henrion,M., 'Propagating uncertainty in Bayesian networks by probabilistic logic sampling',*Uncertainty in Artificial Intelligence 2*,New York: Elsevier Science Publishers, 1988,pp.149-163
- [15] Hu,Z.,Y., and Qu,L.S., 'A simulation algorithm for Bayesian network inference', *Journal of system simulation*, 2004,16,(2), pp.286-288
- [16] Chen,H., Zhou,J.Q., and Wang,Y.H., 'Reliability evaluation on scheme of electric one-line diagram for pubugou hydroelectric power station', *Design of hydroelectric power station*, 1994,10(3), pp.67-73

Interpretation of Mammographic Using Fuzzy Logic for Early Diagnosis of Breast Cancer

Jorge R. Pérez-Gallardo, Beatriz Hernández-Vera, Alberto A. Aguilar-Lasserre,
Rubén Posada-Gómez

Instituto Tecnológico de Orizaba, Veracruz, México
{jrgallardo, bhernandezv, aaguilar, rposada}_@itorizaba.edu.mx

Abstract

Accuracy and interpretability are two important objectives in the design of Fuzzy Logic model. In many real-world applications, expert experiences usually have good interpretability, but their accuracy is not always the best. Applying expert experiences to Fuzzy Logic model can improve accuracy and preserve interpretability. In this study we propose an accessible tool that helps medical interpretation of suspect zones or tumors in mammographics. This paper describes a methodology to locate precisely different kind of lesions in breast cancer patients. The use of Fuzzy Logic model improves the diagnostic efficiency in tumor progression. After applying an image segmentation method to extract regions of interest (ROIs), the values obtained feed the system. The Fuzzy Logic model processes them to achieve Breast Imaging Reporting And Data System (BI-RADS®). Some experimental results on breast images show the feasibility of the propose methodology.

1. Introduction

Breast cancer is the major cause of death by cancer in Mexican female population. The mortality rate from this disease among the malign cancer diseases, according with the National Institute of Statistic, Geography and Information (INEGI), has increased from 13.3% in 2004 to 15% in 2006 [1]. Clinical trials have shown that early detection of breast cancer is a key to breast cancer control [2].

Visual information leads the doctor integrates it in the reasoning to formulate a diagnosis. The most common medical exam for breast cancer diagnosis is the mammography. The breast is X-rayed and reproduced in film-mammogram. The mammography is useful to detect the breast cancer in the initial phase that could show some types of cancers imperceptibles

by grope. The difference between malign and benign cancer is essentially made with base in their morphology. Typical views [3] for diagnostic mammograms include the Cranio-Caudal view (CC) and the Medio Lateral Oblique view (MLO).

Although mammography is an efficient tool in the detection process in the first steps, its disadvantage is its inefficiency in distinguishing between probable malignant and benign anomalies which requires sampling.

Aiming at reducing the discordance in the interpretation of mammographic findings and standardizing the terms for characterization and reporting, the American College of Radiology published, in 1993, the Breast Imaging Reporting and Data System (BI-RADS®), the studies are classified with basis on the lesions' suspicion level into: category 1 – without positive findings; category 2 – benign findings; category 3 – probably benign findings; category 4 – suspicious findings; category 5 – findings highly suggestive of malignancy [4].

Benamrane, Aribi and Kraoula [15] propose an approach for detection and specification of anomalies present in brains imagines. They use Fuzzy Logic, Neural Network and Genetic Algorithms to detect anomalies present in the images. The images are segmented by a segmentation method. Certain morphological and textural attributes are calculates on these areas such as surface, compactness, mean gray level, variance, elongation. The values of these attributes are presented to the input layer of the neural network in order to detect the anomalies and to specify its nature.

Matins et al. [16] analyzed the application of the co-occurrence matrix to the characterization of breast tissue as normal, benign or malignant in mammographic images. A Bayesian neural network is used to evaluate the ability of computed measures such as location of abnormality, radios, breast position and

type of breast tissue to predict the classification for each tissue sample.

In this paper we present a methodology to help medical diagnosis by processing mammography using image processing software and interpreting the result using a Fuzzy Logic model base on the most important factors that a specialist use to evaluate the image. At the end of the evaluation a BI-RADS level is given.

This work is organized as follows. In Section 2, we describe how we use the images processing software to obtain the input factor for the fuzzy logic model. Next, in Section 3, the results are shown. Finally, Section 4 presents some conclusions and recommendations.

2. Medical perspective

To construct a methodology for the detection of anomalies in mammography it is important to have relevant medical knowledge in this area. When a mammography is taken, different factors are taken into account in the interpretation process. Therefore it is important to investigate how the radiologist and the oncologist work.

2.1 Mammography interpretation

There are different factors in mammography interpretation process that radiologist consider when he is making the analysis and diagnosis. The most relevant are:

- Shape of the injury: the injury may vary depending on the type of injury taking into account its shape. The shape is an important issue in the mammography interpretation process due to the malignancy of the tumor is identified easily and vice versa.
- Calcifications: They are microscopic grain of calcium produced by the cells as a result of some benign or malignant process. Calcifications are important marker for breast cancer identification. Calcifications can constitute a first sign of breast cancer. Malignant calcifications may occur with or without the presence of a tumor mass. A large number of microcalcifications in a cluster increase the likelihood of malignancy. Due to their high attenuation properties, they appear as white spots on mammograms.
- Level of gray (LOG): This value indicates the degree of seriousness of the injury. A clearest gray increase the seriousness of the injury. This value could vary depending on the interpretation of each expert, who in this case is the radiologist.
- Texture: It is completely linked to the shape of the injury, as is the more homogeneous it is easier to make the diagnosis. When an injury is very

homogeneous facilitates identification it is a malignant or benign lesion.

- Asymmetry: Any asymmetry or irregularity of the two breasts should cause suspicion, since it may be a sing of cancer. The radiologist compares the same type of images of both breasts from a woman. If both images are like mirror-images, the suspicious of something anormal decrease. If the radiologist discovers any suspicious, they have to do more tests.

2.2 Image analysis

A mayor problem in image analysis is to explain more precise the visual information that they contain. Either by finding local discontinuities of the gray level [5, 6, 14] or by searching areas of the image [6, 7, 14] which present some homogenous properties, the problem can be resolved.

Most of the actual techniques used for analysis of mammographic microcalcifications, first segment the digitized gray-level image into binary regions representing the microcalcifications. Feature analysis is then performed on such binary images [12]. Since mammographic images may often be poor in contrast, lacking in the definition of microcalcification regions, the segmentation process for extraction of such regions is not reliable and accurate [13]. The proposed method uses the expert knowledge to extract information of mammographic images.

First step of the proposed method is getting values of the input factors of the fuzzy logic model. We describe them in section 2.1. To obtain these values a software that allows analyzing digital mammographic images was used. The followed steps (Fig 1) to determine the inputs for the fuzzy logic model are:

Step 1. CC and MLO views of both breasts are imported under a gray level format.

Step 2. Breast is selected and analyzed by making an histogram of gray level. Mean and standard deviation of each view are taken from their gray level histogram.

Step 3. Mean and standard deviation from same view are compared and standardized. Asymmetry percentage is obtained from this process.

Step 4. Using tools of the software, a region of interest (ROI) is then selected from a breast view. This means that ROI can contain a lesion. For this ROI a gray level histogram was created. To fill fuzzy logic variables, the mean and standard deviation of the region was used.

Step 4. By increasing the ROI, specialist set the shape of injury in accordance with the categories mentioned in section 2.4.

Step 5. The existence of calcifications is determined by an expert. If there are any calcifications, he

determinates whether they are micro o macro calcifications.

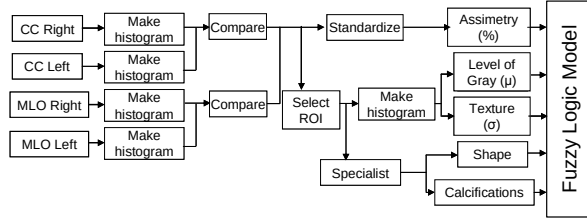


Figure 1. Image analysis sequence

2.3 Fuzzy logic

Fuzzy logic was described by Zadeh in 1965 and it has been applied to problems in various fields. Fuzzy logic produces more realistic answers by replacing the inflexible “yes/no” by a topical adjustment in form of a “more or less” and by introducing linguistic nuances into the process of decision. This can be performed by collecting statistical data and by comparing the probability density functions for the variables.

The first step in the development was the conversion of the measured patterns into a set of fuzzy variables. A set of rules was used to relate the fuzzy variables with the outcome classifications.

Membership functions for each variable were defined and the degree of each variable was divided into subsets. Fuzzyfications of each parameter was achieved by computing the membership values corresponding to each subset. Standard classification has a clear line between “yes” and “no” even when the variable is on the border of the subset.

Fuzzy rule-based systems have been widely applied to many real-world problems [8]. In these systems, information to generate fuzzy if-then rules can be classified into two kinds: numerical information obtained from sensor measurements and linguistic information obtained from expert [9].

2.4 Modeling of Variables

Five input variables were chosen for create a Fuzzy logic model. This model tries to diagnosis the level of malignity of a suspicious area. The variables are explained below:

- Shape of the injury (Shape): We used a medical classification. Shape may be located into one of four categories: There is no injury, the injury is circular, the injury is irregular or the injury has a spicules shape or star – shaped.
- Calcifications (Calcifications): This variable was modeled according to the presence or not of calcium’s grains in the image. If radiologist detects

calcifications, he determinates whether they are microcalcifications or macrocalcifications.

These two variables are determinated by the specialist and they depend on his interpretation.

- Level of gray (LOG): The scale used for modeling this input variable was the computational scale of gray. The scale takes values between 0 (black) to 255 (white). For this entry, we allocate three trapezoidal membership areas (fuzzy set), that represent predicate intervals quoted previously. We named these membership areas as Black, Gray and White. Mean gray level is used as measurement for this variable. See Figure 2.

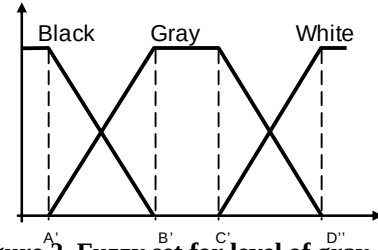


Figure 2. Fuzzy set for level of gray

- Texture (Texture): This variable use computational gray’s scale too. Three trapezoidal membership areas (fuzzy set) represent very homogeneous, homogeneous and less homogeneous. We establish that standard deviation from a gray’s histogram of the ROI is the measure appropriate.
- Assimetry (Assimetry): A percentage of similarity was used for modeling this variable. This percentage was calculated dividing the highest average value of gray’s histograms of both breasts by the lowest one. The scale takes values among 0 to 1. This fuzzy set only has two set name Symmetry and Assymetry.

The value of these three variables was obtained by analyzing each image using software to filter out the images.

The parameters of each fuzzy set were defined previously with the collaboration of the oncologist and the radiologist. For example, to define the level of gray, we showed different kind of gray in order to get a clear value, a gray value and finally a dark one.

The fuzzyfication stage assessed the input values in the membership functions which represent the behavior of fuzzy sets. The level of gray variable was representing by the equation number 1, 2 and 3.

$$\mu_{black}(LevelofGray) = \begin{cases} 1 & ; X \leq 97 \\ 1 - \frac{X-97}{100-97} & ; 97 < X < 100 \\ 0 & ; 100 \leq X \end{cases} = \begin{cases} 1 & ; X \leq 97 \\ \frac{100-X}{3} & ; 97 < X < 100 \\ 0 & ; 100 \leq X \end{cases} \quad 1$$

$$\mu_{gray}(LevelofGray) = \begin{cases} 0 & ; X \leq 4 \\ 1 - \frac{6-X}{6-4} & ; 4 < X < 6 \\ \frac{1}{2} & ; 6 \leq X \leq 7 \\ 1 - \frac{X-7}{9-7} & ; 7 < X < 9 \\ 0 & ; 9 \leq X \end{cases} = \begin{cases} 0 & ; X \leq 4 \\ \frac{X-4}{2} & ; 4 < X < 6 \\ \frac{1}{2} & ; 6 \leq X \leq 7 \\ \frac{9-X}{2} & ; 7 < X < 9 \\ 0 & ; 9 \leq X \end{cases} \quad 2$$

$$\mu_{white}(LevelofGray) = \begin{cases} 0 & ; X \leq 205 \\ 1 - \frac{210 - X}{210 - 205} & ; 205 < X < 210 \\ 1 & ; 210 \leq X \end{cases} \quad \begin{cases} 0 & ; X \leq 205 \\ \frac{X - 205}{5} & ; 205 < X < 210 \\ 1 & ; 210 \leq X \end{cases} \quad 3$$

Texture and Assymetry used the same model of equation but using different values.

The next step consisted on formulate fuzzy rules. The rules were defined with the cooperation of the expert. We divide rules into two groups. First group contains 72 rules. These rules are the result of the combination of the following input factors: shape of the injury, level of gray, texture and asymmetry. The type of injury is the output of this first analysis. This output has 4 categories: (1) no lesion, (2) benign lesion, (3) malign lesion, (4) doubtful. Examples of the rules are shown in the table 1.

Table 1. Examples of fuzzy rules for first group

Inputs				Output
Shape	LOG	Texture	Asymmetry	Type injury
Circular	Black	Very Homogeneous	Asymmetric	1
Circular	White	Homogeneous	Asymmetric	2
Irregular	Gray	Very Homogeneous	Asymmetric	4
Irregular	Gray	Less Homogeneous	Symmetric	1
Star-shaped	Black	Homogeneous	Asymmetric	3
Star-shaped	White	Less Homogeneous	Asymmetric	4
Not injury	Black	Homogeneous	Symmetric	1

The input factor of the second group of fuzzy rules is the type of injury and calcifications. The result is the Breast Image Reporting And Data System (BI-RADS®). In this group there are 12 rules. The table 2 contains some of the rules.

The process can be resumed in the Figure 3.

3. Testing and Result

This methodology has been tested on digital mammographics. We selected several cases form a public data base [17]. Each case has CC and MLO views of breasts, the expert's diagnosis and BI-RADS® classification. Cases were divided into three groups with same number of cases. The first group represents cases that have possible cancer tissue. The second group has cases where a benign tumor was detected. The last group contains cases in which there isn't an injury or calcifications. An example is shown in Figure 4 and 5.

Table 2. Examples of fuzzy rules for second group

Inputs		Output
Type Injury	Calcifications	BI-RADS®
1	0	B1
2	0	B2
3	1	B5
4	3	B3
2	1	B4

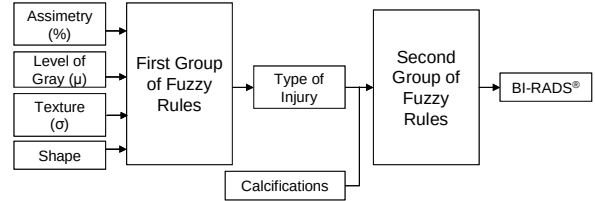


Figure 3. Representation of the sequence of the Fuzzy Logic model

Values of input factors are shown in table 3. Introducing the values in Fuzzy Logic model, BI-RADS® level was obtained. For this case level V was set by model. Level set by oncologist is the same that model calculated.

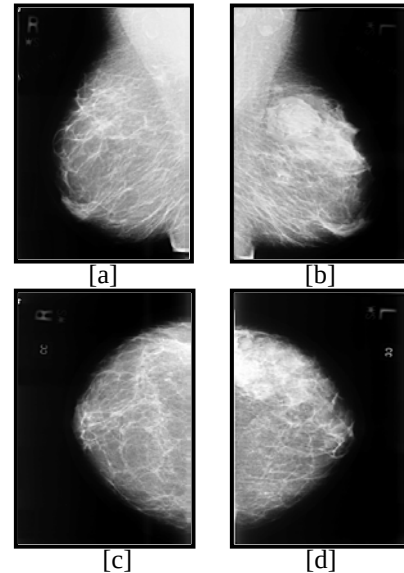
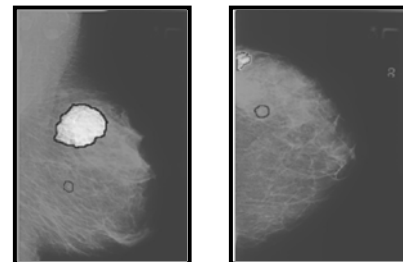


Figure 4. Original Mamographic Images: [a] Right MLO, [b] Left MLO, [c] Right CC, [d] Left CC.



[a] [b]
Figure 5. An injury is detected and selected from original images

Table 3. Values of input variables for Fuzzy Logic model.

Variable	Value
Shape	3
LOG	200.5
Texture	41.5
Asymmetry	0.9496
Calcification	1

Comparing the results after applying the method to the different cases and the previous diagnosis made by the radiologist and oncologist it shows that in 80% of the case with possible cancer the result is the same. For the cases diagnosed with possible benign tumor, we obtain a rate of 90%. With the last group 95% of the case throw the same diagnosis.

4. Conclusion and Recommendations

The primary objective of this paper is to help in the breast cancer diagnosis at mammographic. The proposed methodology helps to identify areas with possible injuries or detect calcifications. For each injury detected, Breast Imaging Reporting and Data System (BI-RADS®) level is given.

Based on the results, methodology provides support to a more detailed clinical diagnosis. This method can be used principally in hospitals without enough money for invest in modern technology because it only needs commercial software.

Comparing with others researches, variables like location of abnormality, radios, compactness and elongation were rejected according with the oncologist's experience. By the other hand, other relevant variables that doctor used for give a diagnosis employing mammographies were included such as shape of the injury and calcifications. Model not only determinates whether there are lesions or not and what kind of lesions (benign or malign), it also classifies lesion following BI-RADS® scale. Doctors' recommendations and subsequences tissue analysis are better supported.

As a future work, we propose to continue with the research by programming special software that integrates images analysis and the fuzzy logic model. Also for to make a better diagnosis we should include social variables such as age, economic level and other social aspects that increase possibility of having cancer.

References

- [1] Instituto Nacional de Estadística, Geografía e Informática (INEGI), <http://www.inegi.gob.mx/inegi/contenidos/espanol/prensa/default.asp?c=269&e=>
- [2] American Cancer Society (ASC), <http://www.cancer.org>
- [3] Imaginis, the women's health resource. <http://www.imaginis.com/breasthealth/abnormalities.asp>
- [4] Kestelman, F.P. et al., "Breast Imaging Reporting and Data System – BI-RADS®: Positive Predictive Value of Categories 3, 4 and 5. A Systematic Literature Review., *Radiol Bras* 40(3), 173–177, (2007)
- [5] Byrd, K., Zeng, J., Chouikha, M. "Performance Assessment of Mammography Image Segmentation Algorithms", *34th Applied Imagery and Pattern Recognition Workshop (AIPR)*, IEEE (2005)
- [6] Kinnard, L. et al., "Automatic Segmentation of Mammographic Masses Using Fuzzy Shadows and Maximum-Likelihood Analysis", *EEE International Symposium on Biomedical Imaging*, pp. 241 – 244 (2002)
- [7] Eltonsy, N. et al., "Error Investigation of Models for Improve Detection of Masses in Screening Mammography", *IEEE International Symposium on Signal Processing and Information Technology*, pp. 794-799 (2005)
- [8] Ruan, D., Huang, C.F., "Fuzzy Sets and Operations Research for Decision Support". *Research for Decision Support. Normal University Press*, Beijing (2000)
- [9] Wang, L.X., Mendel, J.M., "Generating Fuzzy Rules by Learning from Examples", *IEEE Trans Syst., Man., Cybern.*, vol. 22 pp. 1414-1427 (1992)
- [10] Kartalopoulos, S.V., "Understanding Neural Networks and Fuzzy Logic". *IEEE Press*. United Stated (1996)
- [11] Kosko, B.: *Neural Networks and Fuzzy Systems. A dynamical Systems Approach to Machine Intelligence*. Prentice Hall, United Stated (1992)
- [12] Dhawan, A.P., Chitre, Y., Kaiser-Bonasso, C., and Moskowitz, M., "Analysis of Mammographic Microcalcifications Using Gray-level Image Structure Features". *IEEE Trans. on Medical Imaging*, Vol. 15, pp. 246-259, (1996)
- [13] Karssemeijer, N., "Stochastic Model for Automated Detection of Calcifications in Digital Mammograms", *Image Vision Computation*, Vol. 10, pp. 369-375, (1992).
- [14] Mohammed, S., Yang, L. and Fiaidhi J., "A Dynamic Fuzzy Classifier for Detecting Abnormalities in Mammograms", *First Canadian Conference on Computer and Robot Vision (CRV)*. IEEE (2004)
- [15] Benamrane, N., Aribi, A., Kraoula, L., "Fuzzy Neural Network and Genetic Algorithms for Medial Images Interpretation". *Geometric Modeling and Imaging (GMAI)*. IEEE (2006)
- [16] Matins, L. et al., "Classification of Normal, Benign and Malignant Tissues Using Co-occurrence Matrix and Bayesian Neural Network in Mammographic Images". *9th Brazilian Symposium on Neural Network (SBRN)*. IEEE (2006)
- [17] Heath, M., Bowyer, K., Kopans, D., Moore, R., Kegelmeyer, W.P., "The Digital Database for Screening Mammography", *Proceedings of the Fifth International*

Workshop on Digital Mammography, M.J. Yaffe, ed.,
212-218, Medical Physics Publishing, (2001)

Multi-Agent Systems and Distributed AI

Multi-agent ERA model based on belief solves multi-port container stowage problem

Liu Yanbin

Wang Kangping

Guo Dongwei

Pang Wei

Zhou Chunguang

Jilin University

College of computer science and technology

ChangChun 130012, China

cgzhou@jlu.edu.cn

Abstract

ERA is the acronym of environment, reactive rules and agent; the model is a new effective multi-agent cooperation framework, which has been successfully applied in a wide range of areas. Multi-Port Container Stowage (MPCS) is a more practical problem. However, multi-port makes the container stowage problem more difficult. This paper presents a MPCS model based on time series and a multi-agent ERA model based on belief to solve the multi-port container stowage problem. The algorithm introduces the belief concept into agents and creates a multi-agent ERA model. The searching direction and strength of the agents are determined by the learning machine using belief parameter. Agent has the ability to evaluate its searching path. At last, the feasibility and efficiency of the model are validated by the experimental result.

1 Introduction

As a sub-field of distributed artificial intelligence, Agent Theory has developed fast since it was first proposed in 1970s [1]. Multi-Agent System (MAS) is a multi-knowledge formalization model, which has become a hotspot of research in recent years. Generally speaking, an autonomous agent is a system situated within and a part of an environment that senses environment and acts on it, over time, in pursuit of its own agenda and so as to effect what it senses in the future [2]. Agent has the knowledge, targets and abilities. MAS has been widely applied in many areas of artificial intelligence, for example, control process[3], mobile robot[4], air-traffic management[5] and intelligence information recycle[6] etc.

ERA model is a new multi-agent collaboration architecture of MAS, proposed by Jiming Liu[7]. ERA contains

environment, reactive rules and agent, in which agents collaborate to achieve a global solution. It has successfully solved lots of problems, for example, Constraint Satisfaction Problem (CSP)[7] and etc. But, there is no typical application of ERA for complex optimization problems.

Container stowage problem has become a hotspot of optimization research. Atkins et al presented a model depending on certain presumptions[8][9][10][11][12]; Shields constructed the first simulation method[13]; Saginaw, Perakis[14], Shin and Nam[15] managed stowage using the decision-making system. All the methods can only solve small-scale stowage problems, and many of them can only solve the single port problem.

Shift is the removal total numbers of containers in the process of transport. For reducing shifts, this paper presents a multi-agent ERA model based on belief to solve multi-port upload, multi-port unload container stowage problem, which introduces new application of ERA model. The organization of this paper is as follows: section 2 describes MPCS model; section 3 introduces multi-agent ERA model; and section 4 describes the algorithm in detail; at last, section 5 presents and compares the experimental result; we conclude with a discussion of future work.

2 The model of multi-port container stowage

2.1 Problem description

Nowadays, more than 60% maritime transport depend on container[16]. All the shipping companies use big container vessel for the large-scale economy advantage, which increases the shifts. The shift is the most expensive part in transport procedure. To formally describe MPCS problem, this paper defines some variables, as shown in table 1.

Definition 1 Container C_i and C_j are in the same vertical

Table 1. List of Symbols

Variable	Definition
A_i	Agent with ID_i
C_i	Container with ID_i
S_i	The shifts of C_i
S	System shifts
C_{if}	Original port ID of C_i
C_{it}	Destination port ID of C_i
P	Number of ports
P_i	Port with ID_i
CU_i	Number of containers uploaded at P_i
CD_i	Number of containers unloaded at P_i

row (The same x and y coordinates), and C_i is below C_j , if $C_{jt} > C_{it}$, we say that C_j adds one shift to C_i .

Definition 2 The total shifts increased by other containers in the same row are called unload shift of C_i , denoted as SD_i ; the number of necessary moves of container in vessel for uploading C_i ($0 < C_{if} < P$) is called upload shift SL_i . $S_i = SD_i + SL_i$ is called shift of C_i . The total shifts of all containers are termed as system shift S .

The algorithm will ignore some SD_i or SL_i when calculating system shift S . Because the containers with the same destination in the same vertical row can be unloaded at one time, so, when calculating SD_i among all the containers which are in the same vertical row and have the same C_{it} , only the lowest one will be calculated; and the others are ignored. Similarly, some SL_i will be ignored. The objective of multi-port stowage problem is to minimize the system shift by obeying the following constraints.

Constraint 1: All the containers in CU_i will be uploaded to the vessel at port P_i .

Constraint 2: All the containers in CD_i will be unloaded from the vessel at port P_i .

2.2 MPSC problem model based on time series

This paper partitions the three-dimension space of vessel by container size, each available unit equals to the container size. U_{xyz} stands for the (x, y, z) position, x, y and z are the coordinates of axis. U_{xyzf} and U_{xyzt} are the attributes of U_{xyz} , which stand for the earliest time of containers that have already been uploaded to U_{xyz} unit, and latest time that U_{xyz} unit containers should be unloaded, respectively.

The port ID number presents the order of the port in this shipping line, namely, the bigger the port ID, the latter in the line.

Definition 3 Container C_i and C_j are compatible, iff $C_{if} \geq C_{jt}$ or $C_{jf} \geq C_{it}$.

Depending on the U_{xyzf} , U_{xyzt} and the staggered time of container in vessel, This paper presents share policy: compatible containers can share the same available unit. The model will optimize the stowage problem based on CU_i , CD_i and share policy.

U_{xyzf} and U_{xyzt} are determined as follows, C_{xyz} is the set of containers sharing U_{xyz} unit position.

$$U_{xyzf} = \min_{i \in C_{xyz}} C_{if} \quad (1)$$

$$U_{xyzt} = \max_{i \in C_{xyz}} C_{it} \quad (2)$$

Definition 4 Container can be located at position U_{xyz} , iff it satisfies the following two conditions:

- (1) C_i is compatible with all container $C_j \in C_{xyz}$, namely, $C_{if} \geq U_{xyzt}$ or $C_{it} \leq U_{xyzf}$
- (2) C_i and any one container $C_j \in C_{xy(z-1)}$ satisfy $C_{jf} \leq C_{if} \leq C_{jt}$

Here, $C_{xy(z-1)}$ is the immediate container under C_{xyz} in the vertical direction. Constraint (2) avoids the container being hanged in the air when it is uploaded or moved to the position U_{xyz} . Because of the unloading at any moment, there may be some containers in the air in the transport process; the system can adjust the container to remove the condition in this case.

3 Multi-agent ERA model

The ERA model is a distributed MAS, similar to Swarm[17] architecture, which has the self-organization ability. Agent changes according to predefined rules of conduct, enabling the system to achieve a global solution. This paper presents MAS model based on belief, in which agent i represents container i . Agents search path according to the result of learning machine, which makes decisions by belief parameter.

3.1 Environment

The environment is defined as the available space of container vessel, so it is a three-dimensional space, U_{xyz} represents one unit. The data structure of $\mathbf{E}$ is defined as follows:

- (1)Size. The size of $\mathbf{E}$ is calculated by formula $\text{Size} = \sum U_{xyz}$, U_{xyz} is available unit.
- (2)Value. U_{xyz} can be an agent or null. Autonomy agent i moves to (x, y, z) unit, set the value according to formula $U_{xyz} = A_i$.

(3)Shift. The system shift S is an important attribute of environment. It is not only the system optimization objective and scale standard, but also the evaluation basis of agent search path. According to definition 2, S is calculated by S_i .

3.2 Agent

Agents exist in environment and are set values by their positions. Belief is an abstract concept of intelligence, which will affect agent in two ways: one is that belief direction affects agent search direction; the other is that belief strength value affects search step, the greater the value, the longer the step.

Belief adjust is the result of agent evaluates search path. It makes agent achieve objective more quickly and exactly. Let b_i stand for belief strength of agent i . α , β and γ are parameters, which instruct the change of belief. Here, α and β are real numbers; γ is a percentage, which measures the change degree of the belief. b_i is adjusted as follows:

(1)Adjust belief according to system shift S .

- If S reduces and the change degree is stronger than γ , the adjusting uses formula $b_i = b_i + \alpha$; otherwise uses $b_i = b_i + \beta$.
- If S increases and the change degree is stronger than γ , the adjusting uses formula $b_i = b_i - \alpha$; otherwise uses $b_i = b_i - \beta$.

(2)Adjust belief according to agent shift S_i .

- If S_i reduces and the change degree is stronger than γ , the adjusting uses formula $b_i = b_i + \alpha$; otherwise uses $b_i = b_i + \beta$.
- If S_i increases and the change degree is stronger than γ , the adjusting uses formula $b_i = b_i - \alpha$; otherwise uses $b_i = b_i - \beta$.

(3)System will reset belief of agent i , if belief strength is very weak, i.e. $b_i < \theta$. θ is threshold.

Belief direction can be set upward or downward, ERA model uses it and belief strength together, system will reset belief direction if the strength is less than the threshold.

3.3 Reactive rules

The ERA model's move rules will be determined as follows, agent passes its belief to learning machine by parameter, and then, learning machine returns the search direction and step.

Concretely, learning machine makes the search policy based on agent belief. Belief direction decides the search direction and strength decides the search step.

4 Algorithm description

The system creates agents and initializes their random beliefs after the environment has been constructed. Then, the system begins to run iteratively, when agents change their positions and beliefs gradually, the system records the optimal solutions.

Let x, y and z stand for the coordinates of the three-dimensional environment space respectively, MX, MY and MZ stand for X-axis, Y-axis and Z-axis maximal values respectively.

The detailed workflow of the algorithm is shown as follows:

(1)The system constructs m agents with random beliefs, positions, C_{if} and C_{it} . Compatible agents can share an unit according to definition 3, provided that agent positions satisfy definition 4.

(2)(2.1) Agent passes its belief to learning machine as parameters, learning machine returns the search direction and step.

(2.2) According to the result of (2.1), Agent calculates its anticipative z-axis value as the goal floor to move.

(3)Agent moves according to the result of step (2), the movement is based on the position exchange. Let A_i be the current treating individual, the other exchange A_j will be confirmed as follows, and the procedure should ensure that agent positions satisfy definition 4.

(3.1) Let z-axis value T_z be the goal floor of A_i , F_z as current floor.

(3.2) If T_z is greater than F_z , set A_j 's position as U_{xyz_0} , where, $x \in [0, MX], y \in [0, MY], z_0 = \max z, z \in (F_z, T_z]$, satisfying(1) U_{xyz_0} is null or (2) the A_j 's goal floor is F_z and it has not been moved.

(3.3) If T_z is less than F_z , set A_j 's position as U_{xyz_0} , where, $x \in [0, MX], y \in [0, MY], z_0 = \min z, z \in [T_z, F_z)$, satisfying(1) U_{xyz_0} is null or (2) the A_j 's goal floor is F_z and it has not been moved.

(3.4) Mark the movement of A_i and A_j as completed.

(4)Remove the empty position between the agents.

(5)Agent calculates S_i , and then the model gets the system shift S .

(6)(6.1) Set $\gamma = 10$, according to the step (1) of section 3.2, agent adjusts belief based on system shift S .

(6.2) According to the step (2) of section 3.2, agent adjusts belief based on individual shift.

(6.3) Reset a random belief, if an agent's belief weakens under the threshold θ .

(7)If arriving at designated times of iteration then stop. Otherwise, go to step (2).

5 Experimental result

The experiments use ten ports shipping line, namely, nine ports upload and nine ports unload. The vessel maximally contains 900 TEU(Twenty feet Equivalent Units), which accords with the requirements of present navigate. Each experimental result is the average value of 50 runs.

5.1 The effect of threshold θ to experimental result

Agent needs to adjust its belief in the iterative search process, there is an important parameter, threshold θ , which will strongly affect the experimental result.

The data in table 2 is the optimal solution after 200 iterations. The maximal quantity of container is 1200TEU; however, these containers can share the 900TEU vessel because of the share policy.

According to the experimental data, the system result is optimal when the threshold θ is 0.6 times of the initial belief strength. The result is irregular when container's quantity C is 400, because the random algorithm can get better result if the container scale is small. 0.6 S is the best threshold when C is greater than 500.

5.2 ERA experiment result when θ is the optimal value

There is no common data set for the MPCs problem, so, when solving the multi-port stowage problem, the user-defined data set is used, which satisfies the standard of container carrying trade.

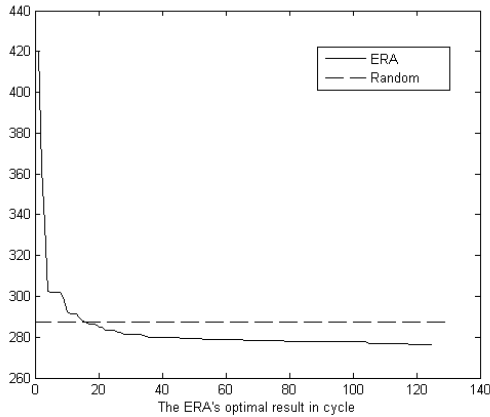


Figure 1. ERA model's optimal solution iterative curve($P = 10, C = 900, \theta = 0.6S, 50runs$)

To verify the algorithm validity, two representative cases are selected in the experiment. The container number is

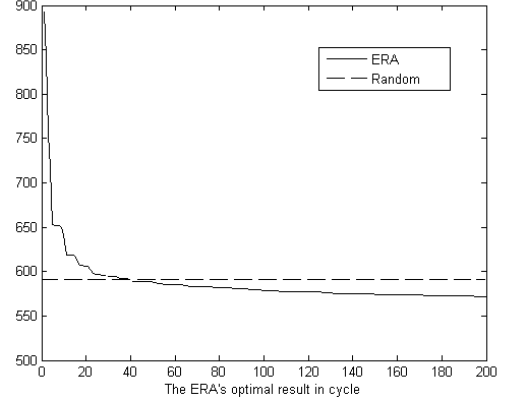


Figure 2. ERA model's optimal solution iterative curve($P = 10, C = 1200, \theta = 0.6S, 50runs$)

900TUE and 1200TEU in figure 1 and figure 2, respectively. P is 10, θ is 0.6 S in common. The results are the average of 50 runs. For comparison, this paper uses random search to find an optimal solution, which will random run 1.5 million times. The system gets the optimal result from the 1.5 million stowage cases to compare with ERA's result. Although 1.5 million is very small in the whole search space, the results are meaningful as the random search method.

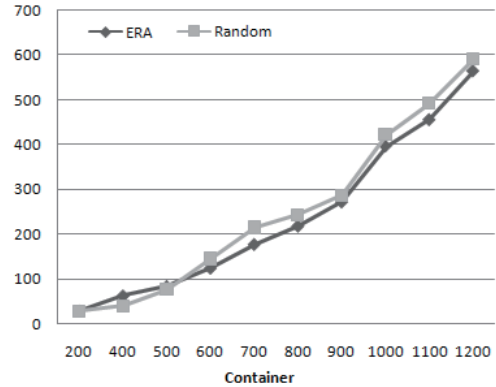


Figure 3. Optimal solutions compare of ERA model and random search

5.3 Analysis of experimental results

After 18 iterations, the optimal result obtained from ERA is better than that from the random search, in which the result is 287 after 1.5 million random searches in 13.4 hours, as shown in figure 1. After 31 iterations the result obtained from ERA is better than 590 of the 1.5 million random searches, which uses 16 hours calculating. The ERA

Table 2. ERA model's optimal solution in different type problems

Minimum Shift	θ						
	0.3S	0.4S	0.5S	0.6S	0.7S	0.8S	
C	400	76	77	71	62	40	52
	500	122	115	104	83	91	91
	600	150	129	129	124	131	145
	700	220	214	208	177	195	211
	800	295	248	227	218	231	287
	900	342	308	284	272	290	336
	1000	490	440	433	397	421	446
	1100	580	552	501	456	497	549
	1200	727	660	630	565	579	646

time is 2.1 and 3.3 minutes respectively in figure 1 and figure 2. In particular, each iteration of ERA is a local search procedure, where each agent's movement creates a feasible solution. The total solutions of one iteration approximate problem scale C .

Figure 3 is the comparison of ERA model ($\theta = 0.6S$) and random search. For solving large-scale problem, ERA's 130 to 200 iterations result is better than 1.5 million runs of random search. The advantage is not very obvious, because general random search can find the better solution in the weak constraint optimization problem. But, this algorithm can find the better solution in searching relatively smaller solution space, which proves its validity.

5.4 Comparing with other algorithms

This paper solves multi-port upload, multi-port unload container stowage problem, the model minimizes system shifts; Most of the existing models solve single port stowage problem [10][12][13][14][15][16]. [12]'s objective is shift minimization; its result is the best of this type models. So, this paper compares between ERA and reference [12].

According to the data in table 3, ERA model and SH algorithm are all square in small-scale problem. The problem scale is very small in the SH (Suspensory Heuristic) algorithm of Reference [12], ERA has solved the 1200TEU scale problem.

6 Conclusion and further work

In this paper, the multi-agent ERA model based on belief was proposed to solve the multi-port container stowage problem. First MPCs problem model based on time series was presented, which makes MAS capable of solving the container stowage problem; then, the model introduces belief concept to agent and constructs ERA rules and model. Nine ports upload, nine ports unload shipping line

Table 3. Optimal solution compare of ERA and SH

Minimum Shift	ERA	SH(TYPE-B LONG)
400	62	81
500	83	95
600	124	109
700	177	-
C 800	218	-
900	272	-
1000	397	-
1100	456	-
1200	565	-

and 900TEU vessel were used in the experiment. The analysis of the optimization solutions for different scale problems and the comparisons with other classical algorithms prove the validity of this model.

The multi-agent model presented in this paper solved MPCs problem without container stowage constraints, and future work will focus on improving this model to satisfy this problem accompanied with more constraints. Through further efforts, we hope it will become a general model in the field of discrete optimization.

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References

- [1] Gerhard Weiss: Multi-agent systems, A modern approach to distributed artificial intelligence. MIT Press, (1999) 315–333.
- [2] Petrie, C.J.: Agent-based engineering, the Web, and intelligence. IEEE Intelligent Systems and Their Applications, 11(1996) 24–29.
- [3] Jennings, N. R., Corera, J. M.: Proceedings of the First International Conference on Multi-agent Systems, ICMAS. IEEE Intelligent Systems and Their Applications, (1995) 423–430.
- [4] Neves, M. C., Oliveira, E.: A Control Architecture for an Autonomous Mobile Robot. Agents'97, ACM(1997).
- [5] Ljungberg, M., Lucas, A.: Proceedings of the Second Pacific Rim International conference on Artificial Intelligence. PRICAI'92, Korea(1992).
- [6] Julian, V, Carrascosa, C, Soler, J.: Una Arquitectura de Sistema Multi-Agente para la Recuperación y Presentación de Información. IV Congreso ISKO-España, EOCONSID'99, (1999) 291–296.
- [7] Jiming Liu, Han Jing, Y.Y. Tang.: Multi-agent oriented constraint satisfaction. Artificial Intelligence, S0004-3702(01)00174-6/FLA, (2004) 1–41.
- [8] W.H. Atkins: Modern marine terminal operations and management. Boyle, Oakland(1991).
- [9] J.G. Kang and Y.D. Kim: Stowage planning in maritime container transportation. Journal of the Operational Research Society, 53(2002) 415–426.
- [10] Daniela Ambrosino, Ann Sciomachen and Elena Tanfani: Stowing a containership: the master bay plan problem. Transportation Research Part A, 38(2004) 81–99.
- [11] R.C. Botter and M.A. Brinati: Stowage container planning: a model getting an optimal solution. IFIP Transactions B, Knowledge-based hybrid systems, (1992) 217–229.
- [12] M. Avriel M. Penn and N. Shpirer etc.: Stowage planning for container ships to reduce the number of shifts. Annals of Operations Research, (1998) 55–71.
- [13] J.J. Shields: Container-ship stowage, a computer aided preplanning system. Marine Technology, (1984) 370–383.
- [14] D.J. Saginaw and A.N. Perakis: A decision support system for containership stowage Planning. Marine Technology, 26(1989) 47–61.
- [15] J.Y. Shin and K.C. Nam: Intelligent decision support system for containership auto-stowage planning. Journal of Korean Institute Port Research, (1995) 19–32.
- [16] Dirk Steenken, Stefan VoB and Robert Stahlbock: Container terminal operation and operations research. a classification and literature review, (2004) 3–49.
- [17] Swarm simulation system.
<http://www.swarm.org/index.html>

Predictive Negotiation Strategy Applied to Multiagent Meeting Scheduling Problems

Gisela Medina

Tecnologico de Monterrey Campus Monterrey
Center for Intelligent Systems
Monterrey N.L., Mexico
gismram78@gmail.com

Leonardo Garrido

Tecnologico de Monterrey Campus Monterrey
Center for Intelligent Systems
Monterrey N.L., Mexico
leonardo.garrido@itesm.mx

Abstract

Although there is several negotiation strategies applied to different negotiation problems, we are proposing the implementation of the modified algorithm of predictive behavior developed by Jakub Brzostowski and Ryszard Kowalczyk ([4]). We demonstrate that this modified algorithm is well performed for this kind of problems using a different representation problem. The performance is measured using the Playbook technique where our modified strategy is compared with several ones which are well performed too (Crawford and Veloso in [2]).

1. Introduction

Applying specific strategies and algorithms of *Multiagent Learning* in problem solving in distributed multiagent environments is a very interesting challenge to investigate and that is why there are many valuable and relevant research work that has emerged. Many of the problems we face all days in home, work, industry, city and school are kinds of problems in where many entities interact to reach an agreement favorable to all participants. The problem definition consists of a set of meetings m , a set of agents a and a set of time slots c which represents the calendar of each agent, each one with a particular density d (a calendar is a set of free/busy slots used in a negotiation process; a calendar density is the number of busy slots in the calendar). The challenge of the multiagent meeting scheduling problem is to schedule the maximum number of meetings resulting in high utility for all agents. There exist strategies used by agents to make negotiations more efficient to allow participants act in a cooperative or competitive way to increase their global or local utility. In this paper we demonstrate that the use of a predictive strategy, originally used in buyer-seller domains, can increase the global utility of all

agents through the use of a predictive model which needs the learning of others agents' calendar preferences.

2. Related Works

In meeting scheduling research work we found the work of Crawford and Veloso [2] which used the concept of playbooks (a set of strategies in which select the best one in every decision point) to select the best strategy. In their paper they used two strategies, *OfferKB* and *Availability declarer*. They demonstrated the convergence to the best strategy using their learning approach in meeting scheduling problems using their playbooks approach. Another work based on the multiagent system for meeting scheduling problems was proposed in [1]. They proposed a new distributed approach based on the DRAC model (distributed reinforcement of arc consistency) to solve dynamic meeting scheduling problems. We work with static assumptions like static preferences to facilitate the use of our algorithm but there are works like this which eliminate assumptions to be close of the real world. There is many research work about meeting scheduling but every one considers different assumptions and others work in different domains. The user profiling and calendar preferences is another research work which help the research in meeting scheduling like the work of Jean Oh in [8] and Leffert in [6]. Learning in dynamic environments is a challenge that not many people is working right now in the meeting scheduling domain; however many research in negotiation strategies using persuasion and based-utility functions (like in [3]) is done in different domains.

3 Modified Predictive Negotiation Strategy

There are some assumptions related to negotiation strategies. When participant makes a proposal he considers availability and preferences; if no slot available then he does not propose anything (depends on negotiation strategy), if the

agent's calendar is almost full, it is not allowed to propose slots not available although the slot is preferred. Besides negotiation strategies consider finalization conditions to avoid loops in the negotiation process (maximum of rounds allowed). Others assumptions are that the calendar is composed of a fixed amount of slots and that all agents have static preferences (no change over time). The negotiation protocol is described as follows. The organizer starts the negotiation sending a proposal to all the invitees; every invitee receives the proposal and according to his preferences he makes a counter proposal, indicating the best and worst slots. The organizer analyzes all proposals sent by all invitees and if he finds a common slot then he sends a confirmation to schedule the meeting in that slot. If the organizer does not find any intersection (a common slot) then he generates a new proposal. The proposal may contain one or more slots of time. The negotiation finishes when an agreement is reached, no new proposals are sent by invitees or when the maximum number of rounds is reached.

3.1 Changes to the original predictive strategy

We use the article written by Jakub Brzostowski [4] for the development of the predictive strategy. He proposed a predictive model based on decisions taken *through* the negotiation, i.e., there is no previous knowledge about the proposals done by partners. He designed the algorithm to be used in seller-buyer domains where the seller tries to make proposals to the buyer in order to avoid his own lost of money (keep it into a range of good proposals). In the other side, the buyer tries to make also good proposals when he negotiates with his private preferences.

We modified the algorithm of Jakub Brzostowski to be applied into the meeting scheduling problem. The performance of this algorithm depends of several factors: total of invitees for meeting, private preferences, calendar density and negotiation strategy. To adjust the algorithm to our meeting scheduling problem we had to change the problem representation and consider some characteristics of the problem. We describe the list of these characteristics as follows.

1. Type of proposals. This is the main difference between the two problems, the meeting scheduling and the seller-buyer. While in the seller-buyer problem the proposals are numeric values, in our meeting scheduling problem are slots of time. Also, in our domain we could have multiple proposals (multiple slots proposed in the same negotiation round) while in the other only one value is required.
2. Multiple participants. We considered many participants involved into negotiation while in the problem described by Brzostowski considered only a seller and a buyer as agents in the negotiation. We had to design an algorithm to consider the effect of the proposals done by more than 2 participants.
3. Range of allowed proposals. The preferences shown in the article [4] are delimited by a minimum and a maximum value and also a specific range to make proposals. In our domain we have a limited number of slots (default value of 50, they represent hours, 5 days of 10 hours). To make proposals agents have to check their preferences and availability. In our domain we have to validate that each proposed slot be available and free.
4. Preferences. In the meeting scheduling problem the preferences are given by a set of hours where the hours more left are more preferred. For instance, a preference list for a specific agent could be [18, 17, 16, 15, 14, 13, 12, 11, 10, 9, 8] which means the more preferred slot is the hour 18 and the less preferred is 8. Here the assumption is that the preferences are static, that is, they don't change over time. The seller-buyer problem domain has also a specification of preferences, it could be a range of values where the seller or buyer can negotiate, for instance a range [100.90, 304.1] that means the seller only can make proposals into this range. In the meeting scheduling domain the participants has a number of slots to negotiate. The main change in our modified predictive algorithm is the way to measure the proposals: it is done by getting the preference value of the slot where higher value means higher preference. That is why the prediction of the other agent's learned preferences must be almost accurate to get better results in the modified predictive strategy.
5. Learning algorithms used to measure meeting scheduling proposals. To measure each other agent's proposal, the predictive algorithm needs to know the other agent's calendar. In our research we propose to use an algorithm to learn other preferences helped by the learning other calendars algorithm. Specifically we need to know the preferences of the other agents, previous to the generation and use of the predictive model. The work of Leffert in [6] was adapted to learn other calendars: we use a Bayesian approach to update beliefs of free and busy slots in the other agent's calendar based on the counter proposals (confirmed, refused, accepted) received by the agent. The work of Jean Oh in [8] was adapted to learn other agent's preferences using a supervised learning method based on K-nearest neighbor to get a preference-based function. It uses the counter proposals sent by other agents and counts the confirmed, refused and accepted slots.

6. Constraints. The meeting scheduling problem has constraints in terms of availability and calendar density. It means that although the slot of time is preferred it could be not available. The calendar density is other constraint in our domain because if the agent's calendar density is high then the number of proposals is reduced and therefore the negotiation is terminated early.
7. Strategies. All agents involved in a negotiation have their own strategies and they do not negotiate with the same strategies all the time.

3.2 Understanding the Predictive Algorithm

Typically, there are two main factors that influence the agent negotiation behavior: time and imitation. The agent's behavior depends to some extent on time and it depends to some extent on imitation. The goal is to find, in terms of functions, the degree of influence of both factors. Brzostowski in article [4] explains this method. During negotiation the available information is the set of own and other proposals, so this input data is used to generate the model of the predictive strategy. To measure the time dependency factor, the *difference method* is used: given a sequence of other agent's proposals, it calculates the difference value between continues proposals and gets the *deltas* values. Over each set of delta values obtained it is possible to get more set of deltas; in this way we have k levels of deltas. The algorithm counts the number of positive and negative delta values and also the sum of all positive and the sum of all negative deltas. With these values it gets the time dependency factor as specified in the following equations.

$$D_k = \frac{|\frac{\sigma_p^k}{m_p^k} + \frac{\sigma_n^k}{m_n^k}|}{\max|\frac{\sigma_p^k}{m_p^k}|, |\frac{\sigma_n^k}{m_n^k}|}$$

$$D_t = \sum_{k \leq m-1} w_k D_k$$

$$w_k = \frac{\max|\sigma_p^k|, |\sigma_n^k|}{\sum_{j=1}^{m-1} \max|\sigma_p^j|, |\sigma_n^j|}$$

To find the behavior dependency factor the algorithm uses again the history of own and other agent's proposals. It measures the relation between the own and other proposal as $r_i = \frac{\Delta^1 b_i}{\Delta^1 s_i}$ (using only the first order difference). For all m concessions the algorithm gets the factor r as follows:

$$r = \sum_{i=1}^m w_i r_i$$

The weights w_i may increase with the value of index i which means that the later offers reflect better the current state of our partner's behavior and therefore, the higher the values of indices i the more important is the quotient r_i . The behavior dependency factor is then calculated as:

$$D_b = \frac{D_1 + D_2}{2}$$

Where D_1 is calculated as follows and D_2 is calculated in the same way that D_t .

$$D_1 = h(r) = \max(1, \min(0, 0.5r))$$

The predictive strategy tries to find the best next proposal using the next formulas that define the model (H_m represents the history of all proposals, s_m represents own proposal and b_m represents other agent's proposal).

$$\hat{b}_{m+1}^t = f_t(H_m) \quad (1)$$

$$\hat{b}_{m+1}^b = f_b(H_m, s_{m+1}) \quad (2)$$

$$f(H_m, s_{m+1}) = \frac{D_t}{D_t + D_b} \times \hat{b}_{m+1}^t + \frac{D_b}{D_t + D_b} \times \hat{b}_{m+1}^b \quad (3)$$

Now we calculate $\hat{b}_{m+1}^t$ and $\hat{b}_{m+1}^b$ according to the next formulas.

$$\hat{b}_{m+1} = f_t(H_m) = b_m + \sum_{j=1}^k \Delta^j b_{m-j} \quad (4)$$

$$\hat{b}_{m+1} = f_b(H_m, s_{m+1}) = b_m + \frac{\Delta b_{m-1}}{\Delta s_{m-1}} (s_{m+1} - s_m) \quad (5)$$

The next step is to use the multistage control process which is a combination of prediction with decision-making about the next offers. Using the model we get $s_{m+1}, s_{m+2}, c, \dots, s_{m+l}$ sequences knowing the $b_{m+1}, b_{m+2}, c, \dots, b_{m+l}$. So we have a sequence of the form: $s_{m+1}, s_{m+2}, \dots, s_{m+k}, s_{m+k} + c, s_{m+k} + 2c, \dots, s_{m+k} + (l-k)c$. To find the concession factor we solve the equation:

$$c : \min(|\hat{b}_{m+l} - (s_{m+k} + (l-k)c)|)$$

Next step is to generate a set of optimal sequences, based in the concession factor c .

$$\begin{aligned} & s_{m+1}^1, s_{m+2}^1, \dots, s_{m+l}^1 \\ & s_{m+1}^2, s_{m+2}^2, \dots, s_{m+l}^2 \\ & \dots \end{aligned}$$

$$s_{m+1}^n, s_{m+2}^n, \dots, s_{m+l}^n$$

Next step is to calculate the influence of each optimal sequence as follows.

$$offer^j = s_m + \text{sgn}(s_{m+1}^j - s_m^j)(|s_{m+1}^j - s_m^j|)^{w_t} \times E \quad (6)$$

$$\text{where } E \text{ is defined as } \text{sgn}\left(\frac{s_{m+l}^j - s_m^j}{l}\right) \left|\frac{s_{m+l}^j - s_m^j}{l}\right|^{w_b}$$

$$offer = offer^1 w_1 + offer^2 w_2 + \dots + offer^n w_n \quad (7)$$

where the weights w_j are proportional to the utility that can be gained by applying j -th sequence of actions s^j according to the prediction.

In conclusion the predictive model is composed of three phases. First phase is to get the training data, that is, the history of proposals H . The predictive agent uses any other strategy instead of the predictive to collect this information. The second phase is to generate the model calculating the time and behavior dependency factors and the concession factor. The third phase is to use the model generating a set of optimal sequences using the concession factor and get the final next predicted proposal. Every time a new proposal is obtained, it is added to the history of proposals and use it next time to calculate next proposal.

3.3 Generating the model

Generate the model means to calculate the factors D_t and D_b for each partner and represents the proportion in which the participant is using a time or behavior tactic respectively. The algorithm uses the information related with all proposals done by organizer and invitees in the negotiation at that specific time (just before to reach the round *modelSize*). The method of differences is used to calculate both factors as proposed by Brzostowski. The next steps simplify this first phase in the algorithm.

1. Learning others preferences. This step is done before running the negotiation and it is useful for the agent to predict new proposals done for participants. The algorithm *Learning Others Calendars* should be executed previously to have a preferences learned model of all other agents. This learned other agent's preferences are used after to measure their proposals. If the agent does not have a learned model then a default model is used.
2. Initialize the model using data collected in the first *modelSize* rounds. The data is conformed of all proposals done by the agent and the other agents. The value of each own and other proposal is obtained from

the own preferences and the learned preferences respectively. This means that all proposals are measured using the preference's value of the slot according to the learned preferences (if it is learned previously for the agent) or static preferences (if no information was generated).

3. Calculate differences in the way is described in [4] using only partners' proposals and the Hamiltonian distance. Every proposal is represented as a set of bits where 1 represents busy slot and 0 represents free slot. The Hamiltonian distance is defined as follows: given two strings of bits, s_1 and s_2 , both of size n , then for every bit i in every string, do the operation where if two bits are equal then the resulting bit is 0 and if two bits are different then the resulting bit is 1.
4. Calculate D_t for each partner. D_t is the proportion in which the other agent is using time tactics. The equations are used without variation because our algorithm converts the proposal format into a numeric value.
5. Calculate D_b for each partner. D_b is the proportion in which the other agent is using behavior tactics. The equations are used without variation because our algorithm converts the proposal format into a numeric value.

3.4 Using the model

Proposing new offers requires that the slots selected are available and selected according the model previously learned. For predicting new other agent's proposals is not necessary to validate availability because the agent only has their learned preferences and not the complete calendar. When the proposals generated are from the agent then it is necessary to validate the availability because he knows his own calendar.

When the model is generated, the next proposals are done according to this one. The next steps are followed to generate the next proposal.

1. Get concession factors based on random sequences. The sequences are generated randomly using the other agent's predicted proposals and the agent proposals. In this step the algorithm uses the model generated in previous section to get the predicted proposal. For every proposal in the random sequence calculates the concession factor based on own and other proposal. After that, get the lower concession factor value between all the proposals in the random sequence.
2. Make optimal predictive sequences. The goal of this step is to generate a set of predicted sequences and added to the data model. These sequences are optimal

because they use the concession factor calculated in previous step. It's done in this form because we want to have a proposal predicted in $r + t$ round where t is the long of this set of predicted sequences. The algorithm does n series of predicted sequences and based on these the algorithm could generate a predicted proposal. The values of t and n are the key for generating good predicted series.

3. Generate a predicted next proposal. In this step the optimal sequences obtained in the previous step are used to generate the proposal. For every optimal sequence, the equation 6 is applied, using s_m as the value of the agent's proposal measured according to his preference value. At this time the utility is calculated as the sum of utility calculated for our agent and the other agents (utility is calculated as the difference of preference value of final and initial proposal in the optimal sequence divided by the total of proposals in the sequence). Finally, summarize every value given by each optimal sequence as defined in the equation 7 where w_i values are the utilities calculated previously. The multiplication of the offer and the utility value is calculated as follows. First of all the utility is expressed in terms of percentage (between 0 to 1), where the utility is divided by the total utility (sum of each optimal sequence utility). The *offer* term in the equation is a string of bits and the w is the utility, so the multiplication is represented as the mutation of one bit, that is the substitution of one bit by another. The bit to change is given by $w \times n$ where n is the total of slots (in our case, 50), so the resulting value is the position of the bit to change: if that slot is not available then the mutation is not applied. Finally to execute the sum of all offer values it is necessary to generate a new proposal taking into account the slots more selected in all the offers and more preferred. Remember that our agent is negotiating with more than one agent, so for every other agent, our agent makes predicted proposals; the algorithm selects one proposal randomly between all proposals predicted using other agents.

4 Experimentation

First of all the testbed where we develop and experiment the negotiation strategies are based on a framework like [5]. We use [2] and [7] article to implement the playbook that selects the best strategy. The learning technique used for predictive algorithm is implemented as a modification of the algorithm described by [6]. Finally the paper written previously to the one we used as a reference for this paper is [3] that describes some other strategies capable to be modelling and predicted.

There is a factor which influences the performance of the Predictive strategy, which is having learned preferences of the other participants because this strategy depends of the quantification of the slots proposed by the invitees, so the organizer (whom is using the predictive strategy) could estimate appropriately the value of them.

1. Using learned preferences. Using learning other preferences algorithm to predict better the next proposals of invitees. Having the preferences learned allows us to estimate appropriately the next proposals which depends on the value of the slots to determine which of them are better and which ones could be eliminated from the generation of the next proposal; obviously the result depends of the the percentage of learned preferences.
2. Using the static model of preferences. When the participant have not learned the preferences, it is used the static preferences, [9, 10, 11, 12, 13, 14, 15, 16, 17, 18]. This should affect the performance of the strategy because the prediction of next invitee proposals could be wrong.

The parameters used in the predictive negotiation algorithm are shown in the table 1.

Parameter	Value
Org. Density	[0, 30]
Inv. Density	[0, 30]
Org. Preference	[9, ..., 18]
Inv. Preference	Vary
Num. Invitees	[1, 3, 6]
Org. Strat. to prepare exp.	<i>Offer</i> (3, 5, 2)
Org. Strat. to prepare exp.	<i>Time</i> (3, 0.2, d , p)
Org. Strat. (predictive model)	<i>Time</i> (3, 1, d , p)
Inv. Strategies	All

Table 1. Use the time tactic to generate the experience data. This table contains all the parameters needed to run the experiment for predictive tactics.

Using the information described in 1 we show the algorithm that generates all the test cases used to learn calendars and preferences. Parameter s_o is the strategy of the organizer that for the purpose of this experiment must be fixed to compare with the predictive strategy. Parameters d_o and d_1 are the calendar densities of organizer and invitee respectively. Parameter p_o is the preference of the organizer that must be fixed. In general we have 1 organizer and 6 invitees. We have a total of $6 \times 3 \times 6 = 108$ meetings to get experiences.

5 Evaluation

We are going to analyse the results of running the predictive negotiation and the impact in performance when the invitees are using any other strategies. The first analysis is about the utilities obtained when the organizer is using the *OfferNKB(3,5,2)* strategy to acquire data for learning preferences. The preferences learned are shown in the figure 1; we see that there are few preferences good learned and this could affect the performance of the predictive strategy.

		9	10	11	12	13	14	15	16	17	18	
L	1080	600	960	840	780	720	900	660	1020	540		10
R	540	600	660	720	780	840	900	960	1020	1080		
N	0		0.4	0.3		0.7		0.4			0	0.7

L	1020	1080	900	780	660	720	960	840	600	540		11
R	540	600	660	900	960	1020	1080	720	780	840		
N	0.4	0.4	0.8	0.4	0.7	0.7	0.7	0.7	0.5	0		0.5

L	660	600	900	960	840	1020	540	780	720	1080		12
R	720	780	840	540	600	660	900	960	1020	1080		
N	0.4	0.6	0.5	0.5	0.7	0.6	0.6	0.3	0.1			0.6

L	1020	600	960	1080	900	660	840	540	780	720		13
R	720	780	840	900	960	1020	1080	540	600	660		
N	0.4	0.2	0.7	0.6	0.8	0.5	0.5		0.2	0		0.5

L	780	660	600	540	1020	720	1080	960	840	900		14
R	900	960	1020	1080	540	600	660	720	780	840		
N	0.1	0.4	0.6	0.8	0.7	0.7	0.6	0.3	0.8	0		0.5

L	720	780	1080	840	1020	960	540	900	600	660		15
R	900	960	1020	1080	720	780	840	540	600	660		
N	0.5	0.5	0.8	0.6	0.7	0.5	0.8	0.2				0.8

Figure 1. Learned vs. Real preferences for each invitee: I0, I1, I2, I3, I4 and I5. This preferences were learned from the results of the experiments where the organizer is using the strategy *OfferNKB(3,5,2)* and the total of meetings generated were 108 where all the invitees are taking different strategies. The meetings also have different number of invitees.

We are comparing some strategies vs. the predictive; in the figure 2 we can see the performance of the Predictive strategy that is good in comparison with the others. The first chart in the figure shows the utility achieved by organizer and invitees. Using Predictive strategy we notice that utility of invitees are higher than organizer, this is coherent with the concept of the Predictive, that is to make proposals

according to the predicted proposals of the invitees; then the objective of the organizer is to try to reach an agreement although he sacrifices his preferences. The second chart shows the average behaviour of both organizer and invitee and we notice that the utility is higher than the others. This results are obtained using only one meeting by strategy and one invitee by meeting. The preferences of both are different and the density is taken from the range $[0, 30]$. The strategy for the invitees in all the meetings is always *Time(3, 0.3, d, p)* while the strategy of organizer in each meeting changes according to the charts.

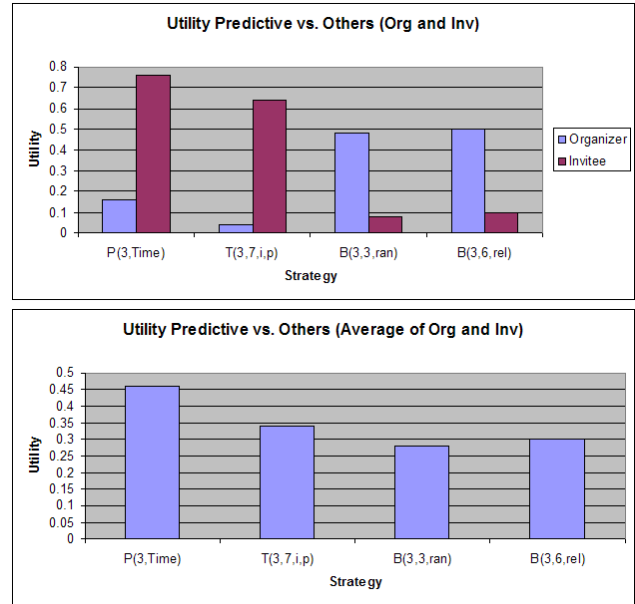


Figure 2. It shows the first try using Predictive vs. Other strategies. The first chart shows the utility for organizer and invitee. The second is averaging both to quality the performance of the Predictive.

6 Conclusions

In this paper we shows the behaviour of the modified predictive strategy againts others good strategies. It results in a good performance and high utility values. There are several negotiation strategies that could apply to multiagent meeting scheduling problems and it is a challenge to demonstrate that some of those at suitable to be applied for this kind of problems. There is no much research on meeting scheduling problems but we propose a new modified algorithm that

performs well against some others. In future work we are going to adapt this new modified predictive algorithm to be applied in dynamic environments.

References

- [1] X. D. Ahlem Ben Hassine. Agent-based approach to dynamic meeting scheduling problems. 2004.
- [2] E. Crawford and M. Veloso. Learning to select negotiation strategies in multi-agent meeting scheduling. In *Proceedings of 12th Portuguese Conference on Artificial Intelligence*, Computer Science Department, Carnegie Mellon University, 2005.
- [3] C. Hou. Modelling agents behaviour in automated negotiation. 2005.
- [4] R. K. Jakub Brzostowski. Predicting partners behaviour in agent negotiation. 2006.
- [5] V. L. Jiaying Shen, IngoWeber. Oar: A formal framework for multi-agent negotiation. 1999.
- [6] A. Leffert. Learning others calendars. 2006.
- [7] B. B. Michael Bowling and M. Veloso. Plays as effective multiagent plans. enabling opponent-adaptive play selection. In *American Association for Artificial Intelligence*, 2004.
- [8] J. Oh and S. F. Smith. Calendar assistants that learn preferences. 2005.

A Vectorial Approach on Flock Traffic Navigation

Carlos Astengo-Noguez
ITESM Campus Monterrey
Av. Eugenio Garza Sada 2501 Sur
64849, N.L., Mexico
castengo@itesm.mx

Luis Miguel Garca Velzquez
ITESM Campus Morelia
Camino a Jess del Monte S/N
58350, Mich. Mexico
luis.miguel.garcia@itesm.mx

Abstract

In a city where vehicles will travel without the aid of traffic lights finding partners to make a flock will be a very important issue. Flock Traffic Navigation assumes that vehicles can communicate with each other in order to negotiate a common bone-structure using their actual positions and destinations. In this paper we present an original method based on vector analysis to determine possible best-partners. The cross product is used to determine direction meanwhile the dot product can be used to discriminate among neighbors choosing for travel partners. The meeting and the splitting point are founded using centroids.

1 Introduction

Statistics show that traffic congestion in big cities is getting worse. The average a commuter spends per year in traffic delays is superior to 47 hrs and vehicles' engines waste an estimate of 2.3 billion gallons of fuel.

According to the 2005 Mobility report [5] a study of the largest 85 U.S. cities, the annual time spent waiting in traffic has increased from 16 hours per capita to 46 hours per capita from 1982 to 2005. In the same period, the annual financial cost of traffic congestion has swollen from \$14 billion to more than \$63 billion (in 2002 US dollars). Each year, Americans burn approximately 5.6 billion gallons of fuel while idling in heavy traffic [6].

With the increasing number of vehicles being incorporated to the traffic in big cities and the pollution generated by them, it seems like a good idea to look for alternative ways of controlling the flow of vehicles.

The 2005 Mobility Report [5] start saying: "Congestion continues to grow in America's urban areas." "...in 2003, congestion caused 3.7 billons hours of travel delay and 2.3 billions gallons of wasted fuel."

Current methods for enabling traffic to flow through intersections include building overpasses and installing traffic

lights. However, the former is only worth the cost at the most congested intersections, and the latter can be quite inefficient, often requiring cars to remain stopped even when no cars are present on the intersecting road [3].

On the sections 2 and 3 we will describe a solution presented in 2006 [1] and examine a critical issue that can be improved using simple vector analysis.

2 Flock Traffic Navigation

Flock traffic navigation based on negotiation (FTN) is a new approach for solving traffic congestion problem in big cities. It is a combination of Artificial Intelligence (AI), Multiagent Technologies, Mathematics and Nature-based algorithms [2].

In FTN vehicles could navigate automatically in groups called "flocks"[1]. Birds, mammals and fish seem to organize themselves effortlessly by traveling in flocks, herds and schools that move in a very coordinated way. A flock consists of a group of discrete objects moving together. In the case of bird flocks, evidence indicates that there is no centralized coordination, but a distributed control mechanism.

Flock traffic navigation allows the coordination of intersections at the flock level, instead of at the individual vehicle level, making it simpler and far safer. To handle flock formation, coordination mechanisms are issued from multi-agent systems.

The mechanism to negotiate [1] starts with an agent who wants to reach its destination. The agent knows an a priori estimation of the travel time that takes to reach its goal from its actual position if he travels alone. In order to win a speed bonus (social bonus) he must agree with other agents (neighbors) to travel together at least for a while. The point in which the two agents agree to get together is called the meeting point and the point where the two agents will separate is called the splitting point. Together they form the called "bone" structure diagram.

Individual reasoning play the main role of this approach. Each agent must compare its a priori travel time estimation

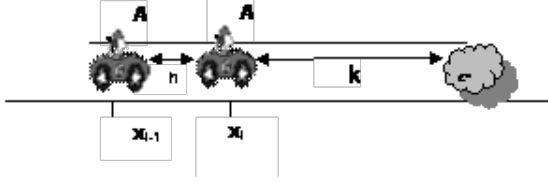


Figure 1. Vehicles move in one lane and only one direction.

versus the new travel time estimation based on the bone-diagram and the social bonus and then make a rational decision.

Decision will be taken according if they are in Nash equilibrium (for two of them there is no incentive to choose other neighbor agent than the agreed one) or if they are in a Pareto Set [7].

If both agents are in Nash equilibrium, they can travel together as partners and can be benefited with the social bonus. In this moment a new virtual agent is created in order to negotiate with future candidates. Agents in a Pareto Set can be added to this “bone” diagram if its addition benefits him without making any of the original partners worse off.

3 Limiting Partners

We investigated simple ways of limiting the number of possible partners with whom it could be beneficial to negotiate for an eventual trip together.

One possible way is to find an upper limit in terms of the starting and ending compatibility points of two given possible partners. Of course, the closest their starting and also their ending points, the greater the possibility of a beneficial joint trip is. We formalize these ideas in the following.

Assume a simple model of one lane, one direction (fig:car1).

Let suppose Agent A_i is at position x_i and A_j is at position x_{i-1} over the same lane. Let h represents the distance between them, and k represents the distance to an hypothetical splitting point S . Because this is a one dimensional analysis S must be the goal for at least one of the agents.

Individually, agent A_i must travel a distance of k units in order to reach its goal. Without losing generality, assume S is the goal for A_i ; supposing constant speeds (1 distance unit / 1 time unit). k will also represent the time for A_i to reach its goal.

Now, assume there is a second agent A_j who wants to reach the same point S . A_j is at a distance of h units from A_i .

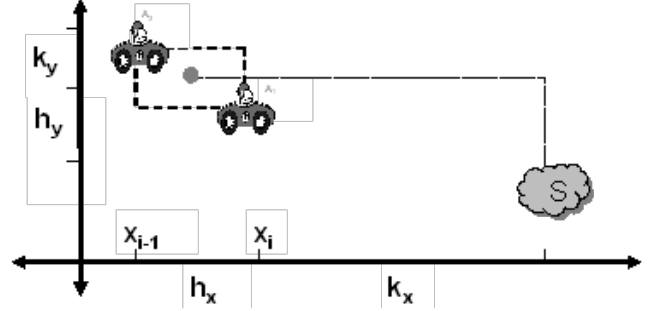


Figure 2. Vehicles move in Euclidean R2 space.

Assuming there is a social bonus ($b > 0$) for travelling together, which means that the “city” allows the join travelers to increase speed by a factor b , this is the selfish benefit of travelling together.

Let us review some cases:

Case 1: Waiting for a Partner

In this approach Agent A_i will wait for A_j to be at x_i . The travelling time or utility function for agent A_i now is: $h + k/b$, so A_i agrees to wait so long as the following inequality remains true.

$$h + \frac{k}{b} \leq k$$

Case 2: Close Encounter

If two directions are allowed, then agent A_i moves to the left in order to meet agent A_j earlier. Because we are assuming constant speed, they will reach the geometrical centre at $h/2$. The new agreement point will comply with:

$$\frac{h}{2} + \left[\frac{k + \frac{h}{2}}{b} \right] \leq k$$

Therefore

$$h \leq 2k \left[\frac{b-1}{b+1} \right]$$

Thus, cooperation is convenient for agent A_i if the distance to agent A_j is not greater than $2k \left[\frac{b-1}{b+1} \right]$. A similar analysis must be done for agent A_j .

Now assuming a more complex situation, both agents are in the Cartesian plane R^2 . They travel with constant velocity and they can earn a social bonus if they decide to cooperate.

If both agents agree for an early encounter, then the new agreement will based in:

$$\frac{h_x}{2} + \frac{h_y}{2} + \left\lceil \frac{k_x + \frac{h_x}{2} + k_y + \frac{h_y}{2}}{b} \right\rceil \leq k_x + k_y$$

Again we find a similar expression for the heuristic,

$$(h_x + h_y) \leq 2(k_x + k_y) \frac{[b-1]}{[b+1]}$$

So Agent A1 will agree if the Manhattan separation distance is no greater than $2(k_x + k_y) \frac{[b-1]}{[b+1]}$. Agent A2 must be analyzed in the same way

If constant velocity is not allowed, the problem is transformed into a variational calculus problem, each agent must build a function:

$$\vec{v} = \frac{ds}{dt}$$

where,

$$dt = \frac{ds}{\vec{v}}$$

and the time must be calculated as:

$$t = \int_0^s \frac{ds}{\vec{v}}$$

Finding appropriate traveling partners can be a complex problem. Once the broadcast radius was settled, the negotiation phase starts by calling for partners. Agents must decide, using game theory, which neighbors will be the best candidates and then start asking each other in order to determine Nash equilibrium points.

In our approach we will reduce the number of neighbors to negotiate using vectorial inner product in order to call just the neighbors that are close according to the angle between them; also taking care about the direction in which each agent is traveling.

4 A Vectorial Perspective.

The basic Problem of two agents and its bone structure diagram could be transformed as vectors on traditional R2 Euclidean Space.

The distance between the starting (or actual) point and the target-point for each agent can be easily transformed as vectors in a free-space. In order to apply vector results we need a common origin point: the meeting point.

The meeting point is calculated as in [1] using the middle point (centroid) between the line that separate both agents. Then the vectors are moved using rigid transformations.

Once the agents' vectors are in the common origin the vector inner product is calculated and the vector cross product is calculated.

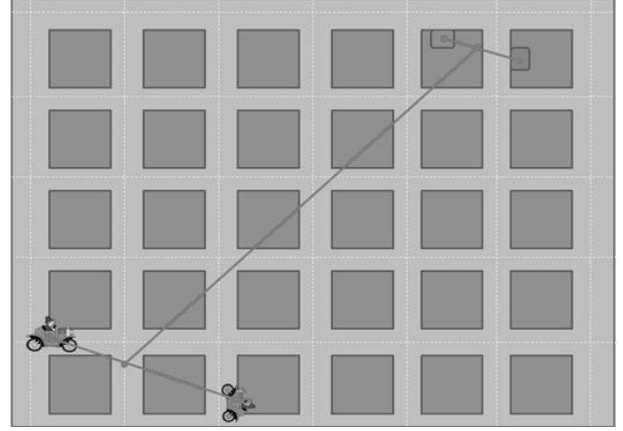


Figure 3. The classical problem of two agents on a city and the bone structure associated.

Using the vector inner product is possible to calculate the minor angle γ between them. Where $AV1 = \text{agentVector1}$ and $AV2 = \text{agentVector1}$

$$\gamma = \arccos \left(\frac{\langle AV1 | AV2 \rangle}{\sqrt{\langle AV1 | AV1 \rangle} \sqrt{\langle AV2 | AV2 \rangle}} \right)$$

As geometrically expects, agents with close destination points will have a small angle, so negotiation must be done only with agents with small angle values. Is clearly that the rigid transformation with the inner product give us enough information but using the cross product could give us information about if the orientation start-target are the same.

5 A Vectorial Algorithm.

The whole algorithm is explained in Table 1.

6 Conclusions and future work

This paper presents an extended algorithm for limiting which agents within the broadcast radius could be better travel partners than classic flock traffic negotiation algorithm that calculate for each agent within the radius the complete route (meeting and splitting points, bone diagram and calculate the path using the social bonus).

With our algorithm the complexity is reduced to calculate just the meeting point, move and make cross product and inner product between each pair of agents.

Using vectorial is just the beginning of a series of improvements to the FTN approach for solving traffic congestions in cities. Future work will be focused on algorithms from computer graphics and the use of quaternions.

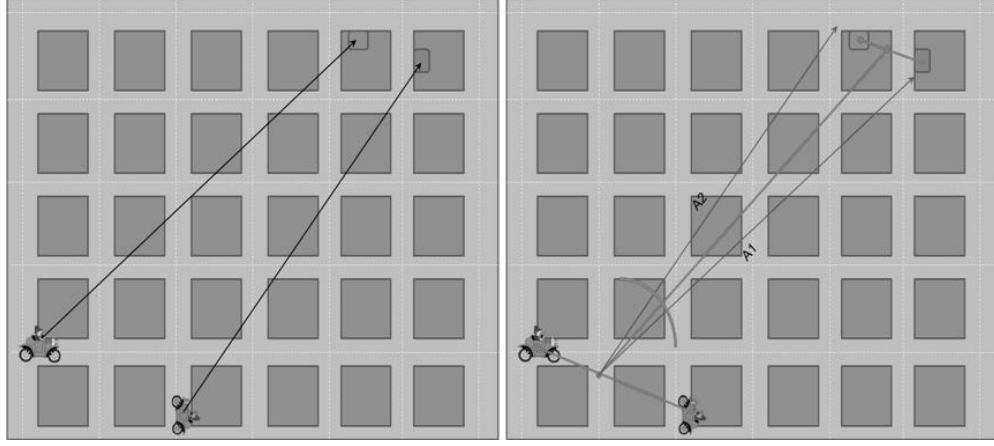


Figure 4. The classical problem transformed in vectors and the translation of vectors to the origin at the meeting point.

Table 1. Algorithm proposed

Input: Agent ID, Start and Target points for each Agent, radius.
Output: Partners subset

for each agent in radius **do**
 Calculate the meeting point using centroidMethod.
 Create the vectors for the agent and the possible partner
 Move vectors to the meeting point

If $\text{proyk}(\text{cross product}) = \lambda k$ with $\lambda > 0$
 delete agent for the agent subset
else
 Calculate the minor angle γ between them as:

$$\gamma = \arccos \left(\frac{\langle AV1 | AV2 \rangle}{\text{sqrt} \langle AV1 | AV1 \rangle \text{sqrt} \langle AV2 | AV2 \rangle} \right)$$

If $\gamma > \pi/4$ **do**
 delete agent for the agent subset
else
 Sort agents from minor γ to the greatest.
endif
endif
endfor

Start negotiation process as in [4];

References

- [1] Astengo-Noguez C. y Brena R. Flock Traffic Navigation Based on Negotiation. Palmerston North, New Zealand : s.n., 2006. Proceedings of 3rd International Conference on Autonomous Robots and Agents (ICARA 2006). pgs. 381-384.
- [2] Astengo-Noguez C. and Sanchez-Ante G. Collective Methods on Flock Traffic Navigation Based on Negotiation, Aguascalientes : Springer, 2007. 6th Mexican International Conference on Artificial Intelligence (MICA 2007).
- [3] Dresner K. and Stone P. Multiagent Traffic Management: An Improved Intersection Control Mechanism. Utrecht, The Netherlands : s.n., July 2005. The Fourth International Joint Conference on Autonomous Agents and Multiagent Systems (AAMAS 05). pgs. 471-477. <http://www.cs.utexas.edu/users/kdresner/pubs/files/2005aamas-dresner.pdf>.
- [4] Dresner K. and Stone P. Multiagent Traffic Management: A Reservation-Based Intersection Control Mechanism. New York, New York : s.n., July 2004. The Third International Joint Conference On Autonomous Agents and Multiagent Systems (AAMAS 04). pgs. 530-537. <http://citeseer.ist.psu.edu/630629.html>.
- [5] Schrank D. and Lomax T. [ed.] 2005 Urban Mobility Report. The Texas A&M University System.

s.l. : Texas Transportation Institute, May 2005.
http://tti.tamu.edu/documents/ums/mobility_report_2005_wappx.pdf.

- [6] Schrank D. and Lomax T. [ed.] 2007 Urban Mobility Report. The Texas A&M University System.
s.l. : Texas Transportation Institute, September 2007.
<http://mobility.tamu.edu/ums>
- [7] Wooldridge, Michael. Introduction to Multiagent Systems. s.l. : John Wiley and Sons, 2002.

An Improvement to Flock Traffic Navigation Algorithm Using Linear Regression Techniques

Lorena Martinez-Elizalde
ITESM Campus Monterrey
Av. Eugenio Garza Sada 2501 Sur
64849, N.L., Mexico
mlorena@itesm.mx

Carlos Astengo-Noguez
ITESM Campus Monterrey
Av. Eugenio Garza Sada 2501 Sur
64849, N.L., Mexico
castengo@itesm.mx

Abstract

Flock Traffic Navigation (FTN) is a recently approach to solve vehicular traffic congestions in cities using multiagent technology. These agents are instructed to cooperate with others in order to form flocks, which gain a speed bonus, therefore improving individual travel time. Identifying partners is a critical issue, since it is a time consuming process. In this paper we present an improvement to the negotiation algorithm using linear regression techniques in order to decide which subset of agents could be the best possible partners. Finally, we also present early experimental data that support these ideas.

1 Introduction

Traffic congestion is one of the leading causes of lost productivity and decreased standard of living in urban settings [1] [2].

According to the 2007 Mobility Report [3], The average weekday peak-period trip takes almost 40 percent longer than the same trip in the middle of the day, compared to 13 percent longer in 1982 [7]. Reducing total congestion saves time and fuel, and leads to decreased vehicle emissions [2]. The Texas Transportation Institute estimates that in 75 of the largest country-regionplace U.S. cities in 2001, \$69.5 billion dollars are wasted in time and fuel costs [7].

In 2006 Critical Issues in Transportation [4] point out that America's economy and quality of life depend on a transportation system that functions well. Transportation connects people to jobs, family, medical care, entertainment, education, and the goods needed for everyday life are possible because of transportation. The revolution in information technology (IT) is expected to bring major societal and economic changes, but the impacts on transportation demand are uncertain. Perhaps most important, the con-

tinued expansion of trade, stimulated by the IT revolution, will increase the stresses on a freight system already facing severe congestion.

Current methods for enabling traffic to flow through intersections include building overpasses and installing traffic lights. However, the former is only worth the cost at the most congested intersections, and the latter can be quite inefficient, often requiring cars to remain stopped even when no cars are present on the intersecting road [1].

2 Flock Traffic Navigation

Flock Traffic Navigation (FTN) based on negotiation is a new approach for solving traffic congestion problems in big cities [5]. In FTN, vehicles can navigate automatically in groups called flocks allowing the coordination of intersections at the flock level, instead of at the individual vehicle level, making it simpler and far safer. To handle flock formation, coordination mechanisms are issued from multi-agent systems.

The mechanism to negotiate [6] starts with an agent who wants to reach its destination. The agent knows an a priori estimation of the travel time that takes to reach its goal from its actual position if he travels alone. In order to win a speed bonus (social bonus) he must agree with other agents (neighbors) to travel together at least for a while. The point in which the two agents agree to get together is called the meeting point and the point where the two agents will separate is called the splitting point. Together they form the called "bone" structure diagram (Fig. 1).

Individual reasoning play the main role of this approach. Each agent must compare its a priori travel time estimation versus the new travel time estimation based on the bone-diagram and the social bonus and then make a rational decision.

Decision will be taken according if they are in Nash equilibrium (for two of them there is no incentive to choose

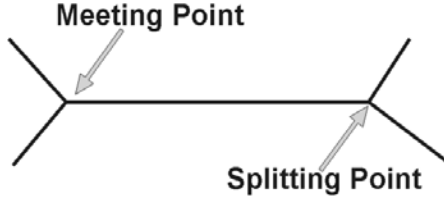


Figure 1. Bone Structure Diagram, Two agents agree to meet in their geometrical intersection according to its metric. Before the meeting point and after the meeting point agents will travel at their maximum individual speed. Between the meeting and splitting points agents can add a social velocity bonus.

other neighbor agent than the agreed one) or if they are in a Pareto Set [7].

If both agents are in Nash equilibrium, they can travel together as partners and can be benefited with the social bonus. In this moment a new virtual agent is created in order to negotiate with future candidates [9]. Agents in a Pareto Set can be added to this “bone” diagram if its addition benefits him without making any of the original partners worse off.

Simulations indicate that flock navigation of autonomous vehicles could substantially save time to users and let traffic flow faster [6].

With this approach the new agents in the Pareto set who want to be added into the bone-diagram have to share the same original splitting point calculated with the first agents who were in Nash equilibrium.

Because one major problem is decide which neighbors are the best to be included in the flock we decide to:

1. Improve the algorithm proposed in [6] by taking advantage of statistical methods.
2. Search for a simple measure for how good could be the negotiation.

3 Linear Regression

Our first goal is to find, primarily a linear, mathematical relationship $y=f(x)$ that pass very close to our agents’ starting and ending points. Geometrically, this means to find the the closest line, in some sense, that pass through the data points.

Let W be a subspace of $\mathbb{R}^n$. There is a unique $n \times r$ matrix P such that, for each column vector b in $\mathbb{R}^n$, the vector Pb is a projection of b on W . This projection matrix P can be

found by selecting any basis $\{a_1, a_2, \dots, a_k\}$ for W and computing $P = A(A^T A)^{-1} A^T$, where A is the $n \times k$ matrix having column vectors $a_1, a_2, \dots, a_k$ [8].

Regression analysis is well known technique used for modeling and analysing of data, using several variables to build a mathematical model that can explain the variation of a dependent variable. Regression is often used to build a linear equation that can predict the behavior of a dependent variable based on n independent variables. We attempt to use the geometrical interpretation of this linear model, which is a straight line that minimizes the distance among points that represent the agent’s starting and ending points.

The classical linear model is of the form:

$$\beta = (X^T X)^{-1} X^T Y$$

That leads in the space R^2

$$y = \beta_0 + \beta_1 x + \epsilon$$

Where x is the independent variable and ϵ is the error term, considered a random variable with $E(\epsilon)=0$ and $cov(y)=\sigma^2 I$. Thus it follows the Gauss-Markov squema. Using the least squares approach to solve equation and find β_0 and β_1 we find the following estimators. The complete process to calculate the estimators can be found on [9] [10] and [11]

For the slope term:

$$\hat{\beta}_1 = \frac{\sum x_i y_i - (\sum x_i)(\sum y_i)/n}{\sum x_i^2 - (\sum x_i)^2/n} = b_1$$

For the intercept term:

$$\hat{\beta}_0 = \bar{y} - \hat{\beta}_1 \bar{x} = b_0$$

Where $\bar{y}$ and $\bar{x}$ represent the mean value of each variable. Once the values are calculated, the questions of how good the model is and how significant it is must be answered. Because we are using a Regression approach the determination coefficient R^2 can be used to address the first question. This parameter refers to the percentage of variance accounted by the model and it can be calculated by:

$$R^2 = 1 - \frac{\sum_i (y_i - \hat{y}_i)^2}{\sum_i (y_i - \bar{y})^2}$$

R^2 gives us values between $[0,1]$; so it can be interpreted as the closer R^2 is to 1 (or 100%) the better the model fits the points given.

The model is significant if it explains a reasonably large amount of the variance. To test this hypothesis, a ratio among the variance of the random variable θ and the residual terms of the model is used. This ratio is distributed with the Fisher F-statistic shape.

$$F = \frac{\sum_i (\hat{y}_i - \bar{y})^2 / p}{\sum_i (y_i - \hat{y}_i)^2 / (n - p - 1)}$$

Where p represents the degrees of freedom. The F statistic helps us to reject the null hypothesis in favor to the alternative; that is, at least some of the regression model coefficients are not equal to zero. [11] The larger the F value, the smallest the p-value used to reject the null hypothesis. In summary the model is better when it has large values in the parameter R^2 and the F statistic.

4 An improvement to the FTN Algorithm

In the classical FTN algorithm a special matrix for negotiation must be filled with the calculation of the meeting and splitting points. Each matrix entrance represents the time to reach the meeting point from the starting point at normal speed plus the travel-time using the social speed bonus plus the travel time from the splitting point to the target or goal, again at normal speed. All these calculations must be done within the neighbors.

Once an agent receive other agents' information about their actual position and destination, they start a negotiation phase that is time consuming, this is because they have to decide via communication if they are or not in Nash equilibrium. In our approach, it is clear that a straight line connects each agent's starting and ending points, now within the broadcast answer (first communication phase on the classical approach) the agent receives neighbors' destinations points, and with this information he can ask himself (he performs the calculations without communicating with other agents) how different could his regression line be if he accepts new partners. This could be done adding one agent at a time using the fastforward regression method.

Let see it with more details, since the shortest path between two certain points is a straight line, we use linear regression to find which final destination points of our neighbors are best fitted to the line generated by the agent own starting and ending points.

Suppose an agent started a call for partners in a chosen radius; assuming the agents share their starting and ending points, this agent starts the regression process using as data, its own travel information, and the destination of the agents who answered the call. Then the agent proceeds to calculate the leverage points (Remember that a leverage point is often an outlier, that means that his presence move the entire line like if he has a special weight), in other words, the agent calculates the regression line, adding systematically one agent's target point at a time, calculating its R^2 and F . In figure 2, we can see an example of a regression line calculated and all its parameters.

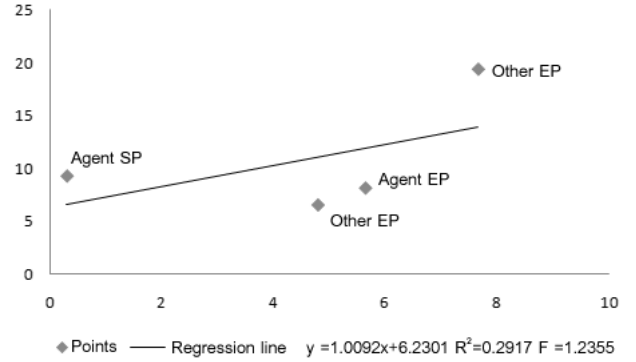


Figure 2. Regression line of one agent and another two agents which are potential neighbors to negotiate with. Because the R^2 is not very good, therefore there should be a better fit removing another point.

Once the calculations are done, the agent chooses the regression with the largest value of F, meaning has the least error, and has the best fit. The agent can then start negotiating with the agents that are near to it, and are going near its final destination. In figure 3, the parameters show improvement, the agent will choose to negotiate only with the agents that participated in this second regression. The algorithm is shown in Table 1.

Once the agents use this algorithm to reduce the number of negotiations among them, they will form new flocks and later, during the trip they could use algorithms for recalculate new splitting points like in [5].

5 Experiments

The algorithm explained in the previous section was programmed in Java in order to compare it against the results in [6].

Each experiment was replicated 100 times on city-sections of 5x5 blocks and 20x20 blocks. In order to test how well the algorithm fits according to the density (number of cars by block), eight subsets of agents were tested in each scenario. The experiments were run on a core-duo 1.5 Ghtz, 2 G of RAM.

Each time that the experiment was run the number of neighbors in which the classical FTN must calculate the entirely route between agents was recorded. Then our algorithm calculates via regression which neighbors are the most appropriate to negotiate and record the final number of agents in the subset.

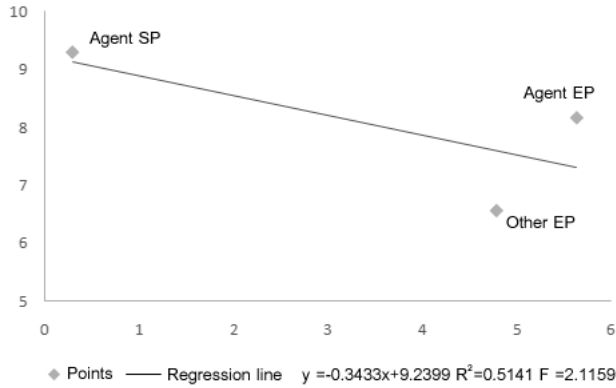


Figure 3. This regression line shows improvement on both parameters R^2 and F . The agent calculating the regression, will start negotiation only with the agents which points participated in this line.

Table 1. Algorithm proposed to calculate with which agents to negotiate

```

Input: N agents, a city section size, radius
Output: Subset of neighbor agents for negotiation
Foreach agent do
  Broadcast for neighbors within the radius
  Ask for the target points
Foreach neighbor do
  Calculate regression, and  $R^2$  and  $F$ 
  Delete neighbor according to fast-forward regression method
end
  Choose neighbors that participated in the regression with the largest value of  $F$ 
end

```

Table 2. Scenario 1, a city section of 5x5 blocks and subsets of number of vehicles in each simulation

5x5							
5	25	50	75	100	250	500	1000
4.71	8.99	11.23	12.76	19.98	41.13	53.66	122

Table 3. Scenario 2, a city section of 20x20 blocks and subsets of number of vehicles in each simulation.

20x20							
10	25	50	75	100	250	500	1000
3.35	5.2	7.21	9.5	12.14	12.77	28.42	64.49

6 Results

In both scenarios the improved algorithm works well, and it is clear that depending on the density of vehicles the algorithm performance is better in high density subsets.

In both scenarios the mode is three, meaning that in most of the cases the agent only has to calculate the entirely shared route with only other few agents. The minimum and maximum number of agents to negotiate with depends on the subset.

7 Conclusions and future work

Flock Traffic Navigation (FTN) is a recently approach to solve vehicular traffic congestions in cities using multiagent technology. Previous work use algorithms that take a big amount of computational time because they included all the

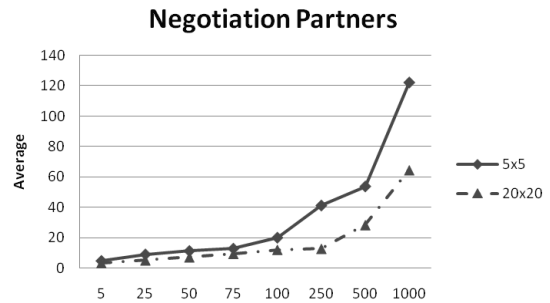


Figure 4. The graph shows the average agents to negotiate with in each scenario.

neighbors for negotiation. In our approach we have developed a new improved algorithm that takes statistical tools in order to reduce the number of agents to negotiate.

Our simulations point out that our approach could substantially save time, due to the reduced number of agents with which to start a negotiation process. The regression analysis gives us not only a subset of best fitting neighbors but also, a measure of fitness.

Our approach is based, like in [6] on a regular square city, future work would include experimental research to prove that we can discriminate neighbors using only the determination coefficient instead of Fishers F-test. We would also like to test our algorithm with different random distributions and apply alternative regression methods like LSD, Principal Components and Ridge.

References

- [1] Dresner, Kurt y Stone, Peter.: Multiagent Traffic Management: A Reservation-Based Intersection Control Mechanism. New York, New York : s.n., July 2004. The Third International Joint Conference On Autonomous Agents and Multiagent Systems (AAMAS 04). pgs. 530-537. <http://citeseer.ist.psu.edu/630629.html>.
- [2] Dresner, Kurt y Stone, Peter.: Multiagent Traffic Management: An Improved Intersection Control Mechanism. Utrecht, The Netherlands : s.n., July 2005. The Fourth International Joint Conference on Autonomous Agents and Multiagent Systems (AAMAS 05). pgs. 471-477. <http://www.cs.utexas.edu/users/kdresner/pubs/files/2005aamas-dresner.pdf>.
- [3] Schrank, David y Lomax, Tim. [ed.]: 2007 Urban Mobility Report. The Texas A&M University System. s.l. : Texas Transportation Institute, September de 2007. <http://mobility.tamu.edu/ums>.
- [4] Njord, John R. y Meyer, Michael D. s.l. : Critical Issues in Transportation. Transportation Research Board of the National Academies, 2006, TRB. <http://onlinepubs.trb.org/onlinepubs/general/CriticalIssues06.pdf>.
- [5] Astengo-Noguez, Carlos y Snchez-Ante, Gildardo. Collective Methods on Flock Traffic Navigation Based on Negotiation. Aguascalientes : Springer, 2007. 6th Mexican International Conference on Artificial Intelligence (MICAI 2007).
- [6] Astengo-Noguez, Carlos y Brena, Ramn. Flock Traffic Navigation Based on Negotiation. Palmerston North, New Zealand : s.n., 2006. Proceedings of 3rd International Conference on Autonomous Robots and Agents (ICARA 2006) . pgs. 381-384.
- [7] Wooldridge, Michael. Introduction to Multiagent Systems. s.l. John Wiley and Sons, 2002.
- [8] John, B. Fraleigh y Raymond, A. Beauregard. Linear Algebra. Addison Wesley, 1995. pg. 538. ISBN: 0-201-8399-7.
- [9] Devore, Jay L. Probability and Statistics For Engineering and the Sciences. Canada : Thomson, 2004. ISBN: 0-534-39933-9.
- [10] Rawlings, John O. Applied Regression Analysis: A Research Tool. United States of America : Wadsworth & Brooks, 1988. ISBN: 0-534-09246-2.
- [11] Lattin, Carroll and Green, Paul E. Analyzing Multivariate Data. Canada : Thomson, 2003. ISBN: 0-534-34974-9.

Implementation of Emotional Learning for Cognitive Tutoring Agents

Usef Faghihi, Pierre Poirier, Daniel Dubois, Mohamed Gaha and Roger Nkambou

Department of Computer Sciences, University of Quebec in Montreal

{faghihi.usef, poirier.pierre, dubois.daniel.2, gaha.mohamed, nkambou.roger}@courrier.uqam.ca

Abstract

We here describe some essential and fundamental learning mechanisms, both “conscious and unconscious”. The proposed architecture is based on functional “consciousness” mechanisms. In this paper, we suggest an Emotional Learning Mechanism (ELM) model, which can both learn new events and influence different types of “conscious” and “unconscious” learning. This will increase the agent's autonomy in unknown or changing environments, and thus improve its fitness. This architecture is inspired by mechanisms believed to exist in humans [6].

1. Introduction

Most cognitive scientists now recognize that emotions play an important role in cognition. Recent developments have supported the study of emotions and their influences in our behavior and different types of learning [18][20]. Accordingly, cognitive modelers have sought to include emotional mechanisms in the cognitive architecture of their agent. They have added an emotional mechanism [5] (Figure 1) into the CTS architecture. The latter was inspired from the latest *neurobiology* and *neuropsychology* theories of human brain function [13] [17]. We here explain how to best insert emotional learning in an artificial agent's cognitive architecture. We also detail how emotional learning gradually helps different types of learning, such as learning of regularities in this architecture. Human reactions and learning are under the influences of emotions and feelings [18]. Our work is based on the OCC (Ortony, Clore, & Collins, 1988) emotion model which extends from current models by defining learning in the emotional mechanism that helps learning and differentiating a variety of emotions. The modern view of emotions within cognitive neuroscience posits two distinct neurological routes from perception to emotions- the so-called “short route” and “long route” [18][17]. These two routes present distinct learning, reaction times and phenomenological profiles, the short route being fast and unconscious and the long route slower and

conscious. Because the two routes support distinct types of learning and reaction times, we believe that both must be included in any cognitive architecture that purports to include emotions into its cognitive cycle. In order to achieve human-like behavior in machines, or simply to better understand human adaptability, we aim to design human-inspired reactions and learning mechanisms.

2. Two routes towards action selection and learning

The amygdala, an important structure in the limbic system, associated directly with motivation and emotional reactions, as well as with reactions of fear and joy, and emotion-loaded memories [13]. Scientifics in neurobiology suggest that there are two routes from perception to the amygdala [13]. In the first route, which is supported by ‘*peripheralists*,’ James [12] suggests that the perception of emotional responses generates conscious emotional feelings. The stream of information coming from the sensory thalamus passes directly through the amygdala (*ESR's path, red-dotted rectangles in Figure 1 at end of text*). If the amygdala recognizes the information, then it reacts directly but implicitly by sending a signal to the motor system for instant reaction [18] or to prepare the body for faster reactions. Peripheral autonomic responses are resubmitted to the *insular region*, which produces an interaction with fundamental ‘cognitive’ processing. According to the second route (*ELR's path, red-dotted rectangles in Figure 1 at end of text*), which Centralists’ endorse, [2], emotional feelings are involved with consciousness and arise from a cognitive ‘*appraisal*’ of emotional stimuli. In this route various cortical areas (primary sensory cortex, unimodal associative cortex, polymodal associative cortex) perform an interpretation of the received information. This interpreted information is then sent to the *hippocampus*, apparently for memory retrieval and temporary storage. All this processing serves to give meaning to the information (categorization by the cortex) and link it to other events (the hippocampus’ explicit episodic memory), before it goes back to the amygdala for emotional evaluation and response.

These two aforementioned routes clearly demonstrate that all information coming from the external world passes through the amygdala before any decision or reaction can occur. As a matter of fact, the amygdala plays two distinct roles in human beings (Rolls, E.T., 2000): (1) It can learn and react directly to some particular or important events independently of the cortex [18], and (2) at the same time it can influence judgment, decision-making as well as many types of learning, such as perceptual, episodic, and procedural. Learning occurs faster when the individual is emotionally stimulated [18]. According to Phelps [16], emotional learning (especially the learning of fears) can happen in both ‘*observational*’ and ‘*instructional*’ manners in humans).

Many computer scientists implemented an Emotional Mechanism in their agent, inspired by the ‘*Centralists*’ approach. In our model, emotion is considered as “*valenced reactions to the external or internal stimuli*”, as defined by OCC model. Gratch and Marsella(2004), Franklin in [7] as well as Velásquez (Velásquez, 1996), proposed their emotional architectures. However, their implementation does not support learning by the mean of *peripheral-central model* which exists in human [12][3]. Secondly, in OCC model emotion *intensity* did not discuss in detail. In fact, our model supports both the *peripheral-central and the Centralists* model which produces emotional learning. The latter influences other types of learning as well as agent behavior. In addition, it seems that in Gratch and Marsella(2004) model emotion rise and fall very fast. We propose cognitive cycles (specific to CTS) and mood[9] to emulate emotions. This would facilitate emotional learning as well as learning of regularities in our agent.

In the present paper, we briefly describe *Centralist and Peripheralist* models. We then explain how CTS can learn emotionally and how this learning can help other types of learning.

3. CTS’s emotional architecture

Our generic agent is an extension of the generic cognitive architecture underlying it, and relies on functional “*consciousness*” mechanisms for much of its operations. Its functional architecture is inspired from LIDA’s [6], with some differences and some extensions. CTS’ [4], current goal is to help astronauts learn to manipulate Canadarm2, the robotic telemanipulator attached to the International Space Station. In doing so, the agent considers a multitude of elements: the current simulator state (position and

configuration of Canadarm2), potential dangers such as an imminent collision, the quality of the learner’s actions, and so on. In CTS, modules communicate with one another by contributing information to Working Memory (WM) through information *codelets*¹. These travel back to modules and other unconscious resources through loops of “*conscious publications*” that broadcast only the most important, urgent, or relevant information (as suggested by Baars,2003). CTS’ actual architecture distinguishes the two routes of emotions. However, emotional learning and how emotional mechanism can help other types of learning was not implemented in the original architecture. In doing so, first, we propose an emotional mechanism in CTS which influences different types of learning in the “long route” (ELR in Figure 1), (such as perceptual, episodic, and procedural), and is an addition to the processing currently present in CTS, somewhat differs from LIDA’s. Second, a new emotional learning to both agents, that is a shorter route that connects the “amygdala” directly to the “basal ganglia” (CTS’ action selection mechanism, its Behavior Network which inspired by [19][14]). The second learning situated in the CTS, “*pseudo-amygdala*” mechanism is a fast process of learning-CTS will never forget these types of information. The short route primes some actions in the Behavior Network(BN), bringing the capacity for quicker learning and action firing, even forcing reflex reaction in cases of strong emotions. In the long route, the amygdala may inject energy to the information located in the WM. Thereafter, it can influence all steps of the cognitive cycle such as decision making, learning and etc, in the agent. We will now describe in more details how amygdala influences CTS’ cognitive cycle.

3.1. Impact of Emotions in CTS’ Cognitive Cycle

The emotional long route involves the *consciousness* mechanism. Emotions influence this mechanism at every step in the cognitive cycle. We briefly describe each step in the cycle and then, in *italics*, explain the influence of emotions in that step.

Step 1: The first stage of the cognitive cycle is to perceive the environment; that is, to recognize and interpret the stimulus.

¹ Based on Hofstadter *et al.*’s idea, a *codelet* is a very simple agent, “a small piece of code that is specialized for some comparatively simple task”. Implementing Baars theory’s *simple processors*, codelets do much of the processing in the architecture.

All incoming information is evaluated by the emotional mechanism when low-level features recognized by the perceptual mechanism are relayed to the emotional codelets, which in turn feed activation to emotional nodes in the Behavior Network.

Step 2: The percept enters WM.

Step 3: Memories are probed and other unconscious resources contribute. All these resources react to the last few consciousness broadcasts (internal processing may take more than one single cognitive cycle).

What is brought back from episodic memory is evaluated by the emotional codelets (as part of emotional intervention ELR: 2 in Figure 1) and receives anew its emotional load.

Step 4: Coalitions assemble. In the reasoning phase, coalitions of information are formed or enriched. Attention codelets join specific coalitions and help them compete with other coalitions toward entering "consciousness".

Emotional codelets observe the WM's content, trying to detect and instill energy to codelets believed to require it and attach a corresponding emotional tag. As a result, emotions influence which information comes to consciousness, and modulate what will be explicitly memorized.

Step 5: The selected coalition is broadcasted. The Attention mechanism spots the most energetic coalition in WM and submits it to the "access consciousness," which broadcasts it to the whole system. With this broadcast, any subsystem (appropriate module or team of codelets) that recognizes the information may react to it.

Steps 6 and 7: Here unconscious behavioral resources (action selection) are recruited. Among the modules that react to broadcasts is the Behavior Network (BN). BN plans actions and, by an emergent selection process, decides upon the most appropriate act to adopt. The selected Behavior then sends away the behavior codelets linked to it.

In this step the emotion codelets stimulate emotional nodes in the BN, priming certain behavior streams, and thereby increasing the likeliness of their firing. This mostly mimics priming effects. The emotional tag (positive or negative valence) attached to the published coalition will influence how resources react. When the BN starts a deliberation, for instance to build a plan, the plan is emotionally evaluated as it is built, the emotions playing a role in the selection of the steps. If the looping concerns the evaluation of a hypothesis, it gives it an emotional evaluation, perhaps from learned lessons from past experiences.

Step 8: Action execution. Motor codelets stimulate the appropriate muscles or internal processes.

Emotions influence the execution, for instance in the speed and the amplitude of the movements.

4. How Can an Emotional Mechanism Learn in CTS?

Stimuli come into WM from the virtual world while CTS' pseudo-amygdala (its Emotional Mechanism or EM) works to detect events of emotional importance (for example, collision risks). On the other hand, this learning happens by the creation of the new nodes (*Observational learning*) as well as the modification (reinforcing or weakening) of the base-level activation and the intensity of reaction for each node in the emotional mechanism. EM learns all broadcasted information in the system. This learning will influence all decision making as well as any type of learning in CTS. This emotionally learned information will never be forgotten by the system [18]. In the next paragraphs, we explain how EM learns in CTS. Each new node in EM includes a context, an action, a result, a cause, and reaction intensity. The strengthening of the learning in each node happens very fast (especially in the case of fears) by strengthening the node's activation according to a sigmoid function. However, if for the information coming to WM, emotional mechanism receives no response, and it was considered as very important by perceptual nodes (Step4 of cognitive cycle), emotional mechanism will create a new, empty node with a context which explains ongoing events. To fill out the action part of the node, emotional mechanism will wait for the consciously-mediated selection of a behavior, and the ensuing broadcasting about the event with external confirmation (*Centralists*) after the execution of the action by CTS. If the selected action from BN received a strong (positive or negative) reinforcement from the environment, the emotional mechanism might learn the broadcasted information instantaneously that is in less than a second (CTS, processes information through cognitive cycles, which happen five times per second [6]). At this point, emotional mechanism has associated the context of the new node with the action selected and executed by CTS. Information carried out to consciousness exactly after the action is applied as the result of the created nodes. In our model, for the codelets situated in the EM, each emotional reaction is associated with two thresholds, λ, ω (Eq. 1). λ controls the emotion activation which means that if intensity goes beyond this threshold, the corresponding codelet in EM will release its output into the system. ω is the previous level of saturation for the codelet. The emotional codelet intensity facing to the stimuli at the time t is calculated by:

$$EIS = \frac{1}{1 + e^{(-EIS_{t-1} * \beta \lambda \omega)_{t.C}}}$$

(Eq. 1)

- EIS_{t-1} is the value of intensity for emotion at the previous step
- λ threshold for emotional activation release
- ω previous saturation
- β mood, a constant [0,1] for learning purposes
- C the number of cognitive cycles
- t actual time

We recall again that this fast emotional learning thereafter could be translated to a direct reaction (before information be broadcasted) due to the fears or a very high level of emotional state of the agent. This prepares CTS to learn how to react faster (with emotional reinforcement) for the next similar or identical events. However, if CTS learned to react very strongly to face the same or similar situation but it turns out that the agent might not react with the same energies, after primary reaction emotional codelets wait for environmental reaction(ER). If corresponding nodes which made reaction need modification for the next reactions (*peripheral-central*), then emotional nodes activations will be modified according to the sigmoid function. In this way, the emotional mechanism creates one emotional codelet (named A_i , as in Amygdala) for each very important stimulus S_i calling for an emotional response, with a connection weight W_i between them. The output of each emotional codelet is primarily obtained by the following equation:

$$A_i = S_i * W_i * EIS \quad (\text{Eq. 2})$$

Usually, CTS recognizes a situation instantaneously and will react in an appropriate time frame. However, sometimes, CTS may need more time to deal with the situation. Maybe it has no behavior ready to offer a reaction; maybe it entered a deliberation to establish a probable cause, or to decide what to do. But as the number of cognitive cycles (C) increases without resolution, the emotional salience of the stimulus increases (think of *fear*). Emotional codelets increase their output until they receive a signal from expectation codelets telling them whether they reacted appropriately, or until they set-off a *reflex* action. However, an emotional codelet may connect or react to some different perceptual nodes ($\sum$ in Eq.3) then we might calculate the output as a sum of all inputs to a neuron according to Eq.2. An emotional codelet energy is thus:

$$E_{A_i} = \sum (S_i * W_i * EIS) * C \quad (\text{Eq. 3})$$

Now, coming to the cases in which corresponding emotional nodes for particular events are already created in EM. If emotional mechanism learned some

particular events as of the highest importance, and for the next similar event it causes a directly (and intense) reaction, it may turn out that, following the execution of the action, CTS determines that such events are not important according to the received reinforcement when CTS observes a collision-risk situation made by astronaut in the micro-world and react directly and too intensively. After, some millisecond it understand that the reaction was wrong. If situation repeated many times, then the salience for reaction will be diminished. In this situation, the emotional mechanism might re-adjust W_i to diminish the importance of the stimulus toward a response. If it happens many times, the emotional mechanism will finish by classifying it as neutral information, giving it a neutral valence during arm manipulation by this astronaut. The opposite scenario may happen when information enters WM and is considered *normal* (neutral) by the emotional mechanism. However, after a conscious broadcasting follows the take of action, CTS receives strong reinforcement feedback. At this point, the system again may re-adjust W_i for corresponding nodes. Learning (W_i adjustment) could happen by calculating the difference between reinforcer (R) and the activation of the A_i codelet.

$$\Delta W_i = \beta * S_i * [R - \sum A_i] * C \quad (\text{Morén, J., 2002}) (\text{Eq. 4})$$

The β parameter is used as a standard learning rate parameter, settable between 0 (no learning) and 1. However, the emotion present in CTS will decay by losing a fixed portion of energy if the concept of actual emotion receives little attention in the following cognitive cycles. This “*peripheral-central*” model of emotional learning implemented in CTS is what Gratch & Marsella and all other models failed to propose. We observed that after implementing emotional learning, CTS is capable of reacting properly to the four percepts mentioned above such as Collision, Collision-Risk events and others. CTS has shown that it learns and subsequently reacts more rapidly to the user’s mistakes than before.

5. How an Emotional Mechanism Can Help Other Types of Learning in CTS?

Implicit Procedural learning and the learning of environmental regularities are implemented in CTS [5]. CTS’ conscious learning happens after information is broadcasted by the Access consciousness, whereas unconscious learning happens both in the perception phase and during the action selection process. The learning of regularities implemented in CTS rests on a bottom-up theory for categorizing inspired by the Hebbian learning theory, and Jackson’s *Pandemonium*

theory [11]. When virtual world information is sent to CTS, it eventually reaches WM. There, the network of information codelets just created by the Perception meets codelets and coalitions already present in WM, with which they form or reinforce links (pursue *ELR dotted rectangles* in Figure 1). Learning in WM (which remains *implicit* until a collection of codelets becomes known as a *coalition* and is broadcasted and memorized by declarative memories) essentially comes from the reinforcement of the links between codelets according to the time that they spend together in WM. More precisely the Learning Mechanism (LM) distinguishes which codelets already have connections with others, creates new links or reinforces the existing ones between all codelets in WM. However, in our new implementation, the LM not only detects and learns new coalitions but tries to find coalitions that were selected by the EM and received supplementary energies (Figure 1, *red plain arrows*). In fact, EM tries to detect and instill a portion of energy (positive or negative which is described as E_{A_i} in Eq.5) to a

particular coalition in the WM. In our model, emotion does not suddenly appear and disappear. In fact, Mood (Figure 1) helps the duration of present emotional states. The energy is instilled by Mood for the following cognitive cycles if the same stimuli come to the WM. In fact, when the entered information is new, depending on the agent's actual Mood (Figure 1), it tries to communicate with Emotion. This produces a portion of positive or negative energy that is instilled into the learning mechanism. It remains constant during the learning phase. Depending on the received information and emotional primary evolution of the situation (such as collision=high-treat or, collision-risk=medium-fear, camera-adjustment=low treat), EM produces a valence reaction. Then, depending on CTS, the actual mood and the received energies from the emotional mechanism, CTS can learn faster (40 cognitive cycles) or may learn normally (80 cognitive cycles) (Figure 2). These emotional interventions which allow concepts to be selected faster by the Attention mechanism for broadcasting by the Access consciousness (Step5 of cognitive cycle) also help LM to learn faster. When the information is broadcasted by consciousness, *explicit learning* in various forms and locations may take place. The links' strength between nodes is simulated along a saturating, sigmoid function. As an experiment in CTS, we tried a camera adjustment scenario in the micro-world. The task is the following: before any movement, the astronaut must perform a camera view adjustment to prevent any further collision in the micro-world. It must be noted that this situation is of low-threat. In Figure 2, the blue curve shows the previously learned mechanism

implemented in CTS. It illustrates how the strength of a link grows if a pair of codelets is brought back into WM as soon as it leaves it- it is never exposed to "memory loss". The link between codelets is constantly reinforced while they spend time together in WM receiving constant energy from EM. This learning requires 80 cognitive cycles. After implementing the CTS' emotional learning mechanism, we observed in CTS that LM learned the information influenced by EM (the red curve) faster than it did normal concepts. The learning process was completed in 50 cycles rather than 80 cycles.

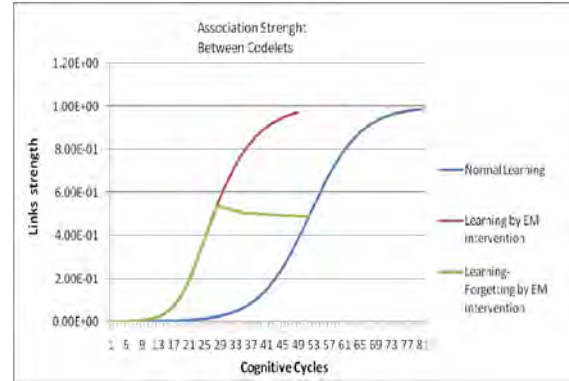


Figure 2 Learning and forgetting between codelets in CTS' WM with and without EM intervention.

However, as illustrated in Figure 2, if codelets that have links together are not called back together into WM soon enough, they will have their links weakened and eventually be forgotten by this natural phenomenon of attenuation. "Memory loss", our mechanism for forgetting, uses the same kind of sigmoid function involved in learning but reversed. In this new learning mechanism, concepts which received emotional tags will not be forgotten easily by CTS. Forgetting the sigmoid curve is so softened. Then as we can see (green line in Figure 2) learning steps happen faster than normal scenarios and forgetting steps happen very slowly. Here is the equation that we use for computing links' strength.

$$Strength = \frac{1}{1 + e^{(-sx*EA_i + d).t.C}} \quad (Eq. 5)$$

Where:

- x: association strength between two codelets
- s: rate of increase of base-level activation (for the links between codelets)
- d: threshold value for conversion into a coalition
- C: the number of cognitive cycles since the creation of the link.
- t: mean time for two codelets passed together in WM.
- E_{A_i} : emotional energy

To obtain the validation of CTS' new emotional learning, we realized some experiments. One of which will be addressed in this paper. We considered that astronauts are inactive for a while in the micro-world which means that CTS might learn how to react in such a situation. In the first essay, a Behavior node (red line in Figure 3) reacts to the broadcasted information in the system (without EM intervention). At this point, a menu must be shown to the astronaut. The latter must then be asked if some helps is needed. In the second essay, (the blue line) codelets learned the same scenario with EM influences and then stimulates a Behavior in the BN. We compare the case of BN normal reaction with the case involving an emotional mechanism. The experiment shows that, with our new emotional mechanism, after broadcasting the problem in the system, CTS starts interacting with the astronaut after 4th cognitive cycles, whereas, without "emotions", CTS reacts to the same situation only much later, at the 13th cognitive cycle. In simple terms, emotions diminish the time required to generate a reaction by instilling energy to the BN. CTS' artificial emotions work as expected.

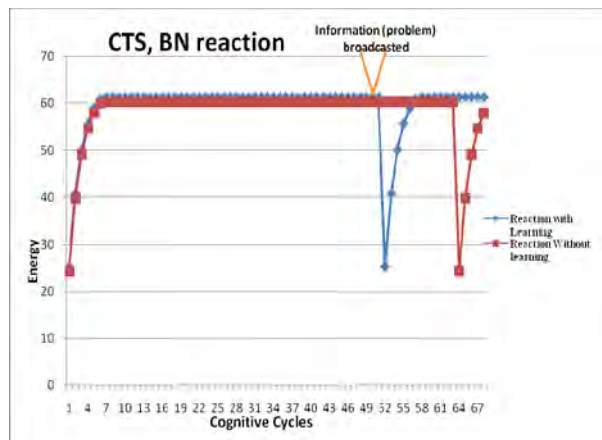


Figure 3 Comparison between normal and emotional-intervention reactions in CTS.

4. CONCLUSION

In this paper we described a new and generic way to implement "emotional learning" in conscious agents and how it could help other types of learning. Our architecture is more plausible neurologically, for it integrates a recent view of the amygdala's dual role with respect to emotions- to get the agent to learn and then react swiftly in emotionally-burdened situations, and supply an emotional assessment to all sorts of

stimuli in working memory which may be used for learning aims. This also helps faster learning for the emotionally-influenced information which enters working memory and is broadcasted later through CTS cognitive cycles. CTS' processing is now also closer to human phenomenology. In fact, from a cognitive-functional point of view, the agent is now better equipped to interact in a world where stimuli are not created equal- some being more pleasurable or more dangerous (physically or socially) than others. As our experiment illustrates, our emotional learning mechanism endowed CTS may when need be react more swiftly (reacting sooner in the cognitive cycle) than the previous CTS architectures. This, because the emotional learning mechanism is kept aware of the ongoing situation and can, in real-time, learn and at the same time contribute emotional valences to the description of the situation. This new emotional learning mechanism thus offers great flexibility in behavior adaptation.

ACKNOWLEDGMENT

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References

- [1]. Baars, B. J., and Franklin, S. (2003). How conscious experience and working memory interact. *Trends in Cognitive Science* 7:166-172.
- [2]. Cannon, W. (1927). The James-Lange theory of emotion: a critical examination and an alternative theory. *Am. J. Psychol.* 39, pp. 106-124.
- [3]. Critchley, H., & al, e. (2002). Fear conditioning in humans: the influence of awareness and autonomic arousal on functional neuroanatomy. *Neuron* 33, pp. 653-663.
- [4]. Dubois, D. R. (2006). "How 'Consciousness' Allows a Cognitive Tutoring Agent Make Good Diagnosis During Astronauts' Training". *Proceedings of the 8th International Conference on Intelligent Tutoring Systems (ITS'2006) LNCS No 4053*.
- [5]. Faghihi, U., & Dubois, D. N. (2007). Learning Mechanisms for a Tutoring Cognitive Agent. *Proceedings of the 7th IEEE International Conference on Advanced Learning Technologies (ICALT 2007)*, July 18-20, Niigata, Japan.

- [6]. Franklin, S., & Patterson, F. (2006). The LIDA architecture: Adding new modes of learning to an intelligent, autonomous, software agent. *Society for Design and Process Science* , Integrated Design and Process Technology, IDPT-2006.
- [7]. Franklin, S., & Ramamurthy, U. (2006). Motivations, Values and Emotions: Three Sides of the same Coin. *Proceedings of the Sixth International Workshop on Epigenetic Robotics, Paris, France, September 2006*,.
- [8]. Gratch, J., & Marsella, S. (2004). A Domain-independent Framework for Modeling Emotion. . *Journal of Cognitive Systems Research*, 5, 4 , 269-306.
- [9]. Hirt, E., McDonald, H., & Melton, R. (1996). *Processing goals and the affect-performance link: mood as main effect or mood as an input?* In: L.L. Martin and A. Tesser, Editors, *Striving and feeling: Interactions among goals, affect, and self-regulation*, Erlbaum, Mahwah, NJ (1996), pp. 303-328.
- [10]. Hofstadter, D. R., and Mitchell, M. (1994). The Copycat Project: A model of mental fluidity and analogy-making. In *Advances in connectionist and neural computation theory, Vol. 2: logical connections*, ed. K. J. Holyoak, and J. A. Barnden. Norwood N.J.: Ablex.
- [11]. Jackson, J. V. (1987). Idea for a Mind. *Siggart Newsletter*, 181:23-26.
- [12]. James, W. (1884). *What is an emotion?* Mind 9, pp. 188-205.
- [13]. Ledoux, J. E. (2000). Emotion circuits in the brain. . *Annual Review of Neuroscience*, 23, 155-184. .
- [14]. Maes, P. (1989). How to do the right thing. *Connection Science* 1:291-323.
- [15]. Ortony, A., Clore, G. L., & Collins, A. (1988). *The cognitive structure of emotions*. Cambridge, UK, Cambridge University Press, .
- [16]. Phelps, E. (2006). Emotion and Cognition: Insights from studies of the human amygdala. *Annual Review of Psychology*, 57, 27-53.
- [17]. Rolls, E. T. (2000). Neurophysiology and functions of the primate amygdala, and the neural basis of emotion. In *The Amygdala: a Functional Analysis* (ed. Aggleton, J. P.). . Oxford Univ. Press, Oxford, UK , p.447-478. .
- [18]. Squire, R. L., and Kandel, R. Eric. (1998). *Memory from mind to molecules*. page 167-171.
- [19]. Tyrrell, T. (1994). An evaluation of Maes' bottom-up mechanism for behavior selection. . *Adaptive Behavior* 2(4):307-348 .
- [20]. Velásquez, J. (1996). *Cathexis: A Computational model for the Generation of Emotions and Their Influence in the Behavior of Autonomous Agents*. S.M. Thesis Department of Electrical Engineering and Computer Science, Massachusetts Institute of Technology.
- [21]. Morén, J. (2002). Emotional and learning :A Computational Model of the Amygdala.

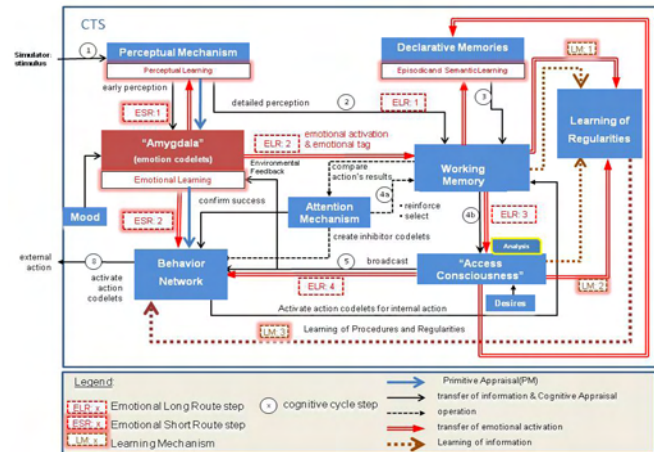


Figure1 Long, short routes, and emotional learning in CTS.

Bioinformatics and Medical Applications

Incremental Learning based Color Iris Recognition

Caitang Sun
Coll. of Computer Science
and Technology, Jilin
University, P.R. China
sunctjlu@126.com

Chunguang Zhou*
Coll. of Computer Science
and Technology, Jilin
University, P.R. China
cgzhou@jlu.edu.cn

Farid Melgani
Dept. of Information and
Communication Technologies,
Univ. of Trento, Trento, Italy
melgani@disi.unitn.it

Libiao Zhang
Coll. of Computer Science and
Technology, Jilin University,
P.R. China
lbzhang@yahoo.cn

De Natale Francesco
Dept. of Information and
Communication Technologies,
Univ. of Trento, Trento, Italy
denatale@ing.unitn.it

Xiaohua Liu
Coll. of Computer Science
and Technology, Jilin
University, P.R. China
xiaohua@jlu.edu.cn

Abstract

In this paper, an incremental learning based color iris recognition method is proposed. The inner and outer boundaries of the iris are localized employing Canny edge detection method and Hough transform algorithm in the red channel and the blue channel respectively. This method exploits some fusion strategies to combine the matching scores between the test image and the templates of each class. The strategies used in this paper are Min, Max, Sum, Product and Weighted Sum. The experiments were carried out on UBIRIS v1.0. The experiments show that, in some scenarios that acquisition conditions change leading to possible decrease of performance, because of its characteristic of learning in practice, the incremental learning method can handle the problem effectively.

1. Introduction

Biometrics has been applied to the security fields for decades, utilizing biometric features to authenticate or identify individuals. Examples of such features include face, fingerprint, retinal vasculature, signature, palm prints, iris, etc. Among them, iris is considered to provide the highest performance [1][2][3].

Most of the previously published papers use only gray level iris images acquired from infra red (IR) cameras. Proença et al. [4] constructed a color iris image database and did some experiments, but employed only the corresponding gray images. Even in [5] which can deal with color images, the authors did not exploit the multidimensionality of the images, but

simply compressed the images using the Adaptive Colour Reduction algorithm to images with a palette of 256 colors. The IR cameras can provide good quality iris images in ideal circumstances. But color cameras are more commonly used in practice. In general, the images acquired by the color cameras can convey more information. What about utilizing color cameras for iris recognition and taking advantage of the different channels? An increasing number of researchers have begun to address this problem. If feasible, the application areas of iris recognition system could be extended, and the system could be more robust. The problem in these kinds of color images is how to completely make use of the information from the different channels? Data fusion strategies can be a solution for it. Christopher et al. [6] analyzed iris recognition based on multispectral images using 120 images from 24 subjects, using sum rule to fuse the matching scores of the images collected by different spectral devices. They found that combining the blue channel with any other channel can provide a substantial improvement in performance, and the IR and Red channels performed very well for the brown irises. But the database they used is small scaled, and the images were captured under good conditions. In the previous work, we studied matching score level data fusion for color iris recognition on UBIRIS v1.0, a dataset purposely including some noise, but we didn't get any improvement compared to the red channel, because the green and blue channels are relatively poor. In this paper, we present an incremental learning method for color iris recognition.

For many biometrics recognition systems, the successfulness of the algorithms depends heavily on

the availability of adequate representative set of training examples. While the collection of such examples may be expensive or time consuming, sometimes new features, for example, images from different points of view, images from various conditions, may be discovered in subsequent episodes. In such scenarios, it is necessary to update the previously trained classifier to accommodate the new information in an incremental fashion. This kind of learning is called incremental learning (IL). It is a type of machine learning approach which can update a classifier automatically when an important experience is available. The most significant aspect of incremental learning is to have a continuously updated model from active learning. It can learn novel information from new data without forgetting the previously acquired knowledge. Learning new information could guarantee the good accuracy of a classifier. It could be applied in the situations where the training sets are updated in consequence and where it is crucial to timely update the models [7][8].

This paper is organized as follows. Section 2 describes color-based iris recognition system and introduce the commonly used score level fusion strategies. Section 3 depicts incremental learning based color iris recognition method. Section 4 reports the experimental results. Section 5 draws some conclusions.

2. Color Image based Iris Recognition

The multispectral sensors (e.g. RGB sensors) have the advantage that the images from them can convey richer information content and thus higher discriminative potential. Unlike images acquired by IR cameras, color images are of the multidimensional type. The problem in this case is how to combine the information coming from the different channels of the multispectral sensors. To deal with this problem, data fusion can represent an efficient solution.

Kittler et al. [9] developed a theoretical framework for fusing the information obtained from multiple classifiers using strategies like the sum, product, max, min, median rules and majority voting. Their experimental comparison of various classifier combination schemes on frontal face + face profile + voice and handwritten digit recognition demonstrated that the combination rule developed under the most restrictive assumptions-(the sum rule) outperforms other classifier combinations schemes. In [10], the author gave formulas for the classification error for the following fusion methods: average, minimum, maximum, median, majority vote and oracle under the assumption that the estimates are independent and

identically distributed (normal distribution $\sim N(p, \delta^2)$ or uniform distribution within $[p-b, p+b]$). Duin [11] argued the properties of the different fusion strategies when they were used to combine the posterior probability of different classifiers. The product rule is good if the classifiers are independent. For independent classifiers, the sum rule is equivalent to the product rule, and it can work well under the condition of a collection of similar classifiers with independent noise behavior. The max rule is used to select the classifier that is most confident of itself, but it will fail in the case of some classifiers being more overtrained than others. It appears to be hard to find examples that a global max/min rule is intuitively a good choice. Czyz et al. [12] pointed out the link between combining the posterior probabilities and combining directly the matching scores when all classifiers operate in the same measurement space in the case of two class problem. Ross and Jain [13] used score level fusion strategies to combine the matching scores of face, fingerprint, and hand-geometry modalities. Alkoot et al [14] evaluated the fusion strategies under the conditions of different kinds of noise and various numbers of classifiers. The fusion strategies demonstrated different performance ranking relationship under different conditions.

2.1. Iris Localization

In this paper, both the inner (pupil) and the outer boundaries are localized utilizing Canny edge detection and Hough transform algorithm [2]. We learn from experiments that, in the red channel, the discriminative property between the inner and outside of the pupil is more remarkable than in the green and blue channels. While the contrast between the iris and the sclera near their boundary are more obvious in the blue channel. Consequently, we first localize the pupil in the red channel, and then locate the outer boundary in the blue channel based on the position of the pupil.

When localizing the pupil, we first select ten candidates based on the Hough voting results. Then, the mean intensity and the smoothness of them are compared. The candidate with lowest mean intensity and smoothness is selected as the pupil.

2.2. Fusion Strategies

Suppose that $D(I, T_{ij})$ is the distance between the test image I and the i th template of the j th class (subject), $i=1,2,\dots,M$, where M is the number of different matchers; $j=1,2,\dots,N$, and N is the number of classes, and suppose $D(I, T_j)$ to be the final distance after fusion between I and the j th class. In the

experiments, we exploit data fusion at the matching score level. There are several fusion strategies commonly used at this level. Those considered in this work are:

Min fusion strategy: Choose the minimum score:

$$D(I, T_j) = \min_{i=1,2,\dots,M} \{D(I, T_{ij})\}, \quad j = 1, 2, \dots, N \quad (1)$$

Max fusion strategy: Choose the maximum score:

$$D(I, T_j) = \max_{i=1,2,\dots,M} \{D(I, T_{ij})\}, \quad j = 1, 2, \dots, N \quad (2)$$

Simple sum fusion strategy: (Sum the scores).

$$D(I, T_j) = \sum_{i=1}^M D(I, T_{ij}), \quad j = 1, 2, \dots, N \quad (3)$$

Product fusion strategy: (Product of the scores)

$$D(I, T_j) = \prod_{i=1}^M D(I, T_{ij}), \quad j = 1, 2, \dots, N \quad (4)$$

Weighted Sum fusion strategy: In this method, the user assigns a weight for each matcher, according to some empirical knowledge or some supervised estimation methods (e.g., least square method) which requires however the use of additional templates.

$$D(I, T_j) = \sum_{i=1}^M w_{ij} D(I, T_{ij}), \quad j = 1, 2, \dots, N, \quad \sum_{i=1}^M w_{ij} = 1 \quad (5)$$

For all strategies, the final matching decision in this paper is:

$$\hat{j} = \arg \min_{j=1,2,\dots,N} \{D(I, T_j)\} \quad (6)$$

That is, the image I is considered to belong to the $\hat{j}$ th class if the distance between I and the $\hat{j}$ th class is the minimum.

3. Incremental Learning based Iris Recognition

Most iris recognition systems use only one template for one class. We may however find in practice several scenarios where acquisition conditions change: (1) the image acquisition device is changed after a period of exploitation; (2) additional devices are integrated in the system; (3) the images are shared with other systems making use of different acquisition devices; (4) it is time consuming to acquire templates at the same time; (5) the pose of the user changes. In any of these scenarios, the performance of the system may degrade if using single template. The changes in acquisition conditions make an update of the templates necessary. A possible solution to this problem is to employ multiple templates. Liu etc. [15] used multiple gallery/probe images in iris recognition system and utilized min rule to fuse the distances. Some improvements were obtained with respect to

conventional systems. But sometimes the users are not able to acquire the templates under all the possible conditions initially. Another solution is incremental learning, which can update the templates automatically. Applying incremental learning to iris recognition means to update the old template of a subject with new information. Consequently, data fusion strategies could also be useful to combine the different templates into a new one. In this paper, we add the new template to the template set, and then use the min, max, sum and product rules to combine the matching scores between them and the test image. The block scheme of an iris recognition system performing incremental learning is illustrated in Figure 1.

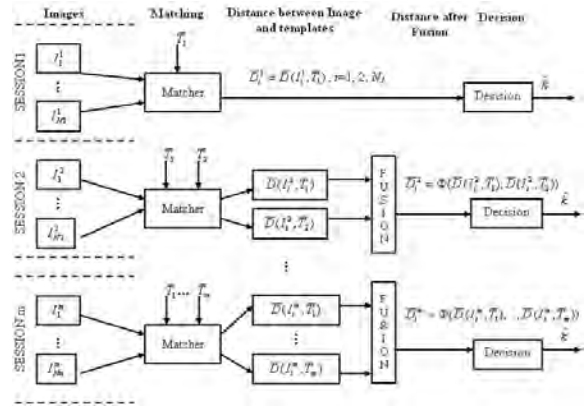


Figure 1. Block scheme of our proposed system based on incremental learning, where $\bar{T}_i$, $i=1,2,\dots,m$, is the template added to the database in session i , and $\bar{D}(I_i^j, \bar{T}_k)$ ($k=1, 2, \dots, N_m$, $k=1, 2, \dots, m$, $j=1, 2, \dots, m$) denotes the distance vector between the k th image and the k th template in session j .

In incremental learning, we define the term 'session' as a sequence of acquisitions and matching carried out on the basis of a given collection of templates. When a new template is added, the session changes. Therefore in session 1, there is only one template for each class, the distance between the test image and the template can be directly used to decide which class the image belongs to. In session 2, an extra template is added to each class. Consequently, in this session, every test image should be compared with two templates for each class, instead of just one. Then the matching scores are combined using certain fusion strategy to calculate the $\bar{D}_i^2 = \Phi(\bar{D}(I_i^1, \bar{T}_1), \bar{D}(I_i^2, \bar{T}_2))$ distance between the image and each class. Φ can be one of the fusion rules described in section 2.2, namely: Min, Max, Simple Sum and Product. As shown in Figure 1, whenever a new template is added, the fusion procedure needs to be

consequently updated.

4. Experimental Results

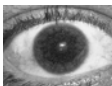
4.1. Dataset description

To evaluate the performance of our proposed methods, we carried them out on UBIRIS.v1.0 dataset [16]. Its main characteristic results from the fact that, opposed to the existing public and free databases (CASIA [17] and UPOL [18]), it purposely incorporates images with several noise factors. It is composed of 1876 color images collected from 241 persons during two distinct sessions using color camera. Table 1 shows some examples of the images in the dataset. The different channels of some of them are shown in Table 2.

Table 1. Examples of images in the UBIRIS dataset

No	Session 1	Session 2	No	Session 1	Session 2
1			61		
7			115		
28			119		
45			170		

Table 2. Different channels of the images

Subjects	Original	Red Channel	Green Channel	Blue Channel
1				
61				
119				
170				

In the first image capture session, the acquisition device was installed inside a dark room. The images are characterized by the minimization of noise factors, especially those relative to reflection, luminosity and contrast. In the second session, the capture location was changed in order to introduce natural luminosity factor. Images collected in this session pretend to simulate the images captured without or with minimal active collaboration from the users. Each subject has 5 to 6 images in each session. For the sake of homogeneity, we only use the first 5 images in our experiments. All images from both sessions are classified with respect to three parameters ('Focus', 'Reflections' and 'Visible Iris') in a three values scale ('Good', 'Average' and 'Bad'). The classification statistics are detailed in Table 3.

Table 3. Quality parameters of the dataset

Parameter	Good	Average	Bad
Focus	73.83%	17.53%	8.63%
Reflections	58.87%	36.78%	4.34%
Iris visibility	36.73%	47.83%	15.44%

From Table 3, we can see that 8.63% of the images are out of focus, and only 73.83% of them are well focused. In 58.87% of the images, the reflection in the irises does influence the performance of iris localization of recognition, but 4.34% of them are difficult to localize or recognize because of the reflection. Only in 36.73% of the images, the irises are not occluded by any other thing, such as eyelashes and eyelids, while 15.44% of them are mostly or completely occluded.

4.2. Results with Incremental Learning

The localization method is described in section 2.1. The accuracy in session 1 is 96.3%, and that in session 2 is about 84.6%. The feature extraction algorithm exploited in the experiments was that presented by Masek and Kovesi [19].

To perform the experiments on incremental learning, we select the 127 subjects that appear both in sessions1 and session2. We exploit the two capture sessions to simulate two successive sessions. In particular, in the first session, we select for each subject the first image of session1 as the template, and use the remaining four images as test images. Then, in the second session, the first image in session2 is added as a new template, and the remaining four are considered as test images. To calculate a global distance between a test image and the subject, we apply four fusion strategies: Min, Max,

Sum and Product, to combine the distances between the test image and the templates in each class. The obtained accuracies are shown in Table 4.

In the first column, the fusion strategies, Min, Max, Sum, Product and WS (Weighted Sum) mean fusing the matching scores of the different channels before applying incremental learning. Table 4 indicates that no fusion strategy, can guarantee amelioration compared to the best single channel, the red channel. The 'Weighted Sum' rule can improve the performance when working with incremental learning method, but it depends on the assignment of the weights.

Table 4. **Results of the incremental learning based iris recognition (%)**

Channel/ Fusion	Incremental Learning Method				Non- Incremental Learning
	Min	Max	Sum	Product	
Red	83.96	77.56	83.37	83.66	76.03
Green	81.00	70.57	81.30	81.50	68.33
Blue	66.73	51.28	66.83	66.73	47.07
Min	83.17	76.97	83.46	83.66	75.07
Max	74.02	57.58	73.62	73.72	53.28
Sum	82.28	75.79	82.97	83.07	71.30
Product	82.38	75.89	83.07	83.17	71.39
WS	84.06	77.66	83.66	83.76	75.94

The fusion strategies in the incremental learning method aim at fusing the matching scores of the test image and all the templates in each class. As can be seen, for each single or fused channel, all the fusion strategies applied to the incremental learning method can improve the accuracy compared to the non-incremental learning method. This is because it employs 2 templates in session 2 in this experiment. The Min rule seems to be the best. The Max rule is worse than the others. There is not much difference between the Sum and Product strategies.

5. Conclusions

In this paper, we present an incremental learning method for color image based iris recognition. Employing this method, all the fusion strategies can improve the performance compared to non-incremental learning method. The strategy 'Min' appears to be the best, while Sum and Product are not very different to it. Though the 'Max' rule is worst than the others, it can also provide a higher accuracy than the red channel based on non-incremental learning method. The experiments indicate that the learning ability can help to increase the accuracy of iris recognition.

In the experiments, we only update the template of one subject by appending the new added one to the template set, and combining the matching scores in authentication mode. In the future, we will try some other algorithms to merge multiple templates into one, and exploit some rules to update the template only when an experience obtains good performance.

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References

- [1] J.G.Daugman, "Probing the uniqueness and randomness of IrisCodes: Results from 200 billion iris pair comparisons", Proc. of the IEEE, 94(11), 2006, pp. 1927-1935.
- [2] R.P.Wildes, "Iris Recognition: An Emerging Biometrics Technology", Proc. of the IEEE, 85(9), 1997, pp. 1348-1363.
- [3] PJ Phillips, WT Scruggs, AJ O'Toole, PJ Flynn, KW Bowyer, CL Schott and M Sharpe, "FRVT 2006 and ICE 2006 Large-Scale Results", NISTIR 7408, 2007.
- [4] Proença. Hugo and Alexandre. Luís A, "UBIRIS: A noisy iris image database". International Conference on Image Analysis and Processing, 2005, pp. 970-977.
- [5] E.Krichen, M.A.Mellakh, etc., "Iris Identification Using Wavelet Packets", 17th International Conference on Pattern Recognition, 4, IEEE Computer Society, Washington, DC, 2004, pp. 335-338.
- [6] C.Boyce, A.Ross, etc, "Multispectral Iris Analysis: A Preliminary Study", Computer Vision and Pattern Recognition Workshop on Biometrics(CVPRW), New York, 2006, pp. 51-59.
- [7] Wei-Min Shen, "Efficient Incremental Induction of Decision Lists - Can Incremental Learning Outperform Non-Incremental Learning?", Technical Report, 1996.
- [8] R.Polikar, "An incremental learning algorithm with confidence estimation for Automated Identification of NDE Signals", IEEE transactions on ultrasonics, ferroelectrics, and frequency control, 51(8), 2004, pp. 990-1001.
- [9] J.Kittler, M.Hatef, R.P.W. Duin and J.Matas, "On Combining Classifiers", IEEE Transactions on Pattern Analysis and Machine Intelligence, 20(3), 1998, pp. 226-239.
- [10] L.I. Kuncheva, "A Theoretical Study on Six Classifier Fusion Strategies", IEEE Transactions on Pattern Analysis and Machine Intelligence, 24(2), 2002, pp. 281-286.
- [11] R.P.W.Duin, "The Combining Classifier: To Train Or Not To Train?" 16th International Conference on Pattern Recognition, II, Quebec, 2002, pp. 765-770.
- [12] J. Czyz, J. Kittler and L. Vandendorpe, "Multiple

- classifier combination for face-based identity verification", Pattern Recognition, 37(7), 2004, pp. 1459-1469.
- [13] A.Ross and A.Jain, "Information fusion in biometrics", Pattern Recognition Letters 24, 2003, pp. 2115-2125.
- [14] F. Alkoot and J. Kittler, "Experimental Evaluation of Expert Fusion Strategies", Pattern Recognition Letters, 20, 1999, pp. 1361-1369.
- [15] X. Liu, K.W. Bowyer and Patrick J. Flynn, "Experimental Evaluation of Iris Recognition", IEEE Computer Society Conference on Computer Vision and Pattern Recognition, 3, 2005, pp. 158-165.
- [16] Hugo Proença and Luís A. Alexandre, "Ubiris iris image database", <http://iris.di.ubi.pt>.
- [17] CASIA Iris Image Database, <http://www.sinobiometrics.com>.
- [18] M.Dobeš and L.Machala, Iris Database, <http://www.inf.upol.cz/iris/>.
- [19] L.Masek, P.Kovesi, "MATLAB Source Code for a Biometric Identification System Based on Iris Patterns", School of Computer Science and Software Engineering, University of Western Australia, 2003.

Design Of The Approximation Function of a Pedometer based on Artificial Neural Network for the Healthy Life Style Promotion in Diabetic Patients

A. Vega-Corona, M. Zárate-Banda, J.M. Barrón-Adame, R.A. Martínez-Celorio

Universidad de Guanajuato
Lascuráin de Retana No. 5 Centro
Guanajuato, Gto. México.
tono@salamanca.ugto.mx

D. Andina

Universidad Politécnica de Madrid
Av. Complutense No. 30. Ciudad Universitaria
Madrid (España).
d.andina@upm.es

Abstract

The present study describes the design of an Artificial Neural Network to synthesize the Approximation Function of a Pedometer for the Healthy Life Style Promotion. Experimentally, the approximation function is synthesized using three basic digital pedometers of low cost, these pedometers were calibrated with an advanced pedometer that calculates calories consumed and computes distance travelled with personal stride input. The synthesized approximation function by means of the designed neural network will allow to reply the calibration experiment for multiple patients with Diabetes Mellitus in Healthy Life Style promotion programs. Artificial Neural Networks have been developed for a wide variety of computational problems in cognition, pattern recognition, and decision making. The Healthy Life Style refer to adequate nutrient ingest, physical activity, time to rest, stress control, and a high self-esteem. The pedometer is a technological device that helps to control the physical activity in the diabetic patient. A brief description of the Artificial Neural Network designed to synthesize the Approximation Function, the obtained Artificial Neural Network structure and results in the Approximation Function synthesis for three patients are presented. The advantages and disadvantages of the method are discussed and our conclusions are presented.

keywords: Artificial Neural Networks, Pedometer, Approximation Function, Diabetes Mellitus, Healthy Life Style.

1 Introduction

In many research areas and in daily life there are present phenomena that have a multivariate order, it means, they can not be explain considering one variable. This can be observed in clinical cases and the *Healthy Life Style (HLS)* definition is not the exception. Every day, diabetic patients face the challenge of keeping a *HLS*, and on it depends their well-being. The treatment guidelines is oriented to provide insulin and hypoglycemic [3], it is based on calculating the corporal weight of the patient W_i , the calories ingested C_i and the physical activity G_i , producing a value determined by the glucose measurement in blood or urine, establishing the necessary insulin level I_i . We can express these parameters as a multi-variable function: $I_i = f(W_i, C_i, G_i)$. The multi-variability, the patient tolerance and response to the different kinds of insulin and hypoglycemic making the treatment complex. Due to some alteration in the habitual behavior of the patient, the established parameters are not always resolatory for personal treatment; for example, the calorific ingest in a social event, a day with big physical activity, or some situation in which the stress level increases the patient response. The established insulin program can be not the adequate for those days. In general, the health services organize the visit to a specialist which is a tiring process. First, the patient goes to the family doctor, who determines if the patient should or should not attend the specialist, while the time elapses and the patient could be unbalanced for a period of 2 or 3 weeks, if this becomes recurrent it predisposes to the early diabetes complications. It also exists other support service to the diabetic patients

treatment as the self-help groups where the health professionals have the responsibility of monitoring the group and their activities. Not only, to participate in early stages in these groups helping the patient to acquire and to maintain the habit for a *HLS*. Despite the therapeutic objective of maintaining the patient's independence and to modify the least possible their *HLS*.

For 10 years, the health professionals have promote the use of auxiliar instruments in the diabetic treatment, like glucometers, pedometers, etc. It is determined by the cost, complexity and handling, the use of the technology and its instruments have not had a satisfactory acceptance, however, we can say that the acceptance is good. During the practice of diabetic patient's care and programs to encourage the self-care like strategy for a *HLS*. The evaluation of devices or instruments (pedometers) show a great variety available in the market; some are more accurate than others, and others more complete but also complex and expensive. It allows to recover information and to verify the behavior of the patient's physical activity. Then, we have observed how the appropriate use of the pedometer contributes to provide appropriate information to determine the personalized treatment. Considering that in Mexico diabetic patients are increasing every day. The human resources for health attention are insufficient and the market will provide different Pedometer and Glucometer models. Health professionals consider the following questions: Is there a reliable way to integrate the obtained information from Pedometers and Glucometers to modify or to personalize the diabetic patient's treatment?, is it possible without any doubt to guarantee a sure treatment for the patient?. This research contributes to generate *ad-hoc* instruments to the patient's necessities and offer as an alternative to the health professionals in order to interpret the derived information of the self-care and responds to the established questions.

In the complex problem solutions as engineering, biology, communications and health [5], the Artificial Intelligence (AI) application is a widely used alternative. The Artificial Neural Networks (ANN) is one of the AI techniques that is used to solve complex problems. Neural networks have the characteristic to function as a Black-Box that transforms an input vector of $m - dimensional$ space to an output vector in $n - dimensional$ space. Table 1 shows a *multi - dimensional* example in the clinical case when two symptoms are presented in a patient that can derive in four possible causes. However, if the symptomatic variables increases it is probable that the diagnosis complicates, and requires a second opinion.

An output function Y_i is shown in table 1, it can take four possible values in function of the combination (X_1, X_2) ,

Cough (X_1)	Headache (X_2)	Diagnose (Y_i)
0	0	Healthy
0	1	Pneumonia
1	0	Flu
1	1	Meningitis

Table 1. Example of the dimensionality in a health problem.

consequently it defines the function as $Y_i = f(X_1, X_2)$, where $(Y_i, X_1, X_2) \in \{0, 1\}$. An ANN can learn the Y_i function and to solve the diagnostic problem described before. Actually, we want the ANN acquires the knowledge of the diagnostic, hence the importance of this kind of AI tool.

2 Objectives

In this research, the general objective is to design an ANN to synthesize the AF of a Pedometer that relates quantity of steps versus calorific burn for the *HLS* Promotion in Diabetic Patients

2.1 Specific objectives

1. Respectively, to build and to define the curve and the AF of the calories burnt in function of number of steps for each person.
2. To design a Feed Forward Neural Network (FFNN), and to train under the Backpropagation algorithm to learn the AF.
3. To evaluate the AF using the Minimum Square Error (MSE).

3 Methodology

3.1 Material

- In this research, two pedometers of the manufacturing Advanced Pendulum DesignTM were used, one economic model of the Series 330 (PT1) and other advanced of the Series 350 (PT2).
- The Matlab Reread 14 (Mathworks Inc.) was used as development software.
- "Walking Advantage ¹" manuals of sportline were used.

¹<http://www.cialtia.com.mx>

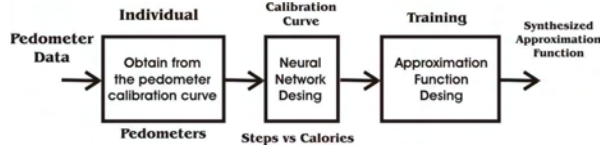


Figure 1. Implemented methodology in order to design the sintezized Approximation Function of the pedometer.

4 Methods

To establish the (AF) construction method of the pedometer figure 1 was followed. It shows the sequence to build the AF from the characteristic curve of the pedometer.

4.1 Obtaining the Calibration Curve of the Pedometers

The first step to synthesize the AF is the construction of the calibration curve of the pedometers. The calibration curve includes calories burn versus the number of steps for each person. In this stage, it was necessary to accomplish the calibration procedure of the mensuration instruments. For the calibration two pedometers were used defined as PT1 and PT2. The PT1 (SPD330) is a basic pedometer which only counts steps, the PT2 (SPD350) is a pedometer that counts steps, calculates distance, calories burn and consequently is considered as the calibration pattern. The achieved calibration procedure making steps mensurations versus calories burnt from three different persons of weight and stride size, which are necessary for the pedometer configuration (See table 2). Configured the PT1 and PT2 pedometers, 10 measures between 35 m and 350 m. (35m, 70m, 105m, 140m, 175m, 210m, 245m, 280m, 315m and 370m) were measure for each person. With PT2 the calories burnt were measured. in order to extract the characteristic curve in each corresponding block, each measurement in both pedometers were averaged and were extracted mean value and standard deviation.

Person	Weight (Kg)	Stride (cm)
In_1	53	75
In_2	83	83
In_3	74	75

Table 2. Configuration parameters for the pedometers.

Therefore, a matrix information was obtained which contains the number of person I_i , where $i = 1, 2, 3$; PT_j the pedometer, where $j = 1, 2$; the k steps measured

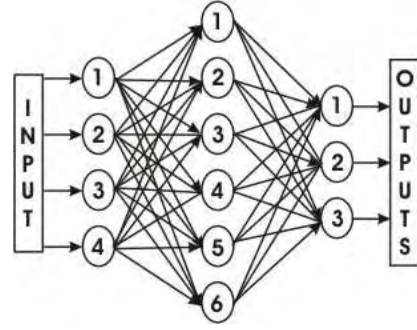


Figure 2. Artificial Neural Network structure applied.

by the j pedometer in the l test is $P_{k,l}^j$ and the calories burnt in the same way are $C_{k,l}^{j=2}$. The mean value of the steps and the calories burnt are $\bar{P}_{k,l}^j = \frac{1}{10} \sum_{j=1,2} P_{k,l}^j$ and $\bar{C}_{k,l}^{j=2} = \frac{1}{10} \sum_{j=2} C_{k,l}^j$ respectively in each distance measured for each person I_i . The $\bar{P}_{k,l}^j$ and $\bar{C}_{k,l}^{j=2}$ mensuration are the couple one that allow the construction of the calibration curve used in the next stage.

4.2 Design Of The Artificial Neural Network (ANN)

The ANN are algorithmic blocks that learn from examples or their input data distribution [6]. Since computers have become more powerful, algorithmic blocks allow the ANN to solve complex problems in many application fields [2]. In this research, in order to synthesize the AF of the pedometer, a progressive ANN [7] has been implemented for three persons. In the experiment, a (4:6:3) structure has been build completely connected with sigmoid activation functions as shown Figure 2, this allows to learn with a single structure the calibration curves obtained in previous stage for the three persons.

4.3 Design Of The Approximation Function (AF)

The design of the AF requires an iterative process in order to train the ANN, as shows Figure 3. The training process consists on introducing each double $(\bar{P}_{k,l}^j, \bar{C}_{k,l}^{j=2})$ to the ANN, where the input will be determined by $\bar{P}_{k,l}^j$ (number of steps) and the output estimated by $\hat{C}_{k,l}^{j=2}$ (calories burnt) that is compared with the real value of calories burn $\bar{C}_{k,l}^{j=2}$ verifying the output error $E = \hat{C}_{k,l}^{j=2} - \bar{C}_{k,l}^{j=2}$. The output error is the most important in the training process because it tries to minimize the error applying the iterative

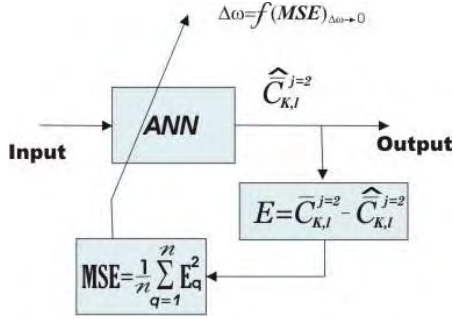


Figure 3. Training process in order to obtain the approximation function minimizing the Mean Squared Error (MSE).

minimization on the Mean Squared Error (MSE), defined as: $MSE = \frac{1}{n} \sum_{q=1}^n E_q^2 = \frac{1}{n} \sum_{q=1}^n [\hat{C}_{k,l}^{j=2} - \bar{C}_{k,l}^{j=2}]^2$, where q , is the q -th element of the calories burn and the estimated calories by the ANN. The training algorithm ends when the MSE is minimum, and when this is achieved, it is said that the ANN has learned the calibration curve and it turns into the AF of the pedometer.

5 Results

In Figure 4, the results of the calibration curves extracted from 10 exercises applied in three persons between 35 and 350 meters. We can appreciate little differences in the graph of the first person, however, when the step number increases, the calories burn coincide. The graphs in Figure 4 shows the mean values $(\bar{P}_{k,l}^{j=2}, \bar{C}_{k,l}^{j=2})$ obtained by the pedometer PT1, that it is lower in cost and the pedometer PT2 chosen as the pattern. Obtained the $(\bar{P}_{k,l}^{j=2}, \bar{C}_{k,l}^{j=2})$ values, the ANN was trained in order to learn the synthesized curves that determine the AF of the pedometer, establishing a minimum error $MSE = 10^{-4}$ and the structure showed in Figure 2, using a *newff* function for a Radial Basis Function Networks. Also, the gradient algorithm was applied in order to train the ANN and to synthesize the AF iteratively, the number of epochs were established in 15000. The result of the training is shown in Figure 5, where we can observe that the MSE was achieved in 12135 iterative epochs. Therefore, the ANN has enough knowledge in order to synthesize the AF of the pedometer and to the measures from the three persons.

6 Discussion

One of the daily life difficulties in the diabetic patients is the physical exercise accomplishment [1], and the physical activity is showed as a pleasure source that helps in the

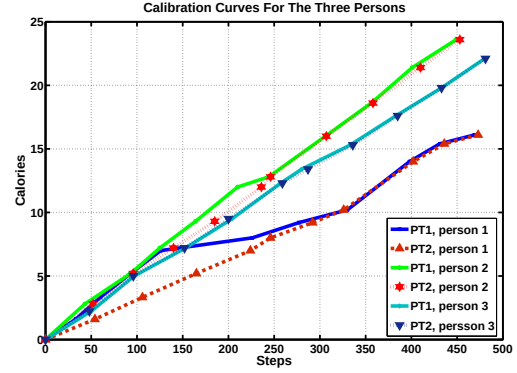


Figure 4. Calibration curves.

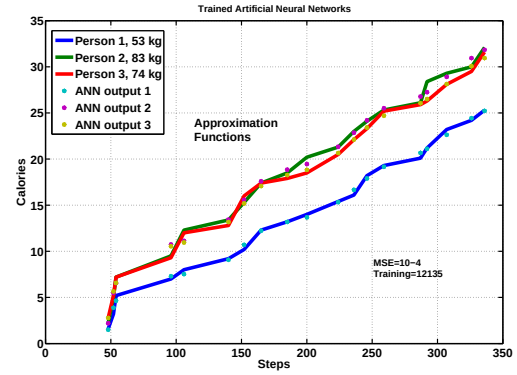


Figure 5. Approximation functions of the pedometers.

weight control and the glucose level in the blood, the persistence of this positive feeling supports the attachment of the patient to the diabetes treatment. In general the patients present low degree of uncertainty regarding to the disease and a high motivation towards the treatment [4]. According to the monitoring development model the treatment includes three phases; 1) the acceptance of the patient with regard to the treatment proposed (that is negotiated among the patient and the health professionals), 2) The Monitoring of the treatment (the patient follows the treatment proposed and his caution health, even if he finds obstacles, and 3) The maintenance of treatment and vigilance of the health (patient takes measures adopted in previous stages with the intention of improving his health and to incorporate them into his life style, transforming the new behaviors in habits. The results in the design of the Approximation Function demonstrate that is possible to train an ANN in order to have reliable data to quantify the physical activity and to arrange information for the adequate insulin supply.

7 Conclusions

- If the information of any pedometer can be validated for a trained ANN, there is a possibility that the health professionals can motivate a bigger number of diabetic patients to maintain an adequate level of physical activity, which contributes with the HLS.
- The studies demonstrate that the physical activity can develop positive feelings in the diabetic patient towards the treatment in anyone of the three phases.
- The result of this study is the ANN trained, which is the input toward the second phase of this investigation, where the ANN application is approve to support the self-monitoring of the groups of mutual help and its attach to the treatment.
- To use an ANN trained in both diagnosis and clinical interpretation has the following advantages: learning, self-organization, failure tolerance, flexibility and real-time response, the advantages present an extensive range of possibilities for the ANN applications in health sciences as a tool for the diagnosis, and the epidemiological surveillance.

Acknowledgments

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References

- [1] Denise Siqueira Pères, Manoel Antônio dos Santos, Maria Lúcia Zanetti and Antônio Augusto Ferronato. Difficulties Of Diabetic Patients In The Illness Control: Feelings And Behaviors. *Revista Latino-Americana de Enfermagem*, 15:1105 – 1112, November-December 2007.
- [2] Diego Andina and Duc Thuong Pham. *Computational Intelligence: for Enineering and Manufacturing*. Springer-Verlag New York, Inc., Secaucus, NJ, USA, 2007.
- [3] Gobierno de los Estados Unidos Mexicanos, Secretaría de Economía, Secretaría de Salud. Norma Oficial Mexicana NOM-015-SSA-1994. In Secretaría de Economía, editor, *NOM-015-SSA-1994*. Diario Oficial de la Federación, 1994.
- [4] João Luís Alves Apóstolo, Catarina Sofia Castro Viveiros, Helena Isabel Ribeiro Nunes and Helena Raquel Faustino Domingues. Illness Uncertainty And Treatment Motivation In Type 2 Diabetes Patients. *Revista Latino-Americana de Enfermagem*, 15:575–582, 08 2007.
- [5] P. J. Lisboa and Piotr S. Szczepaniak and J. C. Mason and Emmanuel C. Ifeakor. *Artificial Neural Networks in Biomedicine*. Perspectives in Neural Computing. Springer-Verlag London Berlin Heidelberg, 2000.
- [6] Pedro Isasi Viñuela and Inés M. Galván León. *Redes de Neuronas Artificiales, Un Enfoque Práctico*. Prentice Hall, 2004.
- [7] Richard O. Duda, Peter E. Hart and David G. Stork. *Pattern Classification*. Wiley Interscience Publication, 2001.

Evolutionary-Neural System to Classify Infant Cry Units for Pathologies Identification in Recently Born Babies

Orion Fausto Reyes-Galaviz
Universidad Autónoma de Tlaxcala
Av. Apizaquito S/N Km 1.5
Apizaco, Tlaxcala. México
orionfrg@yahoo.com

Sergio Daniel Cano-Ortiz
Universidad de Oriente
Santiago de Cuba, Cuba
scano@fie.uo.edu.cu

Carlos Alberto Reyes-García
Inst. Nac. de Astrofísica
Óptica y Electrónica
Puebla, México
kargaxxi@inaoep.mx

Abstract

This work presents an infant cry automatic recognizer development, with the objective of classifying two kinds of infant cries, normal and pathological, from recently born babies. Extraction of acoustic features is used such as MFCC (Mel Frequency Cepstral Coefficients), obtained from Infant Cry Units sound waves, and a genetic feature selection system combined with a feed forward input delay neural network, trained by adaptive learning rate back-propagation. For the experiments, recordings from Cuban and Mexican babies are used, classifying normal and pathological cry in three different experiments; Cuban babies, Mexican Babies, and Cuban & Mexican babies. It is also shown a comparison between a simple traditional feed-forward neural network and another complemented with the proposed genetic feature selection system, to reduce the feature input vectors. In this paper the whole process is described; in which the acoustic features extraction is included, the hybrid system design, implementation, training and testing. The results from some experiments are also shown, in which the infant cry recognition rate obtained is of up to 100% using our genetic system.

1 Introduction

The cry sound produced by an infant is the result of his/her physical and psychological condition and/or from internal/external stimulation. It has been proved that crying caused by pain, hunger, fear, stress, etc. show different cry patterns. An experimented mother can be able of recognizing the singularities between different types of cry, and with this, react adequately to her infant's needs. The experts in neurolinguistics consider the infant cry as the first speech manifestation. It is the first experience on the production of sounds, which is followed by the larynx and oral cav-

ity movements. All of this, combined with the feedback of the hearing capability, will be used for the phonemes production. On these work we focus on recordings from normal and pathological babies, from Mexico and Cuba. The pathological cases are from deaf babies and asphyxiating babies, among others. Children with hearing loss, identified before their first 6 months of life, have a significant improvement in the speech development compared to those whose hearing loss was identified after their first 6 months of age. In the case of the infants that have passed through an asphyxiating period at birth, they are exposed to changes at a neurological level, depending on the asphyxiating range and time they had suffered. According to the American Academy of Pediatrics (AAP), from 2 to 6 out of 1000 recently born babies present asphyxia and 60% of the babies prematurely born and presenting low weight, also suffer an asphyxiating period. From them, 20 to 50% die during their first days of life. From the survivors, 25% develop permanent neurological sequels.

2 State of the Art

Recently, some research efforts on infant cry analysis (ICA) had been made; showing promising results, and highlighting the importance of exploring this field. In [1], Reyes & Orozco classified samples of deaf and normal babies, obtaining recognition results that go from 79.05% up to 97.43%. Petroni used neural networks to differentiate pain and no-pain cries [2]. Tako Ekkel tried to classify newborn crying sound into two categories normal and abnormal (hypoxia), and reports a correct classification result of 85% based on a radial basis neural network [3]. Also, using self organizing maps methodologies, Cano et al, in [4] reported some experiments to classify infant cry units from normal and pathological babies, in these experiments they extracted 12 features directly from the sound waves frequencies, for each sample, the obtained 85% recognition percentage on

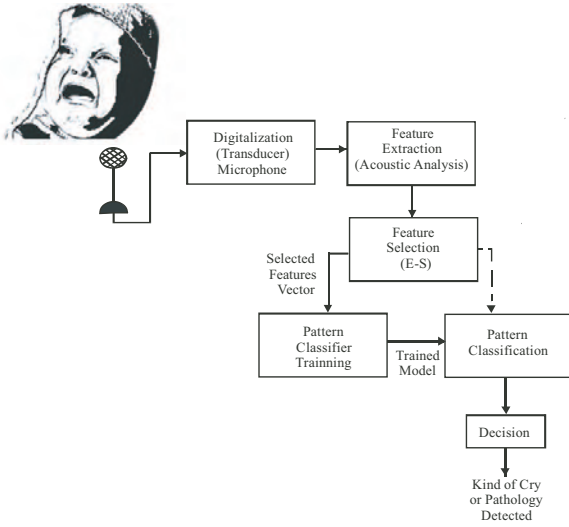


Figure 1. Infant Cry Recognition Process

these experiments.

3 Infant's Cry Automatic Recognition Process

The automatic infant cry recognition process (Fig. 1) is basically a problem of pattern recognition, and it is similar to speech recognition. In our case, the goal is to take baby's cry sound wave units as an input, and at the end obtain the kind of cry or pathology detected. Generally, the Infant Cry Recognition Process is done in two steps; the first step is the acoustic processing, or features extraction, while the second is known as pattern processing or pattern classification. In the proposed system, we have added an extra step between both of them, called feature selection. For our case, in the acoustic analysis, the infant's cry signal is processed to first eliminate silence periods from the sound wave to keep only cry units, and then, from them, to extract relevant features in function of time. The feature set obtained from each cry unit sample is represented by a vector, and each vector is taken as a pattern. Next, all vectors go to an acoustic feature selection module, which will help us to select the best features for the training process, and at the same time to efficiently reduce the input vectors. The selection is done through the use of evolutionary strategies. As for the pattern recognition methods, four main approaches have been traditionally used: pattern comparison, statistical models, knowledge based systems, and connectionist models. We focus in the use of the last one.

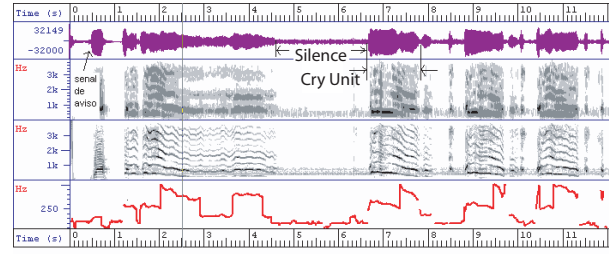


Figure 2. Infant Cry Unit selection and segmentation process

4 Acoustic Processing

The acoustic analysis implies the application and selection of filter techniques, feature extraction, signal segmentation, and normalization. With the application of these techniques we try to describe the signal in terms of its fundamental components. One cry signal is complex and codifies more information than the one needed to be analyzed and processed in real time applications. For this reason, in our cry recognition system we use a feature extraction function as a first plane processor. Its input is a cry signal, and its output is a vector of features that characterizes key elements of the cry sound wave. We have been experimenting with diverse types of acoustic features, emphasizing by their utility Mel Frequency Cepstral Coefficients (MFCC).

4.1 Infant Cry Unit Segmentation

Sergio Cano et al [4] proposed a Cry Unit Segmentation, which consists on a respirational unit (or units) extracted from a single cry sample (Fig. 2). With this method, if one sample has three cry units, we can extract these three samples and exclude the silence contained in the original recording. These cry units were extracted manually one by one on cry samples selected for their clearness, duration and number of cry units contained. Since the acoustic processing requires the same duration on each file, to generate vectors of the same size lengthwise, we segmented each cry unit into 0.4 second samples, this duration was decided by considering the smallest cry unit size obtained after the segmentation.

4.2 Mel Frequency Cepstral Coefficients

The low order cepstral coefficients are sensitive to overall spectral slope and the high order cepstral coefficients are susceptible to noise. This property of the speech spectrum is captured by the Mel spectrum. High order frequencies are weighted on a logarithmic scale whereas lower order

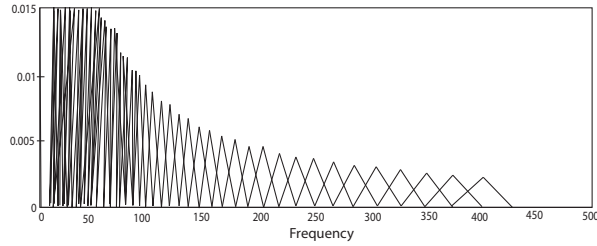


Figure 3. Mel Filter Bank

frequencies are weighted on a linear scale. The Mel scale filter bank is a series of L triangular band pass filters that have been designed to simulate the band pass filtering believed to occur in the auditory system. This corresponds to series of band pass filters with constant bandwidth and spacing on a Mel frequency scale. On a linear frequency scale, this spacing is approximately linear up to 1 KHz and logarithmic at higher frequencies (Fig. 3) [5].

5 Cry Pattern Classification

The acoustic features set obtained in the extraction stage, is generally represented as a vector, and each vector can be taken as a pattern. These vectors are later used to make the acoustic features selection and classification processes. For the present work we are using a classifier corresponding to the type of connectionist models known as neural networks. They are reinforced with evolutionary strategies to select features in order to improve their learning process. Having as a result a Genetic-Neural hybrid system.

5.1 Evolutionary Strategies

The evolutionary strategies are proposed to solve continuous problems in an efficient manner. Its name comes from the German "Evolutionstrategien", so we may frequently see them mentioned as "ES". Their origin was an stochastic scaled method (in other words, following the gradient) using adaptive steps, but with time it has converted in one of the most powerful evolutionary algorithms, giving good results in some parametric problems on real domains. The Evolutionary Strategies make more exploratory searches than genetic algorithms [6].

The main reproduction operator in evolutionary strategies is the mutation, in which a random value is added to each element from an individual to create a new descendant [7]. The selection of parents to form descendants is less strict than in genetic algorithms and genetic programming.

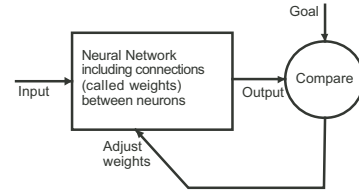


Figure 4. Neural Network Training

5.2 Neural Networks

In a study from DARPA [8] neural networks are defined as systems composed of many simple processing elements, that operate in parallel and whose function is determined by the network's structure, the strength of its connections, and the processing carried out by the processing elements or nodes. We can train a neural network to execute a function in particular, adjusting the values of the connections (weights) between the elements. Generally, the neural networks are adjusted or trained so that an input in particular leads to a specified or desired output (Fig. 4). Here, the network is adjusted based on a comparison between the actual and the desired output, until the network's output matches the desired output [9].

Generally, the training of a neural network can be supervised or not supervised. The methods of supervised training are those used when labeled samples are available. Among the most popular models are the feed-forward networks, trained under supervision with the back-propagation algorithm. For the present work we have used variations of these basic models, which we describe briefly on the next sections.

5.3 Feed-Forward Input Delay Neural Network

Cry data are not static, and any cry sample at any instance in time is dependent on crying patterns before and after that instance in time. A common flaw in the traditional Back-Propagation algorithm is that it does not take this into account. Waibel et al idealized a solution to this problem [12] by proposing a new network architecture called the "Time-Delay-Neural Network" or TDNN. The primary feature of TDNNs is the time-delayed inputs to the nodes. Each time delay is connected to the node via its own weight, and represents input values in past instances in time. TDNNs are also known as Input Delay Neural Networks because the inputs to the neural network are the ones delayed in time. If we delay the input signal by one time unit and let the network receive both the original and the delayed signals, we have a simple time-delay neural network. Of course, we can build a more complex one by delaying the signal at various

lengths. If the input signal is n bits and delayed for m different lengths, then there should be nm input units to encode the total input [9].

5.4 Gradient Descent with Adaptive Learning Rate Back Propagation

The training by gradient descent with adaptive learning rate back propagation, proposed for this project, can be applied to any network as long as its weights, net input, and transfer functions have derivative functions. Back-propagation is used to calculate derivatives of performance with respect to the weight and bias variables. Each variable is adjusted according to gradient descent. At each training epoch, if the performance decreases toward the goal, then the learning rate is increased. If the performance increases, the learning rate is adjusted by a decrement factor and the change, which increased the performance, is not made [10].

The training stops when any of these conditions occurs: *i*) The maximum number of epochs (loops) is reached, *ii*) The maximum amount of time has been exceeded, or *iii*) The performance error has been minimized to the goal.

5.5 Hybrid System

The hybrid system was designed to train the Input Delay Neural Network with the best features selected from the input vectors. To perform this selection, we apply Evolutionary Strategies, which use real numbers for coding the individuals. The system receives the originally sized matrices as an input, and searches by evolutionary techniques, the rows of features that best train a set of Neural Networks. At the end, we obtain an individual with the information of which precise feature rows showed the best overall performance. More information on the design and implementation of this algorithm can be found in [14].

6 System Implementation for the Cry Classification

On the first stage, the infant's cries are collected by recordings obtained directly from doctors of the Mexican National Institute of the Human Communication (INCH, Mexico), the Mexican Institute of Social Security in Puebla, Mexico (IMSS, Puebla), and the Cuban Voice Processing Group (GPV, Santiago de Cuba) at the University of Oriente (UO) with the help of the Hospital Materno Sur in Santiago de Cuba. The cry samples are labeled with the kind of cry that the collector states at the end of each cry recording. Next, we divide each cry recording into its cry units, which later are segmented in samples of 0.4 seconds; these segments are then labeled with a previously established code [13], and each one represents one sample.

In this way, for the present experiments, we have one corpus made out of 334 samples from normal babies, 180 from Mexican babies and 154 from Cuban babies; and 549 from pathological babies, 410 from Mexican babies and 139 from Cuban babies. All these from 0.4 second segments samples. The two corpus were used for separate experiments, as it is later explained. On the next step the samples are processed one by one extracting their MFCC acoustic features, this process is done with the freeware software program Praat 4.2 [11]. The acoustic features are extracted as follows: for each sample we extract 16 coefficients for every 50 milliseconds window, generating vectors of 112 features for each 0.4 second sample. The evolutionary algorithm was designed and programmed using Matlab 7.0 R14, the neural network and the training algorithm were implemented using the Matlab's Neural Network Toolbox [10]. For this work we performed 3 different sets of experiments;

- Cuban Babies Only (C. Babies),
- Mexican Babies Only (M. Babies), and
- Cuban & Mexican Babies (Mixed).

In order to compare the behavior of the proposed hybrid system, a set of experiments are made, where the original vectors are used, without any dimensionality reduction, and later they are reduced to 50 components by means of Principal Component Analysis (PCA), for more comparison results. When we use evolutionary strategies for the acoustic features selection, we search for the best 10, 12, 30, 40 and/or 50 features, these several experiments will show the robustness of the proposed system, and the 12 selected features experiment were selected in order to compare the results with other results in previous works [4]. If n is the number of selected features, Principal Components (PC), or number of the original vector size, then the neural network's architecture consists of n nodes in the input layer, 60% less nodes than the input layer in the hidden layer, and an output layer of 2 nodes, corresponding in pathological and normal cries. The implemented system is interactively adapted; no changes have to be made to the source code to experiment with any input size.

The training is done up to 1000 epochs or until a 1×10^{-8} error is reached. Once the network is trained, we test it using different samples from each class separated previously for this purpose (we used from each corpus 70% for training and 30% features for testing). The recognition precision is shown.

7 Experimental Results

We experimented first with the simple neural network by feeding it the original vector, later the reduced input vectors

Table 1. Results using both corpus, training a simple neural network and using original and reduced vectors.

Experiment	C. Babies	M. Babies	Mixed
Original Vect.	77.38%	100%	86.5%
50 PCs	75%	100%	86.5%

Table 2. Results using both corpus, using the Hybrid System with different feature selection.

Experiment	C. Babies	M. Babies	Mixed
10 Feat.	76.19%	96.29%	86%
12 Feat.	76.19%	98.14%	88.06%
30 Feat.	82.61%	100%	90.29%
40 Feat.	83.93%	100%	90.58%
50 Feat.	91.07%	100%	92%

to 50 PCs, since this number of PCs yielded good results on past experiments [1], [13], and finally with our hybrid system to compare the obtained results. In all these experiments we use 1 time delay unit.

In the case of the simple neural network system, we perform three experiments and choose the best result. As for the hybrid system, to search for the best solution in a multi solution space, only one experiment is done. We do so because we use 15 individuals as the initial population, 20 generations to perform the features search, and the size of the individuals was of 10, 12, 30, 40, and 50 chromosomes. With all these input parameters, there are 300 different training processes needed, which takes much more time to perform each experiment. The results are shown in Table 1, where the second column corresponds to the results obtained from Cuban Babies, the third one shows the results obtained from Mexican Babies, and the last column shows the results obtained by using the Mexican & Cuban cry samples.

On Table 2, we can see the results by using 10, 12, 30, 40, and 50 features selected with the aid of the hybrid system. The only difference on the experiments, is that in the first case, where reduction is taking place with PCA, the reduction of vectors is done before any processing by the neural network. While in the second case, feature selection is made concurrently to the neural network training. On this basis, we are presenting our model as an evolutionary-neural hybrid system.

8 Conclusions and Future Work

The application of feature selection methods, on different kinds of pattern recognition tasks, has become a viable alternative tool. Particularly for those tasks which have to deal with input vectors of large dimensions. As we have shown, the use of evolutionary strategies for the selection of acoustic features, in the infant cry classification problem, has allowed us, not only to work with reduced vectors without losing classification accuracy, but also to improve the results compared to those obtained when applying PCA. Moreover, when we combine the Mexican Cry and the Cuban Cry recordings, the results are also improved by reinforcing the corpus information, and the neural networks are consistent with both corpus.

For future works, in order to improve the performance of the described evolutionary-neural system, we plan to do more experiments using other neural network configurations, a larger number of individuals, and more generations. We are also looking for adequate models to dynamically optimize the parameters of the hybrid model, in order to adapt it to any type of pattern recognition application.

As for the automatic infant cry unit recognition problem, we will continue experimenting with some different types of hybrid systems, such as instance and feature selection hybrid models, boosting neural network ensemble and boosting ensemble of support vector machines. And once we can assure that our system is robust enough to identify the mentioned pathologies, we will try to increment our infant cry corpus with the same kind and some other kinds of pathologies. Particularly with the type of pathologies related to the central nervous system (CNS). This is done firstly because with more cry samples from deaf, asphyxiating, and pathological babies, from anywhere in the world, we can implement a more reliable learning and recognition of those cry classes. Secondly, by adding cries from infants with other pathologies, we can direct the early diagnosis to identify such pathologies.

References

- [1] Orozco Garcia, J., Reyes Garcia, C.A. (2003), Mel-Frequency Cepstrum coefficients Extraction from Infant Cry for Classification of Normal and Pathological Cry with Feedforward Neural Networks, ESANN 2003, Bruges, Belgium.
- [2] Marco Petroni, Alfred S. Malowany, C. Celeste Johnston, Bonnie J. Stevens, (1995). Identification of pain from infant cry vocalizations using artificial neural networks (ANNs), The International Society for Optical Engineering. Volume 2492. Part two of two. Paper #: 2492-79.

- [3] Ekkel, T, (2002). "Neural Network-Based Classification of Cries from Infants Suffering from Hypoxia-Related CNS Damage", Master's Thesis. University of Twente, The Netherlands.
- [4] Sergio D. Cano, Daniel I. Escobedo y Eddy Coello, "El Uso de los Mapas Auto-Organizados de Kohonen en la Clasificacin de Unidades de Llanto Infantil", Voice Processing Group, 1st Workshop AIRENE, Universidad Catlica del Norte, Chile, 1999, pp 24-29.
- [5] Gold, B., Morgan, N. (2000), Speech and Audio Signal Processing. Processing and perception of speech and music. John Wiley & Sons, Inc.
- [6] Santo Orcero, David. Estrategias Evolutivas www.orcero.org/irbis/disertacion/node217.html. 2004.
- [7] Hussain, Talib S., An Introduction to Evolutionary Computation, Department of Computing and Information Science Queens University, Kingston, Ont. K7L 3N6. 1998.
- [8] DARPA Neural Network Study, AFCEA International Press, 1988, p. 60.
- [9] Limin Fu., Neural Networks in Computer Intelligence. McGraw-Hill International Editions, Computer Science Series, 1994.
- [10] Neural Network Toolbox Guide, Matlab V.6.0.8, Developed by MathWoks, Inc.
- [11] Boersma, P., Weenink, D. Praat v. 4.0.8. A system for doing phonetics by computer. Institute of Phonetic Sciences of the University of Amsterdam. February, 2002.
- [12] A. Weibel, T. Hanazawa, G. Hinton, K. Shikano, and K.J. Lang, "Phoneme Recognition Using Time Delay Neural Networks," IEEE Trans. Acoustics, Speech, Signal Proc., ASSP-37: 32'339, 1989
- [13] Orion F. Reyes Galaviz. "Clasificacin de Llanto de Bebs para Identificacin de Hipoacsia y Asfixia por medio de un Sistema Hbrido (Gentico Neuronal)" Master's Thesis on Computer Science, at the Apizaco Institute of Technology (ITA), March, 2005. <http://solar6.ingenieria.uatx.mx/orionfrg/tesis.pdf>
- [14] Orion F. Reyes Galaviz, Carlos A. Reyes Garca "Infant Cry Classification to Identify Hypoacoustics and Aphyxia Comparing an Evolutionary-Neural System with a Simple Neural Network System" en MICAI 2005. Monterrey Mexico, noviembre14-18, 2005. Lecture Notes on Artificial Intelligence 3789. ISBN 3-540-29896-7 ISSN 302-9743.

Applications

Artificial Intelligence Applied into Pneumatic Flexible Manipulator

Juan-Manuel Ramos-Arreguin¹, Jesus Carlos Pedraza Ortega², Efren Gorrostieta³ and Rene de Jesus Romero-Troncoso⁴

1 Universidad Tecnológica de San Juan del Río; jramos@mecamex.net

2 Universidad Autónoma de Querétaro, Facultad de Informática; e-mail: efren.hurtado@usa.net

3 Facultad de Ingeniería Mecánica, Eléctrica y Electrónica; e-mail: troncoso@salamanca.ugto.mx

5 Universidad Autónoma de Querétaro, Facultad de Informática; e-mail: caryoko@yahoo.com

Abstract

Several researches with flexible manipulator robot had been developed using an electrical actuator while other research works deal with pneumatic control, but both investigation lines had been used in separate way. Our goal is simulate the control position angle for the flexible arm, limited to one degree of freedom. We propose an interaction between the pneumatic control and the flexible manipulator robot, being this, the original part of the present work. In this proposal, a speed feedback is used for control position, taken from the pneumatic actuator displacement. We use a fuzzy logic algorithm to adjust the control values, getting a fuzzy algorithm with better results than classic control PID. The project is focused to the engineering application in pneumatic control and flexible manipulators. Result shows a soft movement behavior when speed feedback is used, combined with fuzzy logic, let us to get better results than the classic control approach.

Keywords: Fuzzy control, PID, flexible manipulator, adaptive process, pneumatic actuator.

1. Introduction

In principle, a flexible manipulator robot with pneumatic actuator is light, cheap, and has the advantage to handle a higher power-weight, with respect to robots with electric actuators [1]. The modeling of the flexible manipulators has been made for almost 35 years [2] [3], where almost all the works use electrical or hydraulic actuators, and those of pneumatic type are less used due to their non linear nature. Later, an analytical method was used to

examine the performance of a flexible manipulator, using a high-torque stepper motor. Then, a modeling and control was developed for a two-link flexible robot arm [4], without actuator, just simulation. Moreover, a position control was developed using fuzzy neuronal algorithm [5]. Also, a robust control was developed for one-link flexible robot [6], where the actuator is a DC motor and a non linear control was developed using a motor shaft [7]. Finally, an artificial intelligent control was implemented in a single-link flexible robot arm, driven by a permanent magnet (PM) synchronous servo motor [4]. Along this research line, a Thermo-mechanical model has been developed and a dynamic model analysis was developed [9], but these works are not considering a pneumatic control for an actuator applied in a flexible manipulator robot, and this work is focused in it.

Based on this type of works, a flexible manipulator with pneumatic actuator was developed, where a mechanical system and pneumatic behavior are involved, to give a flexible arm movement [9], which leads to the Thermo-Mechanical model. From the point of view of the control engineering, this model allows to predict the behavior of different variables that take part in the physical process to control.

2. Thermo-mechanical model

Figure 1(a) shows a diagram of the pneumatic actuator, where X , $\dot{X}$, $\ddot{X}$ are the position of the piston actuator, the speed and the acceleration of the piston rod with respect to the cylinder, respectively; additionally we have the internal pressures P_{a1} , P_{c1} , P_{c2} and P_{a2} , that appear at the left side pad of the cylinder, in the chamber of the piston at the left side of the rod, the chamber at the right side rod, and at the

pad of right side of the cylinder, respectively; the actuator force F_a ; A_1 , A_2 and A_3 are the free air flow area of the valves at the left side of the cylinder, at the right side of the cylinder and the air return; a 5/2 valve is used. Although the model needs the three valves, and we use only two, considering the valves A_1 and A_2 with the same value, due to the fact that when the rod comes down, the valve enabled is A_2 and when it goes up, A_1 is enabled, therefore the used configuration is like the one shown on figure 1(b).

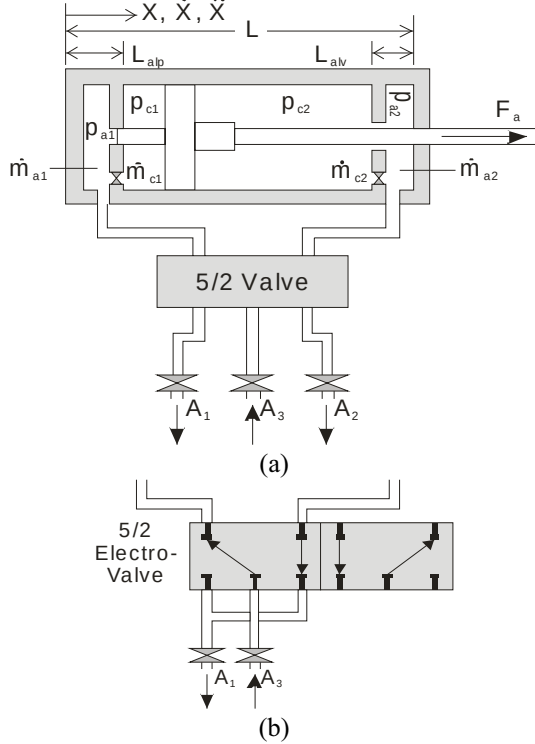


Figure 1. Diagram of a pneumatic piston with damping on both sides. (a) The three valves needed for the mathematical model. (b) Using only two valves for simulation and practice process, where $A_1 = A_2$.

The set of equations (1) shows the Thermo-Mechanical model, which describes the dynamics of the pneumatic actuator. The model calculates the changes of the internal pressures of the cylinder; P_{a1} , P_{a2} , P_{c1} and P_{c2} , depending of the cylinder's rod position. Due to the effects generated by the damping pads at the ends of the cylinder, it is necessary to divide the Thermo-Mechanical model in three intervals; pad of the piston side ($0 \leq X < L_{alp}$), middle ($L_{alp} \leq X < L - L_{alv}$), and the pad of the piston rod side ($L - L_{alv} \leq X \leq L$), as observed in the equation set (1).

For the interval $0 \leq X \leq L$:

$$\dot{X} = \frac{d}{dt} X \quad (1a)$$

$$D\dot{X} = \frac{d^2}{dt^2} X \quad (1b)$$

For the interval $0 \leq X \leq L_{alp}$:

$$\dot{P}_{a1} = \frac{kRT_0}{A_{ap} \left(X + \frac{\Delta A_p}{A_{ap}} \right)} \left[\dot{m}_{1a} - \dot{m}_{1c} - \frac{A_{ap}}{RT_0} P_{a1} DX \right] \quad (1c)$$

$$\dot{P}_{c1} = \frac{kRT_0}{(A_p - A_{ap})X} \left[\dot{m}_{1c} - \frac{(A_p - A_{ap})}{RT_0} P_{c1} DX \right] \quad (1d)$$

For the interval $L_{alp} < X \leq L$:

$$\dot{P}_{a1} = \frac{kRT_0}{A_p (X + \Delta)} \left[\dot{m}_{1a} - \frac{A_p}{RT_0} P_{a1} DX \right] \quad (1e)$$

$$\dot{P}_{c1} = \frac{kRT_0}{A_p (X + \Delta)} \left[\dot{m}_{1c} - \frac{A_p}{RT_0} P_{c1} DX \right] \quad (1f)$$

For the interval $0 \leq X < (L - L_{alv})$:

$$\dot{P}_{c2} = \frac{kRT_0}{(A_p - A_v)(L - X + \Delta)} \left[\dot{m}_{2c} + \frac{(A_p - A_v)}{RT_0} P_{c2} DX \right] \quad (1g)$$

$$\dot{P}_{a2} = \frac{kRT_0}{(A_p - A_v)(L - X + \Delta)} \left[\dot{m}_{2a} + \frac{(A_p - A_v)}{RT_0} P_{a2} DX \right] \quad (1h)$$

For the interval $(L - L_{alv}) \leq X \leq L$:

$$\dot{P}_{c2} = \frac{kRT_0}{(A_p - A_v)(L - X)} \left[\dot{m}_{2c} + \frac{(A_p - A_v)}{RT_0} P_{c2} DX \right] \quad (1i)$$

$$\dot{P}_{a2} = \frac{kRT_0}{L - X + \frac{\Delta A_p}{A_{av} - A_v}} \left[\frac{\dot{m}_{2a} - \dot{m}_{2c}}{A_{av} - A_v} + \frac{P_{a2} DX}{RT_0} \right] \quad (1j)$$

Where A_p , A_v , A_{av} and A_{ap} , are the free air flow area from the outside inwards of the cylinder's left side, from the outside inwards of the right side of the cylinder, at the left pad, and at the right pad of the cylinder, respectively.

3. Fuzzy control system

On figure 3, the block diagram of the control proposal is shown. In this diagram a PID control is involved by using a feedback, where an adjustment is carried out on the control parameters. The details will be explained in next section. As we can observe on figure 3, θ_p is the set point, θ is the current position, u is the vector of control variables formed by the free air flow on the valves, as shown in the equation (2). Also, e is the position error in the output angle of the

mechanism at the time T_k , and it includes the proportional error e_p , integral error e_i , and derivative error e_d , as observed in the equations (3) to (5), then ΔV is the speed difference at two intervals of time defined in the equation (6), and finally K is the set of control values for the PID controller (7).

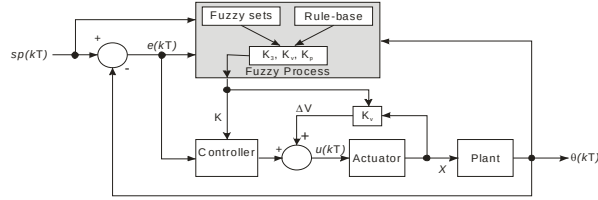


Figure 3. Diagram to blocks of the fuzzy control for the Thermo-Mechanical model.

$$u = [A_1, A_2, A_3] \quad (2)$$

Where A_1 , A_2 and A_3 , are the areas of the external valves, used by the controller, as shown in the figure 1(a). The valve A_3 is the pressure input, and the valves A_1 and A_2 are the air return to the atmosphere. In the next section we will talk more about it.

$$e_p(T_k) = \theta_p - \theta \quad (3)$$

$$e_i(T_k) = e(T_k) + e(T_{k-1}) + e(T_{k-2}) \quad (4)$$

$$e_d = e(T_k) - e(T_{k-1}) \quad (5)$$

$$\Delta V = V(T_k) - V(T_{k-1}) \quad (6)$$

$$K = [K_p, K_d, K_i, K_v] \quad (7)$$

The K vector represents the fuzzy adaptation process for the proportionality, derivative, integral and speed constants, obtained by a fuzzy logic algorithm to improve the behavior of the system. In this control scheme, the feedback of the speed changes is important, due to the fact that a better behavior at the plant output was obtained; avoiding abrupt changes in the displacement speed generated by the pneumatic actuator, as well as the angular speed behavior of the plant, and therefore a vibration problem in the system is avoided. The control equation that defines the air free flow area is defined in the equation (8).

$$A_j(T_k) = A_{j0} + K_{pj}e_p(T_k) + K_{ij}e_i(T_k) + K_{dj}e_d(T_k) + K_v\Delta V(T_k) \quad (8)$$

Where T_k represents the sample time, considered of 50 ms; j represents the valve which is controlled; K_p , K_d , and K_i , are the constants of the law of the PID

control, and K_v is the speed change constant. The K vector values are obtained using a fuzzy logic algorithm, which will be explained in the next section.

4. Fuzzy control algorithm

Due to the fact that a combination of high non linearity of the pneumatic system and the thermo-mechanical model complexity, a single rule set can not be used. When the rod goes up, it has a particular behavior, but when it goes down a different behavior (hysteresis) occurs, very close to one of the cylinder sides where a damping effect is present. Therefore, the use an individual fuzzy set for each control parameter and different set of rules is necessary. In this section we talk about how we get the fuzzy sets to the flexible manipulator control and the application in the equation (8).

The fuzzy process considers several inputs signals: like the error (e_p), the set point (θ_p), and the output angle (θ). The outputs are the K_p , K_i , K_d and K_v signals applied to the control equation (8). As starting point, a system characterization is performed in a hand-tuning way, to get an idea about the intervals we need to use, and later the intervals were modified to get an improved performance. We are using the direct Mamdani method (Kazuo, 1997), and the defuzzification method is the centroid. To perform the control evaluation, we are using the MATLAB software with the Fuzzy Logic Toolbox, and we use the thermo-mechanical model as the plant.

To find the fuzzy sets needed to control the system, first we consider K_v , K_3 , K_i and K_d equal to zero, and only apply the fuzzy process to K_p . Once we get a good response, a fuzzy process is applied to K_i , later to K_d , K_v , and finally to K_3 . Next, we must do a few changes in order to obtain a better system behavior. Now, we explain the fuzzy sets and rules applied to each control parameter.

4.1. The K_p parameter

The fuzzy process for K_p needs three inputs and one output, as shown in figure 4. The fuzzy sets for all inputs and outputs use a triangular shape membership functions for each class. The classes for the input theta are extreme, low and positive. The classes for the input error are; negative big, negative half, negative small, zero, positive small, positive half and positive big. The classes for the input set point are; set point 1, and set point 2. The classes for the output are; small, half, big and very big. This parameter defines the system speed. The complete fuzzy rules set are shown in table 1.

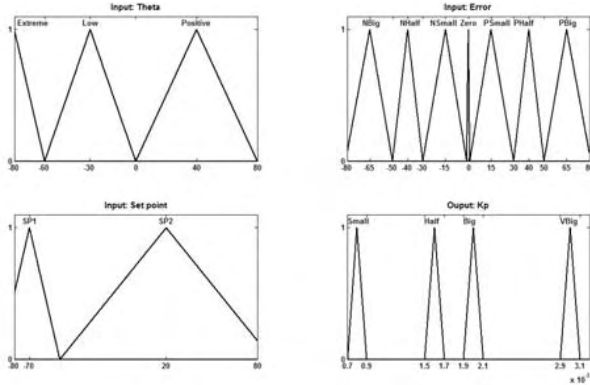


Figure 4. Fuzzy sets for K_p control constant.

Table 1. Rules used for K_p .

Rule 1	If e is NSmall then K_p is Big
Rule 2	If θ_p is SP1 then K_p is Big
	If θ is not Extreme and e is Zero then K_p is Small
Rule 4	If θ is not Positive and e is PSmall then K_p is Small
Rule 5	If θ is not Positive and e is PHalf then K_p is VBig
Rule 6	If θ is not Positive and e is PBig then K_p is Big
Rule 7	If θ is Positive and e is PSmall then K_p is Big

4.2. The K_v parameter

The fuzzy process for K_v has the same inputs than parameter K_p , as show in figure 5, using triangular membership functions for each class. The classes for the theta input are; low and high. The classes for the input error are; negative and positive. The classes for the input set point are; negative 5, negative 4, negative 3, negative 2, negative 1 and positive. The classes for the output are; zero, very small, small, half, regular and high. The K_v parameter helps us to avoid an abrupt change of the speed in the system movement. The complete set of fuzzy rules are shown in table 2.

4.3. The K_3 parameter

The fuzzy process for K_3 needs only one input and one output, use triangular membership functions for each class as shown in figure 6. The classes for the input set point are; very down, negative down, regular down, down and up. The classes for the output are; very few, few, half, high, very high and all. This parameter helps to control the flow air input of the system, and due to this effect, it is important to

minimize the overshoot at the output variable, θ . Then, the fuzzy rules are shown in table 3.

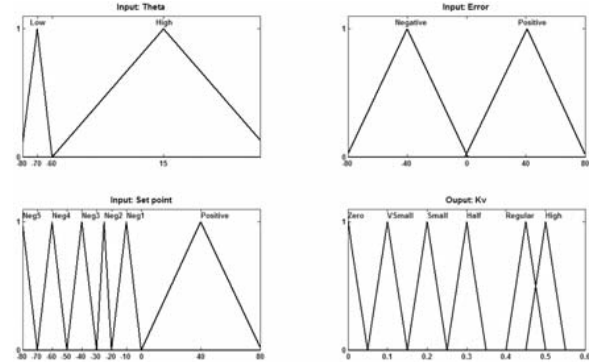


Figure 5. Fuzzy sets for K_v control constant.

Table 2. Rules used for K_v .

Rule 1	If θ is Low or e is Negative and θ_p is Positive then K_v is Small
Rule 2	If θ is High and e is Positive and θ_p is Neg1 then K_v is Regular
Rule 3	If θ is High and e is Positive and θ_p is Neg2 then K_v is Half
Rule 4	If θ is High and e is Positive and θ_p is Neg3 then K_v is VSmall
Rule 5	If θ is High and e is Positive and θ_p is Neg4 then K_v is Half
Rule 6	If θ is High and e is Positive and θ_p is Neg5 then K_v is High

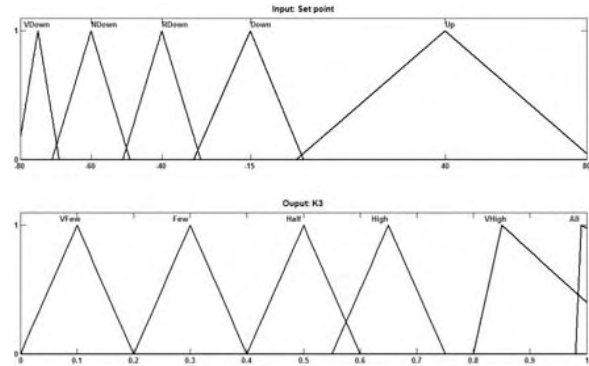


Fig. 6: Fuzzy sets for K_3 control constant.

Table 3. Rules used on Fuzzy2.

Rule 1	If θ_p is Up then K_3 is All
Rule 2	If θ_p is Down then K_3 is VHigh
Rule 3	If θ_p is RDown then K_3 is Half
Rule 4	If θ_p is NDown then K_3 is Few
Rule 5	If θ_p is VDown then K_3 is VFew

4.4. The K_d and K_i parameters

The optimal value of those parameters happens to be zero. To obtain this optimal value, the control parameters K_d and K_i are not used in control equation (8). We conclude it by hand-tuning method applied to the simulation process.

4.5. The final control equation

The equation (9) shows the final control equation. The values of the valves represent a percentage of the aperture, where 0 means totally closed and 1 is totally open. The valve A_2 is equal to the A_1 , because in practice it is the same valve, but the mathematical model needs the three values. The value of A_3 depends of A_1 , we only need to control the air return of the system, to get a better behavior; this valve is important to stop the piston displacement when the arm is close to the set point.

$$\begin{aligned} A_1(T_k) &= A_{10} + K_p e_p(T_k) + K_v \Delta V(T_k) \\ A_2(T_k) &= A_1(T_k) \\ A_3(T_k) &= K_3 A_1(T_k) \end{aligned} \quad (9)$$

4.6. Experimental description

The figure 7 shows the flexible manipulator prototype simulation with pneumatic actuator, where the arm is made of PVC material. Remember that on figure 1 we have a scheme that shows the air flow in both directions, and the air return is controlled by two independent proportional valves. Then, these valves are used to avoid that some of the cameras remains without pressure, generating a braking effect to the advance of the piston.

5. Results

Due to the high non-linearity of the system, the hand-tuning method is used to set the member ship intervals used in the fuzzy process for each input and output. First, the integral, derivative and speed parameter (K_i , K_d and K_v), are setting 0, and K_3 is equal to 1. The K_p value start in 0.1 and is increased until an oscillation is present. As second step, K_3 is decreased until the output has a behavior without an oscillation. The third step is to adjust the K_v to avoid the vibration problem on the system, getting a behavior without a fast speed changes. This procedure is used with different set points to test all interval of θ for the

arm. The K_3 value is important because is the pneumatic brake for the system and depends of the reference, in other words, the major problem to control the system, is when the reference is negative, due to the gravity and the mechanical characteristics, as much negative is the reference, K_3 is close to zero. The Error input is used to know if the arm goes up or goes down. The K_i and K_p are not necessary to get a nice system behavior.

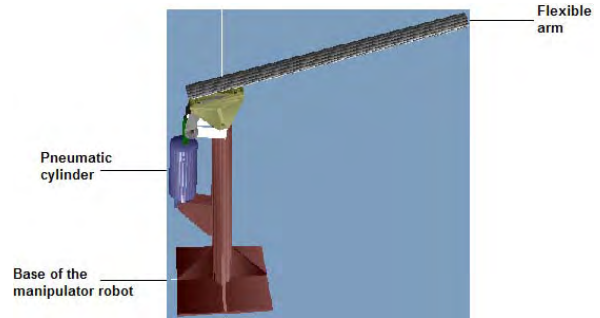


Figure 7. Flexible manipulator robot prototype with pneumatic actuator.

The reference values used to test the control are: 60° , 12° , -72° , 25° and 79° . The results are shown in figure 9, including a comparison of results between classic control and fuzzy control. In the figure 9 (a), we can see the rod movement. The system needs 1.9 seconds to go from 0 m to the 63.2% of 0.02 m, with a slope of 0.5263 seg^{-1} . The response arrives to the 98% of the final value in 5 seconds. The figure 9 (b) shows the angle of the flexible arm, also by observation we can see that when the angle θ is greater than 0, the system has a different behavior when θ is smaller than 0. When θ is greater than 0, the system needs 5 seconds to arrive at 98% of set point, when is going down and 2 seconds when it goes up. When θ is smaller than 0, the systems needs 6 seconds to get a 98% of set point going in down direction, and 3 seconds in up direction. The figure 9(c) is a comparison between the PID control and fuzzy control response. The figure 9(c) shows a better control behavior when the fuzzy algorithm is used than the PID control algorithm, especially when the set point is close to -80° . At this point, the rod position is close to 0.1 m, and the rod is close to the damping area, and it must support the inertial forces generated by the mechanism weight.

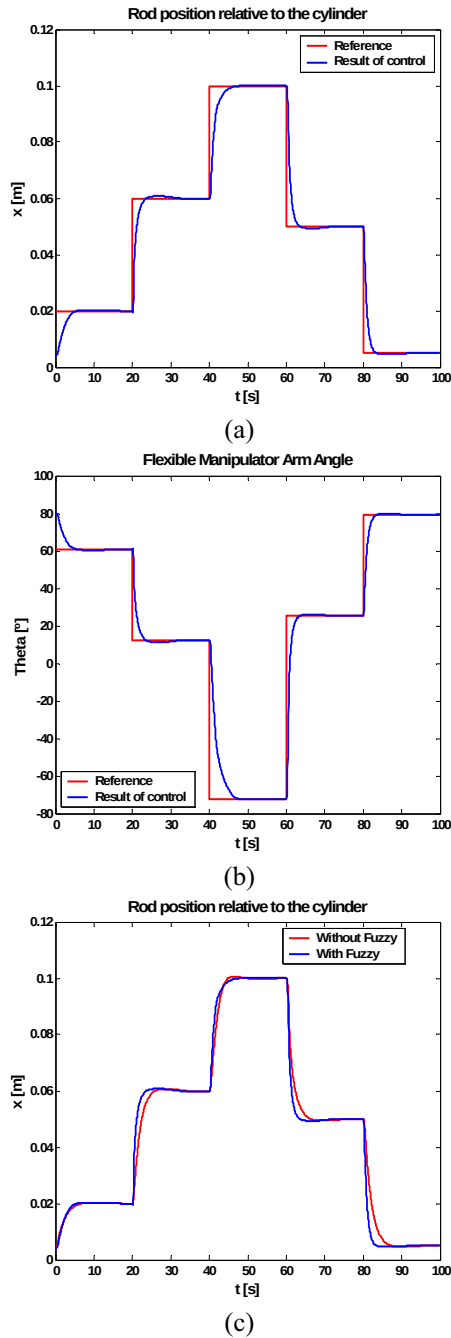


Figure 9. Control results of the flexible manipulator robot and comparison between fuzzy control and PID control. (a) Rod displacement; (b) Output angle behavior, θ ; (c) Comparison between a PID control and Fuzzy control.

By trying to use the K_d and K_i parameters, a small oscillation is present in the system, as shown in figure 10. For that, we are not using these parameters.

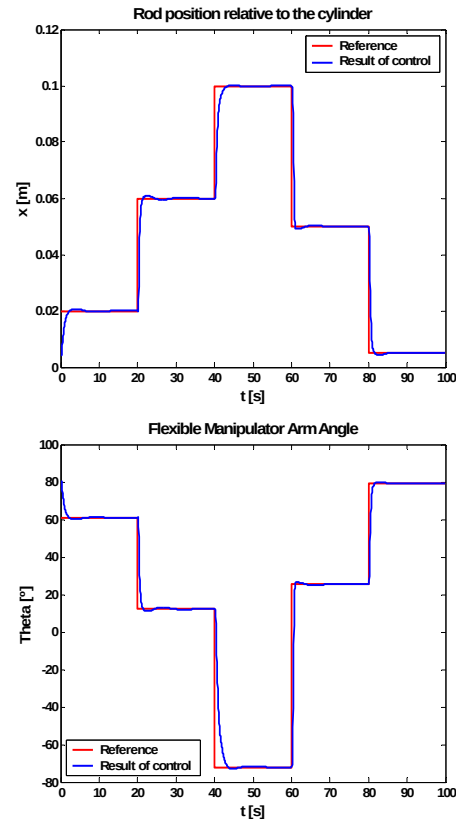


Figure 10. System behavior when K_i and K_d are used.

6. CONCLUSIONS

The novelty of this work is in the establishment of the connection between the pneumatic control and flexible manipulating robots lines, when applying a pneumatic actuator as an element of force generation for its movement, instead of an actuator of electrical type. Taking the advantage of being a light and clean manipulator with a greater relation force-weight.

A control alternative is proposed, by means of an adaptive fuzzy system, making an adjustment to the control parameters, where the feedback of the speed differences remarkably improves the response of the system output. The feedback of the speed difference is a good option, when it works with nonlinear systems, and its speed behavior has too many variations that can cause vibrations in the final effector. If a classical PID control is used, to get a good behavior, the space work's arm must be divided in several intervals, with different constant values, and movement direction is important, as shows in simulation Juan et al. (2006). For it, the PID control comes to be complex, and a Fuzzy control is easier to do an adjustment and a better result is obtained.

Other reported techniques use systems with pneumatic servomechanisms to generate the movement, for example the proportional valves, nevertheless, they do not consider the rest of the system in the aspect of dynamics [11]. In this case, a complete system is considered, that is, a pneumatic actuator and the structure have been included in the Thermo-mechanical model of a manipulator with one degree of freedom, to obtain a position controller of the flexible arm.

This approach is successful to control the position angle of the flexible arm, having as result a smooth movement, besides to have started an interaction between the work of pneumatic control and flexible manipulator lines.

7. References

- [1] A.B. Smith, C.D. Jones, and E.F. Roberts, "Article Title", *Journal*, Publisher, Location, Date, pp. 1-10.
- [2] Jones, C.D., A.B. Smith, and E.F. Roberts, *Book Title*, Publisher, Location, Date.
- [1] M. Constantinos. "Development of Advanced Actuators Using Shape Memory Alloys and Electrorheological Fluids", *Research in Nondestructive Evaluation*, Springer New York ISSN 0934-9847, 14, 1-32. 2002.
- [2] John M., "Automatic Feedback Control of a Vibrating Flexible Beam". *MS Thesis*, Department of Mechanical Engineering, Massachusetts Institute of Technology, 1972
- [3] Book W.J., Daniel W.E., Paul L.M., "Design and Control Considerations for Industrial and Space Manipulators", *Proceedings of the Joint Automatic Control Conference*, Austin, TX, 591-598, 1974
- [4] Xuru D., Tzyh J.T., Antal K.B., "A Novel Approach to the Modeling and Control of Flexible Robot Arms", 1988
- [5] Yeon G.C., Han H.T., Chang G.K., "The Study on Position Control of a Flexible Robot Manipulator using Fuzzy Neural Networks", *Third International Conference on Knowledge-Based Intelligent Information Engineering Systems*, Adelaide, Australia, 226-229, 1999.
- [6] Panu, C., David B., "A Case Study of Robust Control Experiment on One-link Flexible Robot Arm", *Proceedings of the 38th Conference on Decision and Control*, Phoenix, Arizona USA, 4319-4324, 1999.
- [7] Hisseine D., Lohmann B., "Nonlinear Tracking Control for a Lightweight Flexible Robot", *IEEE International Conference on Systems, Man, and Cybernetics*, Nashville, TN, USA, 5, 3360-3365, 2000.
- [8] Rong J.W., Meng C.L., "Intelligent Optimal Control of Single-Link Flexible Robot Arm", *IEEE Transactions on Industrial Electronics*, 51, 201-220, 2004.
- [9] Fernando F.K.M., José E.V., "Dynamic Model Analysis of a Pneumatically Operated Flexible Arm", *WSEAS Transactions on Systems*, 4, 49-54, 2005.
- [10] Juan M.R., Efrén G., José E.V., José C.P., René J.R., Bernardo R.P., "Pneumatic Cylinder Control for a Flexible Manipulator Robot", *International conference on Dynamics, Instrumentation and Control (CDIC'06)*, Querétaro, México, 2006.
- [11] Janiszowski K.B., "Adaptation, Modelling of Dynamic Drives and Controller Design in Servomechanism Pneumatic Systems", *IEE Control Theory and Applications*, 151, 234-245, 2004.

Knowledge Consistency Maintenance Architecture for Bullwhip Effect in Global SCM

Hsieh Ching-Shun¹, Hung Jui-Wen², Lo Kuo-Yen³

¹Department of Commercial Technology and Management, Ling Tung University, Taiwan, R.O.C

^{2,3}Department of Information Management, Ling Tung University, Taiwan, R.O.C

¹sjsoon@mail.ltu.edu.tw, ²hung@mail.ltu.edu.tw, ³log@mail.ltu.edu.tw

Abstract

With e-business and enterprise global layout trend, SCM efficiency is a key of improving competitions. In global transnational enterprise, widely area duplicates inconsistency issues became seriously in the SCM as a result of using caching, middleware and database replication technologies to improve Bullwhip Effect problem. We proposed novel global knowledge message consistency model - DHWSRP (Distributed Heterogeneous Web Service Routing based Portal) that can ripple propagate updating SCM copy that existed in internal heterogeneity of enterprise database with specified OKMS (Original Knowledge Message Source) partially dependency relationships. The thesis also proposed a new scalable replica routing policies dynamic reconfiguration distributed forwarding data path in hierarchical enterprises knowledge database. On the basis of the ubiquitous data consistency framework, many active updating flows can be pushed by AI agents using real time, periodic, reservation, and queuing with priority/degradation/preemption etc. methods. Apply scenario can eliminate in GSCM (global SCM).

1. Introduction

The wide area global knowledge message consistency [3][4] in GTHEO (Global Transnational Hierarchy Enterprise Organizations) become increasingly important in the GSCM[5][6] as a result of using caching, middleware and database replication technologies to improve it. The study was divided into 2 steps. A novel global dependent replica updating platform was proposed in first step, which consists of active, real time, automation, and global FPC (Forwarding Portal Clusters) web services routing technologies. When OKMS object was triggered by update event, all replicas with the specific OKMS dependent relationship could be automatically and real time updated. The second step propose a specific

OKMS dependent replicas consistency maintenance project which provide adaptive adjustable update routing policies, and value added update services based on usage mining results of replicas with specific OKMS dependency relationships」.

Section 2 of this paper summarizes related work. The active OKMS dependency copies wide area consistency processing models are presented in Sect. 3-1. In Sect. 3-2, we illustrate a OKMS agents [7] and web service [8][9][10][11] driven architecture. In Sect. 3-3, we illustrate the agent through a variety of flexibility and customization message publishes mechanism to transfer OKMS update copy. In Sect. 3-4, Forwarding EP (Enterprise Portal) agent use pipeline mechanism automatically conversion of all kinds of knowledge message format and update GTHEO replicas with OKMS dependencies. In Sect. 3-5, we show the system UML Sequential diagram. The OKMS hierarchical ripple propagation routing algorithm discussion is given in Sect. 4. Finally, the distribution of web service processing functions over distinct and heterogeneous entities must be achieved according to capabilities. Current cluster based EP do not provide these needed mechanisms and architectures to create a distributed message path composed of multiple web service processing functions located in different clusters. This is exactly the aim of DHWSRP which provides load balancing and high performance forwarding using the optimization framework proposed in Sect. 5.

2 Related work

Enterprises application must deal with more replica data than ever before, but the replica dataset original version is locked up in a patchwork of disparate systems: legacy systems, relational databases, flat files, document repositories, data warehouses, Microsoft excel files, web pages, email and more other data sources. Beyond a company's firewall is an ocean of additional data in partner, customer and supplier

systems. Because these systems were never designed to work together, it's extremely challenging to take full advantage of the information locked up in all that data. Yet today's competitive market requires that companies do exactly that.

Today's data integration solution [12][13] have been proposed of several kinds data Integration middleware. If replicas having partial dependency relationships with specific OKMS object inside GTHEO can't keep mutual consistency, the following problems will be happened. 1. The Bullwhip Effect will be occurred in knowledge message flow chain operation. 2. Replicas consuming application can't make sure the newest version copies. 3. It is easy for GTHEO knowledge base to occur horizontal information lacks harmony. 4. The OKMS can't control its replicas updating flow, transaction, usage mining results and it can't support different priority and value added updating services. 5. The dependent replicas usage mining results can't be sent to OKMS to make adaptive replicas updating route policies and value added updating services.

3 Concept model description

3.1 Concept model of OKMS consistency

In this paper, a novel framework was proposed which consists of active, real time, automation, and global routing knowledge replica from OKMS domain to GTHEO set. See in Figure 1.

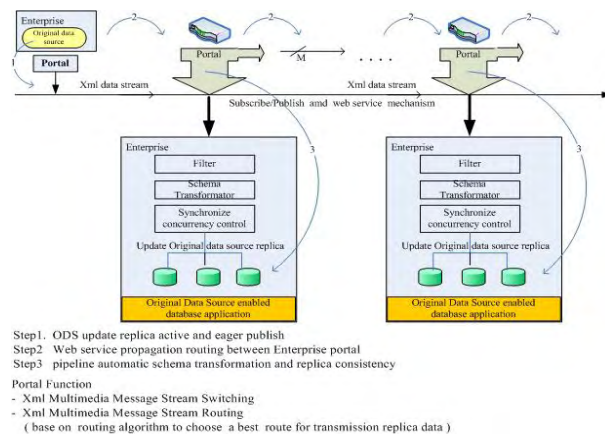


Figure 1. DHWS RP concept model

Propagation when OKMS object occurred add/insert events, according to subscribe lists. New copies will push to all global GTHEO portal which contained dependency replicas. The EP server then make one of the choice about discard/rerouting to

lower hierarchy portal, automatic pipeline processing and schema transformation. All wide area OKMS dependent replicas consistency and correctness can be automatic maintenance. Enterprise intranet application system need not change any source code. Infrastructure Model was shown in Figure 2.

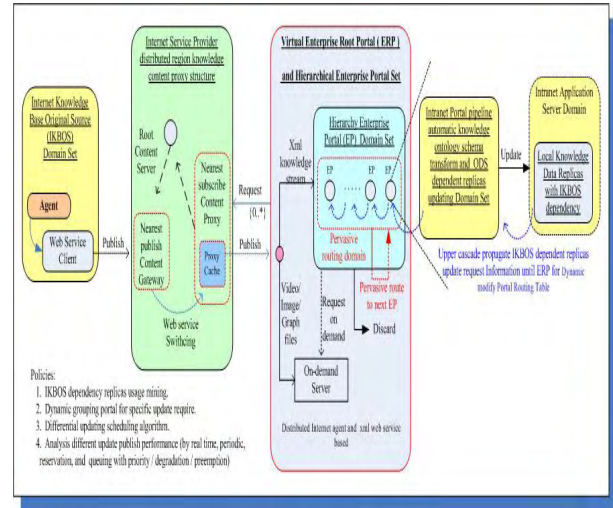


Figure 2. Infrastructure Model

3.2 OKMS Replicas Operation Model

Intranet replicas operation model have three kinds of application. In Figure 3. replicas with partial specific remote OKMS dependency relationships were denoted by (a). Replicas with OKMS role in extranet enterprise flow chain was denoted by (b). Replicas with partial dependency with a specific self-enterprise intranet OKMS was denoted by (c). Each EP can be two roles : FP (Forwarding Portal) and FPC (Forwarding Portal Cluster).

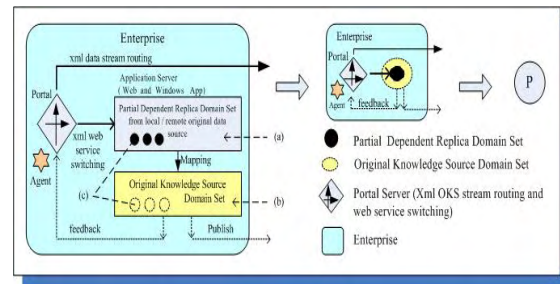


Figure 3. Enterprise Portal Model

3.3 the OKMS publish mechanism

In this section, which consists of OKMS subscribe and ripple propagation message chain shown in Figure

4. and Figure 5. When OKMS object was triggered by update event, the message chain was propagation in active, real time, and automation. Then all replicas with the specific OKMS dependent relationship could be globally updated.

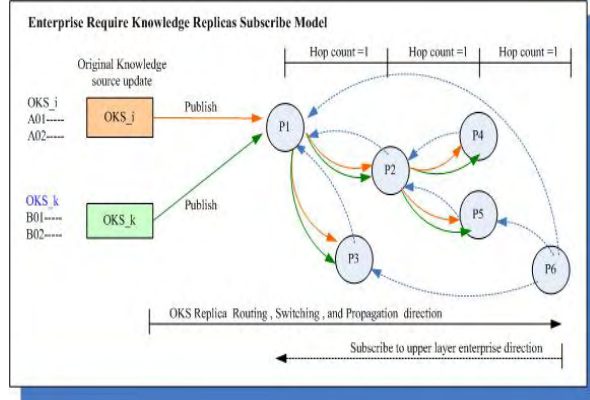


Figure 4. OKMS ripple propagation chain

3.3 Portal agent pipeline conversion knowledge ontology messages

The EP server then make one of the choices about discard/ reroute to lower hierarchy portal/automatic pipeline processing and schema transformation. See in Figure 5.

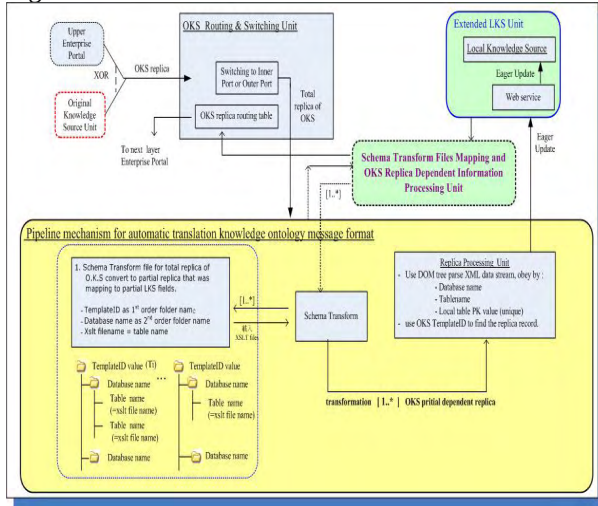


Figure 5. pipeline conversion OKMS

3.4 System Sequential diagram

In Figure 6 describe the UML sequential diagram about the concept model of system object.

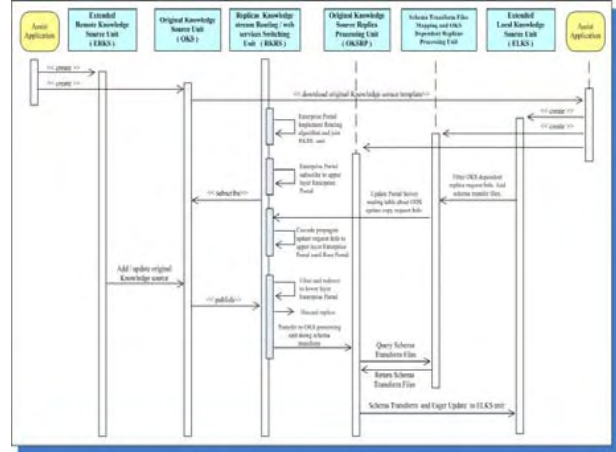


Figure 6. system object sequential diagram

4 Ripple propagation OKMS routing algorithm

Routing algorithm was built into each EP for assisting the xml data stream route path choosing.

4.1 Basic principle

The routing table fields description include: 1.OKMS dependency replica updates request information always maintained by EP 2.Information Included UUID value, PK value, Hop count, web service client etc. need to log in portal routing table from lower hierarchy enterprise request issues. 3. Portal's replica update routing information can come from intranet and extranet enterprise. 4. When a OKMS update replica publish to Portal, it will discard, send to processing unit, or reroute to lower portal depend on routing table check results. 5. For each route request from lower hierarchy portal will add a HC value to routing table's Hop Count value field.

Table 1. Routing table field's definition

Routing Table (RT) fields	Description
action state flag (ASF)	denote when add/delete OKMS dependent replicas in enterprise LKS
OKMS template ID (UUID)	OKMS template identify defined by OKMS site
OKMS replica Primary key value(PK)	instance of specific OKMS template
local connect / remote connect (LC / RC)	dependency replica update request issued by local enterprise or lower level enterprise portal
Destination Portal ID (DPID)	original enterprise Identify that issues the update request
Hop Count value (HC)	the distance between the enterprise contained the newest routing information and request enterprise
WSC_port	call next stop portal web service

4.2 Routing algorithms

Apply the rules to each route request. The ASF(Action State Flag) is sent from lower layer EP by portal agent and web services.

```

If (ASF==add) {
  //ie: rule:2-1 : ASF == add
  If (route request info.(UUID - PK - DPTD) sent from lower portal does not exist){
    //rule:2-1-1
    add this information into RT and upper cascade propagate this add event registry
  }
  Else If (route request info (WSC_port) identical){
    //rule:2-1-2-a
    replace with the new route request info. and cascade propagate to upper portal when TTL trigger.
  }
  Else If (route request info (WSC_port) not identical){
    //rule:2-1-2-b
    If (new HC value < RT's HC){
      //rule:2-1-2-b-1
      replace with the new route request info. and cascade call back to lower hierarchy portal to disable the route with the same DPID when its WSC_port <> null
    }
    Else {
      //rule:2-1-2-b-2
      Local portal do nothing then cascade call back to lower hierarchy portal to disable the route with the same DPID when its WSC_port <> null
    }
  }
}
Else {
  //ie: rule:2-2 : ASF == delete
  If (route request info does not exist in RT){
    //rule:2-2-1
    Local portal do nothing
  }
  Else {
    //rule:2-2-2
    Find and delete the record in RT, and upper cascade propagate delete event registry
  }
}
Recursive

```

Figure 7. DHWS routing algorithms

The following; Figure 8; is an demonstration for using above algorithm to adjust the P1 portal routing table

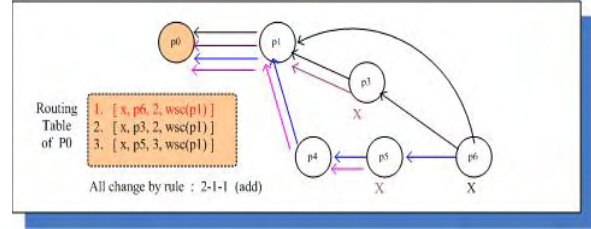
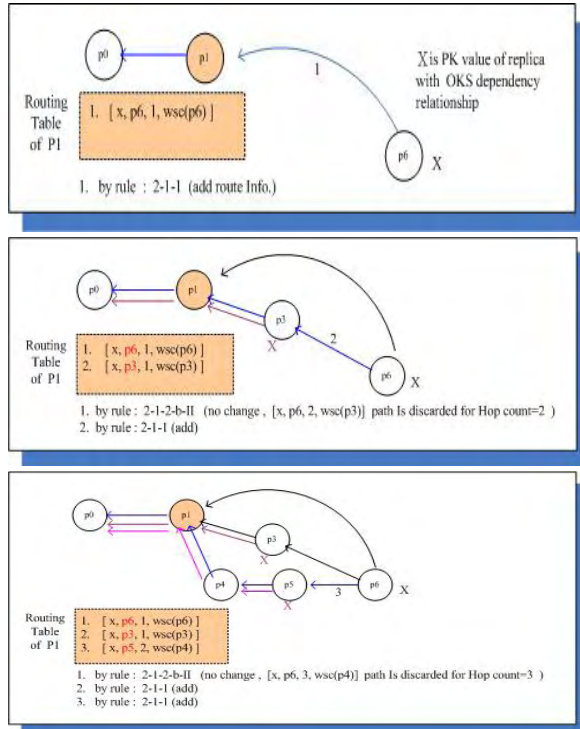


Figure 8. demonstration for OKMS routing algorithm

5 DHWSRP model

5.1 Motivation

This section aims at deriving a mathematical model for the DHWSRP forwarding plane to enable performance evaluation and optimal solution of the distributed routing data path. The analysis and modeling in fact leads to the well known shortest path optimization problem that has been extensively studied in the literature. Consequently, the paper will limit the analytical work to the derivation of the model and the mathematical expressions to minimize. The choice of an optimization criterion or performance metric determines the analytical model and the techniques that will be selected for optimization.

For simplicity, the paper focuses only on web service loss performance since adding an end-to-end constraint does not change the model. The objective is to minimize web service loss rate by selecting the appropriate set of FPC to form the routing data path. As described in Sect. 3.3 the distributed EP topology is composed of multiple interconnected EPs located in different Forwarding Portal Clusters. To enable load sharing over the DHWSRP, the same EP function can exist in several Forwarding Portal Clusters. For a given EP, the next EP (or neighbor EP) may be located in the same Forwarding Cluster or in a different Forwarding Cluster. The distributed data path should take the shortest path, to minimize web service loss from the ingress to the egress EP along the logical distributed EP topology. It is more logical to forward web services from a given EP_i to a next EP_j located in the same Forwarding Cluster rather than send them to the EP_j (with the same functionality) located in a different Forwarding Cluster. This is due to the web service loss that affects traffic performance between clusters (see Sect. 5.4, performance evaluation). Moving update replicas from an EP to other EPs located in other clusters should be minimized. The ERP (Enterprise Root Portal), root of EP topology, is responsible for calculating and creating the DHWSRP data path by

configuring the EP web service forwarding tables involved in the data path web service processing. The objective for the ERP is to determine the most efficient path with respect to a given constraint (web service loss rate) to satisfy the requirements of knowledge traffic. This is known as the constraint-based path selection problem.

The ERP (Enterprise Root Portal) can get optimize web service routing table according by receiving dependency replica updating request from low layer EPs. Then ERP configure each EP web service forwarding table based on a shortest path algorithm to create the appropriate distributed data path.

5.2 Problem formulation

For a given data path, a set of EPs are needed to perform web service processing functions from an ingress EP to an egress EP, and these EPs can be physically distributed and can be duplicated. Hence, several paths are possible between ingress and egress EP nodes since multiple duplicated EPs can perform the distributed data path.

For a given datapath, DP is used to denote a vector of EPj involved in the datapath formation:

$$DP = (EP_j) \text{ for } 1 \leq j \leq F. \quad (1)$$

which is also equivalent to a vector of sets A involved in the data path formation:

$$DP = (A_j) \text{ for } 1 \leq j \leq F. \quad (2)$$

Figure 9 shows an example of a given FPC data path DP in enterprise intranet and extranet environment.

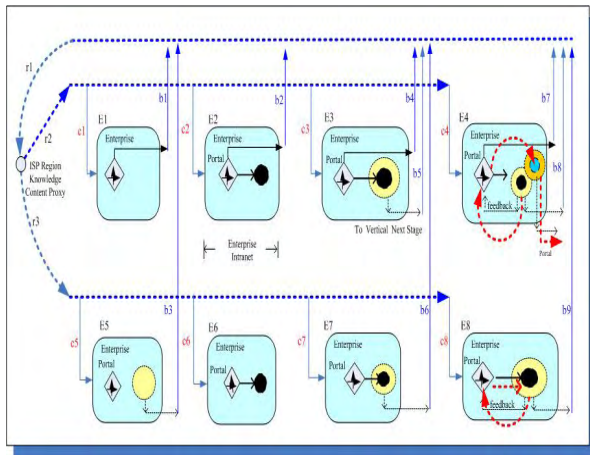


Figure 9. A graph model of sets of Forwarding Portal Clusters (FPCs)

Symbol description : Enterprise is denoted by E. Routing path is denoted by r. flow path is denoted by c

and b. EP encountered OKMS dependency replicas flow will generate three kinds of message flow path , as follows.

Apply in enterprise extranet data path, like formula (3).

$$E5 \rightarrow b3 \rightarrow r1 \rightarrow r3 \rightarrow \{c7|c8\} \rightarrow \{E7|E8\} \rightarrow \{b6|b9\} \rightarrow r1 \rightarrow r3 \quad (3)$$

For redirect updating replica to other hierarchy enterprise, flow path like formula (4).

$$E5 \rightarrow b3 \rightarrow r1 \rightarrow r2 \rightarrow \{c1|c2|c3|c4\} \rightarrow \{E1|E2|E3|E4\} \rightarrow \{b1|b2|b4|b7\} \rightarrow r1 \rightarrow r2 \quad (4)$$

For synchronize trigger multiple EP horizontal replicas consistency, the feedback intranet OKMS updating copy must be built in, like formula (5)

$$E8 \rightarrow b9 \rightarrow r1 \rightarrow \{\{r2 \rightarrow \text{same above (a)}\} | \{r3 \rightarrow \text{same above (b)}\}\} \quad (5)$$

6 Conclusion and future work

The thesis proposed mainly contribution as follows.

1. The OKMS dependency replicas usage mining rate is a function of dependency replica operation response time and turnaround time.
2. The updating route policy scheduling rate is a function of usage mining and reservation priority. The production rate of activity represented as a time variable.
3. Different value added update service schema was also analyzed. With these resolution models, optimization of the correct copy inputs and switching time of each activity in a recursive routing are calculated to obtain the minimum total cost of the project. The result can expand wide area enterprise replicas contained OKMS dependency relationships with real time wide area knowledge consistency correctness and maintenance abilities for GSCM.

References

- [1] Wearable Technology Special Issue of the IEEE Engineering in Medicine and Biology Magazine 2003., 22 Digi-Walker step counter [http://www.digiwalker.com]
- [2] Raskovic D, Martin T, Jovanov E: Medical Monitoring Applications for Wearable Computing. The Computer Journal 2004, 47(4):495-504.
- [3] Yung Bok Kim and Soon Woo Lee, "Performance Evaluation of Mobile Agents for Knowledge-Based Web Information Services KES-AMSTA", Lecture Notes in Computer Science, Vol. 4496, pp. 209-218, Springer, 2007.

- [4] Ruey-Kei Chiu and Kuo-Chin Tsai and Chi-Ming Chang and S. C. Lenny Koh and Kuan-Chih Lin, "The implementation of an agile information delivery system in building service-oriented e-healthcare network", *Int. J. of Enterprise Network Management*, Vol. 1, pp. 283-298, Inderscience Publishers, March 14 2007.
- [5] Istepanian RSH, Jovanov E, Zhang YT: Guest Editorial Introduction to the Special Section on M-Health: Beyond Seamless Mobility and Global Wireless Health-Care Connectivity. *IEEE Transactions on Information Technology in Biomedicine* 2004, 8(4):405-414.
- [6] Ákos Hajnal and David Isern : Knowledge Driven Architecture for Home Care CEEMAS, *Lecture Notes in Computer Science*, Vol. 4696, pp. 173-182, Springer, 2007.
- [7] Aurora Vizcaíno and Juan Pablo Soto and Javier Portillo-Rodríguez and Mario Piattini, "A Multi-agent Model to Develop Knowledge Management Systems", *HICSS*, p. 203, IEEE Computer Society, 2007.
- [8] Mahmoud Brahimi and Mahmoud Boufaïda and Lionel Seinturier, "Integrating Web Services within Cooperative Multi Agent Architecture", *AICT/ICIW*, p. 197, IEEE Computer Society, 2006.
- [9] Nicolas Singer and Jean-Marie Pecatte and Sylvie Trouilhet, "Combining Web Services and Multi-Agent Technology to Increase Web Cooperative Capacities", *International Conference on Internet and Web Applications and Services (ICIW'07)*, Mauritius, 13/05/07-19/05/07, p. (electronic medium), IEEE, Mai 2007.
- [10] Oh Byung Kwon, "Multi-agent system approach to context-aware coordinated web services under general market mechanism" , *Decision Support Systems*, 41(2), pp. 380-399, 2006.
- [11] Anna Maria Jankowska and Karl Kurbel and Denny Schreber, "An architecture for agent-based mobile Supply Chain Event Management *Int. J. of Mobile Communications*", Vol. 5, pp. 243-258, Inderscience Publishers, February 11 2007.
- [12] K. Walsh, A. Vahdat, and J. Yang. Enabling Wide-Area Replication of Database Services with Continuous Consistency.
- [13] M. Patiño-Martinez, R. Jimenez-Peris, B. Kemme, G. Alonso. "Consistent Database Replication at the Middleware Level". *ACM Transactions on Computer Systems (TOCS)*. In Press. 2003

Multi-Leak Diagnosis in Pipelines – A Comparison of Approaches

C. Verde
Instituto de Ingeniería
UNAM
Coyoacán DF, México
verde@servidor.unam.mx

R. Morales-Menendez, L.E. Garza,
Research and Graduate Program
Tecnológico de Monterrey
Monterrey NL, México
{rmm, legarza}@itesm.mx

A. Vargas, P. Velasquez-Roug, C. Rea, C.T. Aparicio, J.O. De la Fuente
MSc Students
Tecnológico de Monterrey
Monterrey NL México
{A00777924,A00362720,A00797489,A00952040,A00608453}@itesm.mx

Abstract

Leaks on pipelines can cause strong economic losses and environmental problems if these are not detected on time. The problem of detecting leaks is even more complicated when the pipelines are too large, difficult to reach by maintenance personnel, and equipped with minimum instrumentation. A comparison of four fault diagnosis approaches based on Output Observers, Artificial Neural Networks, Particle Filtering and Principal Components Analysis are presented. Simulated results of multi-leaks in pipelines showed that Particle Filtering techniques outperform the other approaches. However, a combined solution is proposed.

1 Introduction

The leak diagnosis is a challenge for the engineering sciences, since the economic losses and environmental damage produced by a deteriorated network are very significant [23]. Thus, different point of views, tools and backgrounds have been explored to implement monitoring systems for aqueducts, oil lines, networks of pipelines etc. As Wang pointed out [31]: *no single method can always meet all the requirements and each technique has its advantages and disadvantages in different circumstances*. An overview of diverse diagnosis tools for pipelines considering hardware and software tools is presented in [31], [18] and [21]. In the case of distribution networks, genetic algorithms [27] and wavelet analysis [1] have been used to detect leaks position.

The community of control has developed a general framework for the fault diagnosis of dynamic systems which allows the generation of fault symptoms, called residuals, by software [25] and [17]. Diverse mathematical model-based procedures have been developed in particular for leak diagnosis. Billman [2], Shields [26], Korbicz [18] and [13] designed residual generators using a finite dimension model and assuming fix space discretization in the set of partial differential equations which describes the fluid behavior. However, these methods can be only applied to detect leaks in limited cases. Assuming sequential leaks and scanning the pipeline with an adaptive law to estimate the leak position is a novel approach given in [29]. This procedure is successful if a new leak appears after the previous one has been located. However if a new leak occurs before the previous one is located the adaptive algorithm fails.

It is important to note that the location methods based on steady state conditions without test signals (for instance [9]) have a common property: if they are applied for the multiple leaks case, all of them delivered wrong location of the faults.

In [19] a proposal is made to capture the leaks patterns with the frequency response diagram considering a valve perturbation in the line. Ferrante [12] presented a method based on transient response using the Fourier Transform of pressure signals. However, all these methods have practical limitations if dominant nonlinear effects are presented in the fluid and a test signal is required. Brunone [3] reported a method based on unsteady state tests with the capacity to detect two leaks, assuming one leak is presented after the other.

Recently, da Silva [7] published a method based on a fuzzy classification of the transient response with multiple leaks; however the amount of historical data needed for training could be a limitation.

Then, from the technical point of view, innovative supervision systems for a pipeline must improve both hardware and software aspects. More accuracy and rapid sensors together with new algorithms assuming rather fault conditions must be developed. These issues must be complemented with a comparison of different methods for leak diagnosis which evaluate the diverse approaches under similar conditions. This task motivated this research project in which generic procedures were selected for leak diagnosis and tested them with the same leak model under similar conditions. The four procedures are: Artificial Neural Networks (ANN), Output Observers (OO) [6], Particle Filtering (PF) [22] and Principal Component Analysis (PCA)[14].

The main contribution of this paper is the performance analysis of the selected diagnosis methods under similar leaks conditions for a simulated hydraulic system. The presentation is organized as follows. In Section 2 a lumped parameter model of the pressurized water pipeline of the UNAM with flow sensors and pressure measurements taken at the extremes of the pipeline is introduced. Section 3, describes briefly each method. Section 4 presents the results for each approach and the discussions; and finally the conclusions are given in Section 5.

2 Discretized Model and Problem Formulation

The partial differential equations of movement and continuity of the fluid in a pipeline with boundary conditions are approximated in general by a finite dimensional model and implies a compromise between transient accuracy and computational complexity [8]. For the leaks detection issue, each leak generates a new boundary condition which must be considered in the discretization process to get an adequate model. Then the accuracy of any leak detector is a function of the discretization.

Assuming a pipeline of length L where for simplicity the space z is divided into $n = 4$ uniform cells of unknown size $\Delta z = \frac{L}{4}$; this discretization interval do not changes strongly the medium frequency components of transient pressure. The leaks are located at points $\{p_j = j\Delta z\}_{j=1}^{n=3}$. Each leak outflow is characterized by its parameter λ_j . A lumped parameters approximated model of the fluid can be written as 4 pairs of coupled nonlinear dynamic equations

given by the following set [5].

$$\dot{Q}_i = \frac{a_1}{\Delta z} (H_i - H_{i+1}) - \mu Q_i |Q_i| \quad i = 1, \dots, n \quad (1)$$

$$\dot{H}_i = \frac{a_2}{\Delta z} (Q_{i-1} - Q_i - u_{ti}(\lambda_i \sqrt{H_i})) \quad i = 2, \dots, n+1 \quad (2)$$

where the subscript i is associated to the variable of section i , with constants $a_1 = gA$, $a_2 = \frac{c^2}{gA}$ and $u_{ti} = u(t - t_i)$ the unit step function associated with the occurrence time t_i of the leak i . In this set, $H_1 = H(t, 0)$ and $H_5 = H(t, L)$ are the system inputs and the independent variable t has been removed to simplify the notation [30]. Figure 1 describes the variables positions of this model.

It is important to note that the integration time for the digital simulation of this model could be interpolated to satisfy Courant condition.

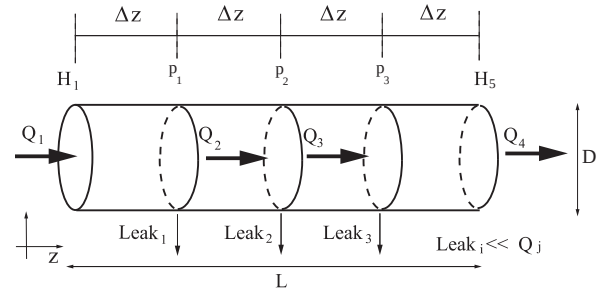


Figure 1. Graphical representation of the mathematical model variables.

The iron galvanized pilot pipeline is 132.56 m long with a diameter of 0.105 m, thickness of 4.7 mm, friction coefficient $f = 0.04$, pressure wave speed $c = 1284 \text{ m/s}$ and gravity acceleration $g = 9.81 \text{ m/s}^2$. The pipeline is integrated with a store tank of 7.4 m^3 , a hydraulic pump of 5 HP of variable speed and a valve at the end of the line. Experiments with operation point variations are carried out by the pump controlled by a pressure valve. Four valves are installed to emulate the leaks. The line is instrumented with two types of flow sensors; (1) Panametrics two-channel transport sensor based on ultrasonic pulses with an accuracy of $\pm 0.05 \text{ m/s}$, and poor bandwidth, 5 s of actualization rate, and (2) Signet 2540 paddle wheel flow sensors with an actualization rate of 0.1 s. The ultrasonic sensors are used in the experiments only to measure steady state flows. The pressure measurements are taken with a WIKA piezo-resistive and thin film pressure transmitter. The measured signals are sampled at points using a data acquisition system of National Instruments with a sampling frequency below 100 Hz. Then, in general the leak detection issue can be formulated as follows.

Problem. Assuming that it is known the evolution time of the set of variables $K = \{H_1, H_5, Q_1, Q_4\}$ starting from the alarm generation, the isolation issue of a leak in a pipeline consists in the identification of the set of unknown parameters $\Theta = \{p_i, \lambda_i, t_i\}$ which satisfies simultaneously the equations (1) and (2) with $i = \{1, 2, 3\}$ and $\lambda_i > 0$. As it has been mentioned in the introduction diverse methods can be used to solve this problem.

3 Methods

3.1 Artificial Neural Network

ANN present several attractive properties such as universal function approximation capabilities, insensible to noisy or missing data, and accommodation of multiple non-linear variables for unknown interactions. A feed-forward architecture was exploited considering 4 input neurons ($Q_1(t)$, $Q_1(t-1)$, $Q_4(t)$, $Q_4(t-1)$) for both inlet and outlet flows. The hidden layer has 4 neurons. On the output layer, there are 3 (on/off) neurons for each possible leak. The model parameters were learned with the back-propagation algorithm based on data generated from a non-linear model of the pipeline.

A physical model of the process is not needed, *ANN* extracts this knowledge from experimental data. However, *ANN* demands great amount of data for the training step (of each leak condition). In this case, 15,000 data were used during the training step.

3.2 Output Observers

A bank of residual generators for each leak were designed based on the unknown inputs observer approach [24] and on the discretized model. For robustness issues, each residual is sensible to all leaks except for a specific one [28]. A Luenberger observer was designed, where the residual information is generated from the difference of the real and estimated system output. A filter was needed for noisy signals.

Each residual generator has three possible states, the transition among them occurs when the steady state of the residual signal changes. Because the size of the leak is unknown, there is not threshold for the aforementioned state transitions. Left plot in Fig. 2 shows an example of this state transition. Fig. 3 shows the performance of the leak detection system with three residuals. Top plot shows the true state condition. Second plot shows the behavior of the first residual, which shows 2 states changes (t_1, t_2). Third plot corresponds to the second residual, which shows 1 state change (t_1). Last plot represents the third residual, which also shows 1 state change (t_2).

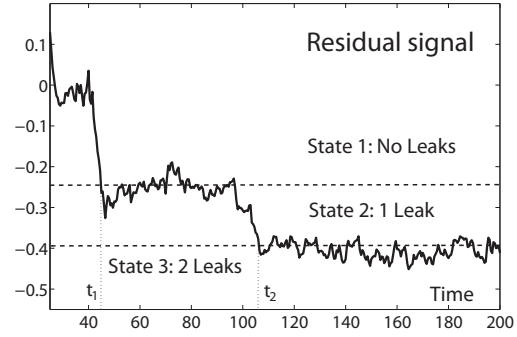


Figure 2. Residual behavior. This plot shows a residual signal where several states are changing: 1-state (0,t1), 2-state (t1,t2), and 3-state (t2,200).

3.3 Particle Filtering

Particle Filtering (*PF*) has attracted more attention in the non-linear non-Gaussian state estimation field; however, it has shown successful results in fault diagnosis. *PF* approximates an optimal filtering using the probability density function with a set of samples (*particles*) and maintains the distribution updated as new observations are made over time.

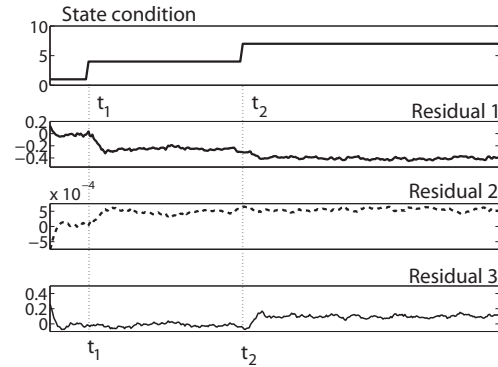


Figure 3. Residual behavior. This plot describes the state condition behavior based on the three residuals: no leaks (0,t1), 1 leak in section 3 (t1, t2), 2 leaks in section 2 and 3 (t2, 200)

The identification of pipeline leaks given the input/output flow is an inference task that demands to com-

pute the distribution $p(s_{0:t}|y_{1:t})$, where s_t represents the states of the leaks, and y_t represents the observed flows over time. This probability distribution is derivable from the full posterior probability distribution $p(x_{0:t}, s_{0:t}|y_{1:t})$ (x_t are hidden process states) using standard marginalization. This cannot be done analytically, thus numerical approximation methods such as *PF* techniques must be use [10].

While *PF* is applicable, [11] showed that greater efficiency can be achieved by considering that the *Kalman* filter part of the model is analytically solvable. The factorization $p(x_{0:t}, s_{0:t}|y_{1:t}) = p(x_{0:t}|y_{1:t}, s_{0:t}) p(s_{0:t}|y_{1:t})$ based on the Rao-Blackwell (RB) equation allows to visualize the inference task in two steps. The density $p(x_{0:t}|y_{1:t}, s_{0:t})$ is Gaussian and it can be computed analytically if the marginal posterior density $p(s_{0:t}|y_{1:t})$ is known. Basically, *PF* estimates the distribution of s_t and the analytical computation for the mean and variance of s_t . This result in the *RBPF* algorithm [11].

However, [22] proposes a further improvement over the *RBPF* algorithm: *look-ahead RBPF*. While evaluating the importance weights for particles at time t , *look-ahead RBPF* looks ahead one step to see which way the measurements are going at time $t + 1$. It then uses this information to compute better weights at time t . The basic sequential steps are *Kalman prediction*, *selection*, *sequential importance sampling*, and *Kalman updating*.

For *look-ahead RBPF* is not necessary to have a priori knowledge of the physical model. A state space representation of the system (for each leak condition) based on experimental data is enough. The *PF* is an algorithm that consumes a considerable computing time.

There are similar research projects in this field. [20] proposes a state augmentation technique and an artificial dynamics of the leak parameters are introduced in the model in order to track the leak with good results in gas pipelines.

3.4 Principal Component Analysis

The main idea is to use a physical model of the pipeline with no leaks to compute the *PCA* model [15]: $\mathcal{P}\mathcal{X} = \mathcal{Y}$, where $\mathcal{P}$ is the principal components matrix, $\mathcal{X}$ is the original data, and $\mathcal{Y}$ is the orthogonal (uncorrelated) data set. The $\mathcal{X}$ matrix considers the inlet/outlet flow and head pressure with 4 regressive values of each one.

Dynamic *PCA* (*DPCA*) cope with dynamic data for online leak detection, just adding to the observations in $\mathcal{X}$ the previous values of the variables. [16] proposed a residual generation and a control limit (*CL*) that indicates when there is a leak. *DPCA* is not able to find the leak position; the leak position can be detected (and its magnitude) by a combined approach proposed by [4].

4 Simulation Results

Consider the water pilot pipeline described in section 2 with $a_1 = 0.0849 \text{ m}^3/\text{s}^2$ and $\mu = 21.99 \text{ m}^{-3}$ and assume a leak appearing at $t = 0$ which is characterized by $\lambda_i = 6.2283 \times 10^{-5}$ located at $p_i = i\Delta z$ distance.

A Markov chain was used in order to mimic 8 possible states. Every state is related to the presence of a leak in sections 1, 2 and 3. Because the stochastic nature of some algorithms, 30 data sets were generated for testing the different diagnosis approaches. The first 10 testing data sets are noise free, the following 20 data sets have a white noise with variances of $\sigma_1 = 0.0001$ and $\sigma_2 = 0.00015$ respectively. The simulation time was 500 s with sampling time of 0.5 s (1,000 data points). Figure 4 shows an example of the 30 testing data sets. The problem is stated as follows: given only the input and output flows detect any leak and its location.

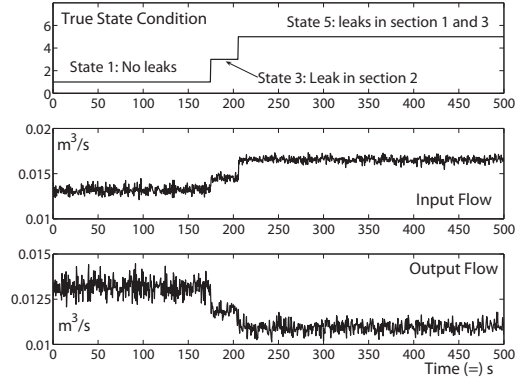


Figure 4. Transient response of the pipeline for different state condition. Top plot shows the different state condition over time. Middle plot presents the inlet flow over time, while bottom plot shows outlet flow.

Four approaches were tested with 30 testing data sets. The results are indicated as the average of the diagnosis error, which is computed as the time-steps having state condition wrongly identified divided by the total number of time-steps.

To illustrate the variability of the diagnosis error in some approaches, the results are shown as a box and whisker plot Fig. 5. The boxes have lines at the lower quartile, median, and upper quartile values. The whiskers are lines extending from each end of a box to show the extent of the rest of the data. The boxes are notched. Notches represent a robust estimate of the uncertainty about the medians for box-to-

box comparison. Outliers are data values beyond the ends of the whiskers; outliers are shown as circles.

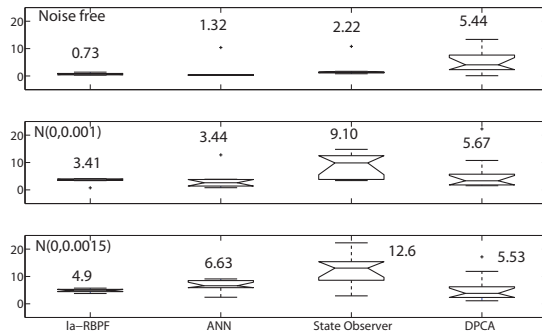


Figure 5. Leak detection error. Plots show error for different approaches; top plots correspond to noise free condition; while middle and bottom plots exhibit noise.

4.1 Discussion

PF and *ANN* showed the better results of all the tested conditions (noise free and noisy signal), although the variance increases according the level of noise. These methods do not need a physical model of the system, and can learn the model from data. The learning step in both cases is very time consuming because every leak condition should be considered during this training step. Sometimes, it is almost impossible to reproduce every feasible leak condition, especially when more segments are required to make a more precise diagnosis. Given this experimental knowledge, *PF* and *ANN* can easily detect and diagnose the most likely leak condition or the most similar pattern (input-output relation).

If the actual pipeline leak was no considered into the experimental knowledge, both approaches will detect/diagnose the most similar pattern based on the observations.

The Output Observer (*OO*) approach gave the third better results. However, it has important limitations, because *OO* demands a good knowledge of the physical system. In fig. 5 the average and variance of the diagnosis error are relatively high, especially in noisy environments.

PCA approach was discarded; because it only detect the leak presence, not the leak position.

5 Conclusions

This paper presented a comparison of four approaches: Particle Filtering (*PF*), Artificial Neural Networks (*ANN*),

Output Observer (*ANN*) and Principles Component Analysis (*PCA*).

Results showed that *PF* approach had the best results in terms of error diagnosis (average and variance); however, the training step in this algorithm is very time consuming.

Also, for all approaches noise undermines their precision. Even though there are many research projects, there is no unique solution for this problem. Based on the results shown in Fig. 5 a combined proposal is considered. This proposal considers the minimum instrumentation in the system.

- **Detection:** *PF* algorithm can be exploited in order to detect online the presence of a leak or leaks. This algorithm can manage nonlinear system. It is very efficient, fast and precise given the experimental knowledge of the system. Experimental knowledge
- **Location/Size:** A non-linear observer based on the mathematical model can find the leak (position and size).

Acknowledgments

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References

- [1] I. Ai-Shidhani, S. Beck, and W. Staszewski, "Leak Monitoring in Pipelines Networks using Wavelet Analysis ," *Key Engineering Material*, vol. 245-246, pp. 51–28, 2003.
- [2] L. Billmann and R. Isermann, "Automatica," *Leak Detection Methods for Pipelines*, vol. 23, no. 3, pp. 381–385, 1987.
- [3] B. Brunone, "Transient Test-based Technique for Leak Detection in Outfall Pipes," *J. of Water Resources Planning and Management*, vol. 125, no. 5, pp. 302–206, 1999.
- [4] R. Carrera and C. Verde, "Localizador Automático de Fugas en un Ducto," *Ingeniería Hidráulica en México*, vol. XVI, no. 2, pp. 139–151, 2001.
- [5] H. Chaudry, *Applied Hydraulic Transients*. New York, USA: Vann Nostrand Reinhold Company, 1979.
- [6] J. Chen and R. J. Patton, *Robust Model-Based Fault Diagnosis for Dynamic Systems*. Boston, USA: Kluwer Academic Publishers, 1999.

- [7] H. da Silva, C. Morooka, I. Guilherme, T. da Fonseca, and J. Mendes, "Leak Detection in Petroleum Pipelines using Fuzzy System," *J. of Petroleum Science and Eng.*, no. 49, pp. 223–238, 2005.
- [8] A. B. de Almeida and E. Koelle, *Fluid Transients in Pipe Networks*. Boston, USA: Computational Mechanics Publications Co-publisher Applied Science, 1992.
- [9] J. Dinis, A. Wojtanowicz, and S. Scott, "Leak Detection in Liquid Subsea Flowlines with no Recorded Feed Rate," *J. of Energy Resources Technology*, vol. 121, pp. 161–166, 1999.
- [10] A. Doucet, N. de Freitas, and N. Gordon, Eds., *Sequential Monte Carlo Methods in Practice*. Springer-Verlag, 2001.
- [11] A. Doucet, N. de Freitas, K. Murphy, and S. Russell, "Rao-Blackwellised particle filtering for dynamic Bayesian networks," in *Uncertainty in Artificial Intelligence*, C. Boutilier and M. Godsztmidt, Eds. Morgan Kaufmann Publishers, 2000, pp. 176–183.
- [12] M. Ferrante and B. Brunone, "Pipe System Diagnosis and Leak Detection by Unsteady-State Tests: 1 Harmonic Analysis," *Advances in Water Resources*, vol. 26, pp. 95–105, 2003.
- [13] G. Geiger, W. Gregoritz, and D. Matko, "Leak Detection and Localisation in Pipes and Pipelines," in *European Symposium on Computer Aided Process Eng-10*, vol. 2, May 2000, pp. 781–786.
- [14] J. Gertler and J. Cao, "Design of Optimal Structured Residuals from Partial Principal Component Models for Fault Diagnosis in Linear Systems," *J of Process Control*, pp. 585–603, 2005.
- [15] M. Harkat, G. Mourot, and J. Ragot, "An Improved PCA Scheme for Sensor FDI: Application to an Air Quality Monitoring Network," *J. of Process Control*, vol. 16, pp. 625–634, 2006.
- [16] R. Isermann, *Fault-Diagnosis Systems. An Introduction from Fault Detection to Fault Tolerance*. Germany: Springer-Verlag, 2005.
- [17] —, *Fault Diagnosis System*. Germany: Springer-Verlag, 2006.
- [18] J. Korbicz, J. Koscielny, Z. Kowalczyk, and W. Cholewa, *Fault Diagnosis*. Germany: Springer-Verlag, 2004.
- [19] P. Lee, J. Vitkovsky, M. Lambert, A. Simpson, and J. Liggett, "Leak Location using the Pattern of the Frequency Response Diagram in Pipelines: a Numerical Study," *J. of Sound and Vibration*, vol. 284, pp. 1051–1075, 2005.
- [20] M. Liu, S. Zang, and D. Zhou, "Fast Leak Detection and Location of Gas Pipelines Based on Adaptive Particle Filter," *Int. J. of Applied Mathematics and Computer Science*, vol. 15, no. 4, pp. 541–550, 2005.
- [21] J. Makar and N. Chagnon, "Inspecting Systems," *American Water Works Association*, vol. 91, pp. 36–46, July 1999.
- [22] R. Morales-Menéndez, N. de Freitas, and D. Poole, "Real-Time Monitoring of Complex Industrial Processes with Particle Filters," in *Advances in NIPS 16*. Cambridge, MA: MIT Press, 2002.
- [23] W. Muhlbauer, *Pipeline Risk Management Manual*. Burlington: Gulf Professional Publishing, 2004.
- [24] P. Müller and M. Huo, "Design of Observers for Linear Systems with Unknown Inputs," *IEEE Trans on Automatic Control*, no. 37, pp. 871–875, 1992.
- [25] R. Patton, P. Frank, and R. Clark, *Issues of Fault Diagnosis for Dynamic Systems*. London: Springer, 2000.
- [26] D. Shields, S. Ashton, and S. Daley, "Design of Non-linear Observers for Detecting Faults in Hydraulic Sub-sea Pipelines," *Control Engineering Practice*, vol. 9, no. 3, pp. 297–311, 2001.
- [27] K. Tanand, B. Karney, M. Pendlebury, and F. Zhang, "Inverse Transient Calibration of Water Distribution Systems using Genetic Algorithms," in *Proc Conf on Computing and Control for the Water Industry*, Exeter, UK, 1999.
- [28] C. Verde, "Multi-Leak Detection and Isolation in Fluid Pipelines," *Control Eng. Practice*, vol. 9, pp. 673–682, 2001.
- [29] —, "Accommodation of Multi-Leak Location in a Pipeline," *Control Eng. Practice*, vol. 13, no. 8, pp. 1071–1078, 2005.
- [30] C. Verde, N. Visairo, and S. Gentil, "Two Leaks Isolation in a Pipeline by Transient Response," *Applied Water Resources*, vol. 30, pp. 1711–1721, 2007.
- [31] X. Wang, M. Lambert, A. Simpson, J. Liggett, and J. Vitkovsky, "Leak Detection in Pipelines using the Damping of Fluid Transients," *J of Hydraulic Eng*, vol. 128, no. 7, pp. 697–711, 2005.

Compression Strength Prediction of Mixtures Concrete with Scrap Tire with Neural Network Approach

J.L. Acevedo-Dávila, L. M. Torres-Treviño
Corporación Mexicana de Investigación en Materiales
: jacevedo@comimsa.com.mx, ltorres@comimsa.com.mx

Abstract

The compressive strength of mixtures made with scrap tire id presented. In this study, neural network modeling was applied for predicting compressive strength of mixtures containing variable size of tire scrap. This modeling allows avoiding a large number of trial mixtures tests and provides a new alternative for designing new constructive components at lower costs. Results shown that neural model showed an excellent performance and accurate and highly reproducible predictions.

1. Introduction

Every year, 25 million tires are used in Mexico. However, once their service period has been passed, elimination of these ones has become a very serious problem for the environment, and not only landfills or dumps, but backyards and even streets have become places to storage them. Tires accumulation represent a latent danger, on only because of potential environmental threat, but also for promoting reproduction of rats, bugs and different kinds of vermin, including dengue fever transmitting mosquitoes. Innovative solutions to face the challenges related to waste management have been widely developed. Among the most promising solutions, tire's rubber is being used in mixtures for asphalt concretes, vapor production from incinerated tires or fuel for cement production furnaces; recycling of milled tire rubber as a raw material for plastic and rubber products. Tires have also been used to create artificial maritime reefs and for concrete and mortars. About this last application, technical literature reports using tires chips as a substitute for some concrete aggregates [1, 2, 3] in order to improve mechanical properties, such as impact, compression and bending strengths. However, development of research in this areas requires a long periods of time and expensive costs. Because of this, small companies find difficult to develop and design

their own concrete mixtures. Neural networks modelation represents an economical way to carry it out, taking advantage of the predictive capabilities of these computational paradigms. There are some published works that report[4 ,5 ,6 , 7, 8] similar applications of neural networks, mostly used to predict some specific properties of mixtures, but the neural network of maximum sensibility (NNMS) employed in the present work has not been previously used for this kind of studies. Furthermore, the mixtures considered here have not been studied by other researchers [9].

Neural network of maximum sensibility (NNMS) is a simple emulation of the natural neural system commonly found in several advanced life forms (Fig. 1.a). The basic component of the artificial neural network (ANN) is a process unit (PU) that mimics the functions and characteristics of a single neuron (Fig. 1.b). The PU has inputs ("dendrites"), an output ("axon"), a nucleus ("soma") and also presents a certain level of connection with another process unit usually called weight ("synapse"). The interconnections between all these neurons form a network.

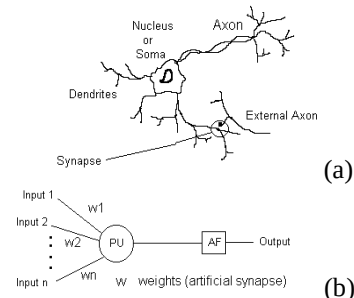


Figure 1. (a) The natural and (b) the artificial neuron.

The behavior of an ANN can be modified by changing the weights. The procedure that is followed to accomplish this is called "learning rule" [9]. The

application of the NNMS basically involves two procedures: Learning and Activation processes. The activation procedure is determined by the maximum sensibility. If a neuron is the most activated, and its signal value above a margin, then a maximum sensibility state is achieved and the neuron can give its corresponding output response or can learn new signals. The other case is when maximum sensibility is inactive, and it becomes necessary to approximate an output response with the contribution of active neurons. In the learning procedure it is essential to allot a possible new signal to a new neuron to learn. The activation procedure is described in Equation Set 1. SN represents the activation of every neuron and m represents the margin, while FA corresponds to an activation function (usually a Gaussian one). The $maxp$ function determines the position and value of the vector's maximum element. When it is necessary to learn a new signal, a specific neuron (n) must be chosen. There are two ways to select new neurons. The first one takes new neurons from predefined memory structures. When the memory structure is fixed, and there is not enough space, then the space occupied by the least used neuron is taken. Equations 1, 2 and 3 must be sequentially applied.

Activation equations in NNMS

$$SN_n = FA(\sqrt{(W_{n,i} - X_i)^2}) \quad (1)$$

$$(w, v) = \max p(SN) \quad (2)$$

$$O_j = \begin{cases} AC_{w,j} & v > m \\ (AC_{n,j} * SN_n) / SN_n & otherwise \end{cases} \quad (3)$$

Learning equations in NNMS

$$W_{w,i} = \begin{cases} (W_{w,i} + X_i) / 2 \\ X_i \end{cases} \quad (4)$$

$$AC_{w,j} = \begin{cases} (AC_{n,j} * SN_n) / SN_n & v > m \\ DO_j & otherwise \end{cases} \quad (5)$$

The learning procedure is described by equations (4) and (5). X_i is an input vector and DO is a desired output vector. The NNMS has been proved successfully to approximate some mathematical functions. Its performance is very similar to that of other neural networks like the one using gradient backpropagation or Levenberg - Marquardt learning rule. A further advantage of NNMS is that it can learn fast on line, and it can learn new information without an exhaustive training procedure.

2. Experiments

2.1 Raw Materials and specimen preparations

Materials used consisted of cement Portland type I, a coarse or fine limestone aggregate (with a particle size of 6.35 mm in the first case, and of 4.76 mm in the second case), and water. The properties of both aggregates are as shown in table 1. The particle size analysis of all used materials was carried out according to the Mexican NMX-C-077-ONNCCE standard. Two particle sizes of scrap tire were used (1.75 y 0.3 cm), as they can be seen in Figure 2. As for mix proportions the control specimen (no scrap tire) was set at 1:0.58:2.85:2.2 (cement:water:coarse aggregate: fine aggregate). In the case of mix proportions with scrap tire was set at 1:0.66:2.56:2.19:0.29 (cement:water:coarse aggregate: fine aggregate:scrap tire).

Table 1. Properties of coarse and fine aggregate

	Coarse	Fine
Properties	g/cm ³	
Bulk Density	1.453	1.651
Dry Density	1.613	1.715
Dry weight	2.666	2.617
Superficially saturated dry weight	2.674	2.660
% Absorption	0.30	0.26
Apparent weight	2.674	2.660
% Aggregate Humidity	0.05	0.10





Figure 2. Tire chips with 0.3 y 1.75 cm average length.

2.2 Compressive Strength

The compressive strength tests were conducted according to the ASTM C39 Standard. A Tinius Olsen Machine was used to carry out these tests. Compressive strength of the cylinders was measured after a curing period of 3, 7, 14 or 28 days. Three tests were carried out for each curing time, per each mixture composition.

2.3 Methodology employed for application of the NNMS

The methodology employed for the application of NNMS to study a specific problem involves two basic processes. The first one consists in developing a neural networks model and the second one is a subroutine connected to the neural model, which uses programmed data input sequences. This methodology has the following steps:

- 1) Acquisition of resulting experimental data from an appropriate experimental design.
- 2) Utilization of obtained data to train the neural network in order to generate a model. Figure 3
- 3) Validation of model using additional experimental results that were not previously used during the neural network training process.
- 4) Prediction of process performance applying the validated model and using an appropriate new sequence of inputs. Figure 4

It is noteworthy to emphasize that it is necessary to apply a defined sequence of inputs. The performance of this methodology depends on the quality of the model, which has to be validated; otherwise the model's quality will be poor, resulting in non reliable performance on the prediction stage.

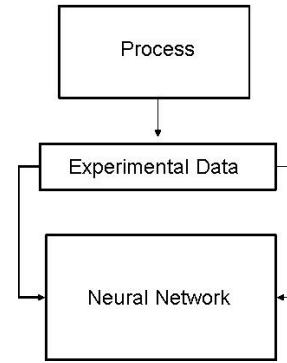


Figure 3. Schematic sequence that shows how experimental results are acquired and used to feed the corresponding neural network

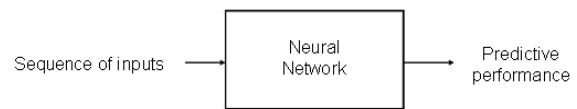


Figure 4. Structure of prediction procedure using a neural network

3. Results and discussion

3.1 Compressive Strength

Compressive strength values for concrete cylinders as a function of curing time and tire chips length are given in Table 2. It is observed that as longer the curing time, higher the strength for all evaluated conditions. However, when comparing compressive strength values as a function of tire chips length it is found that as the length increases compressive strength decreases.

Table 2. Experimental data to rubberized concrete mix.

Mix without scrap tire				
Curing time (Days)	3	7	14	28
Compressive Strength (Mpa)	11.8	14.4	18.4	22.2
	12.1	14.0	21.2	19.5
	11.9	14.2	19.8	22.8
Average	11.9	14.2	19.8	21.5
Mix with scrap tire (10 wt %)				
Average length 0.3 cm				

Curing time (Days)	3	7	14	28
Compressive Strength (Mpa)	5.53	6.86	7.44	8.65
	5.81	6.96	7.25	9.58
	4.41	6.76	8.38	8.32
Average	5.25	6.86	7.69	8.85
Average length 1.75 cm				
Curing time (Days)	3	7	14	28
Compressive Strength (Mpa)	5.06	5.99	6.53	7.23
	5.48	6.62	6.80	7.31
	5.28	5.74	6.78	7.24
Average	5.27	6.11	6.70	7.26

3.2 Neural network of maximum sensibility

The architectural layout employed to predict the compressive strength of concrete cylinders after certain curing times, (3, 7, 14 and 28 days) is shown in Figure 5. This architectural layout shows also that a NNMS with two inputs and one output was used. One input corresponded to average chips length and second input represents the concrete's water curing time. The output corresponds to calculated compressive strength values. Since the percentage of Portland cement and the water/cement ratio were kept constant for all mixtures studied, they were not considered as inputs. The complete raw data set (a total of 36 measurements corresponding to 12 experiments, each one of them replicated three times) was used in the training procedure. Then, the average value of each three replicas per condition was calculated. The complete set of average values was used to validate the neural network.

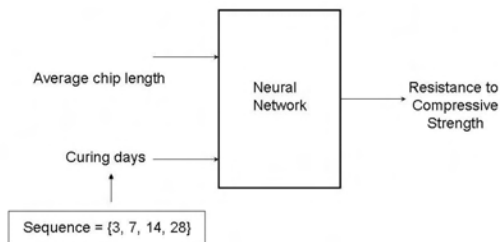


Figure 5. Architectural layout used to predict the compressive strength of rubberized concrete

The performances of the neural network, with training and validation data are shown in Figures 6 and 7.

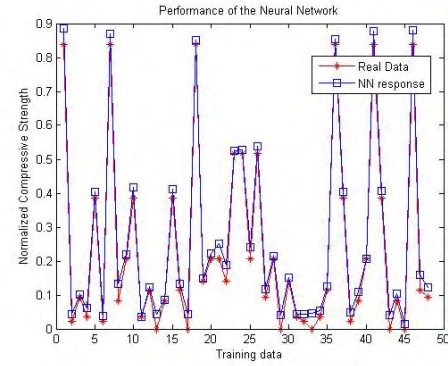


Figure 6. Performance of the neural network with training data

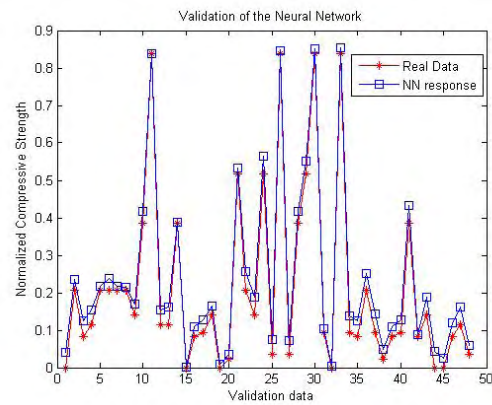


Figure 7. Performance of the neural network with random validation data

As it can be seen in both figures, a very good agreement between calculated and experimental values was found, which indicates that the model describes very well the compressive strength behavior the evaluated concrete cylinders.

Neural network's performance behavior, when compared to experimental results for tire scrap reinforced concrete showed a very good agreement, as it is shown in Figure 8. It is easily noticed how graphs follow similar trends.

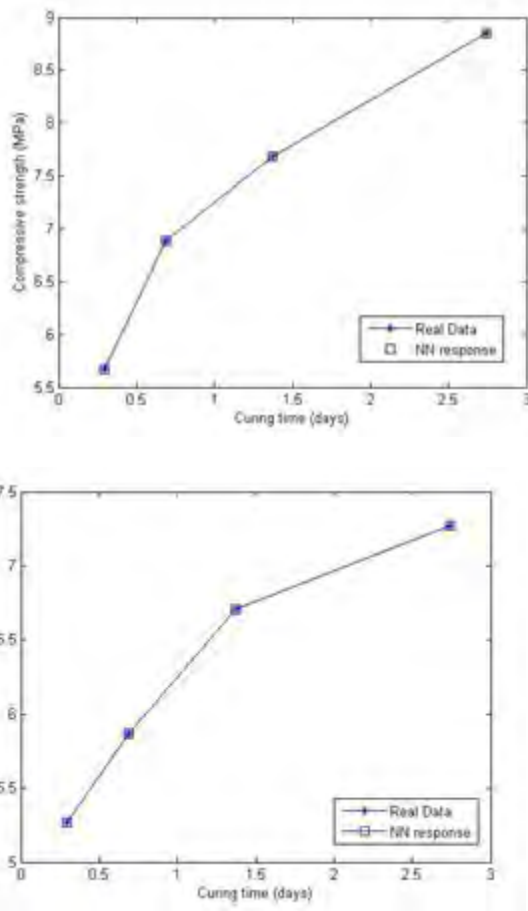


Figure 8. Neural Network prediction performance against real data: scrap tire 0.3 y 1.75 cm respectively

4. Conclusions

Neural network modeling resulted to be a very accurate tool to get highly reproducible predictions. An increased addition of tire chip resulted in a general decrease in the measured compressive strength of the cylinders, but it has to be considered that the main goal of this work was focused to the application of neural networks and not the improvement of mechanical properties. However, it is expected an increase in toughness as tire chips are been added, a behavior that could be useful in designing anti-seismic construction materials, but more research has to be done in this field.

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5 References

- [1] Sukontasukkul P, Chaikaew Ch, Properties of concrete pedestrian block mixed with crumb rubber. *Construction and Building Materials*, 2006; 20: 450–457.
- [2] Turatsinze A, Bonnet S, Granju JL. Potential of rubber aggregates to modify properties of cement based-mortars: Improvement in cracking shrinkage resistance. *Construction and Building Materials* 2007; 21: 176–181.
- [3] Li G, Garrick G, Eggers J, Abadie Ch, Stubblefield MA, Pang S. Waste tire fiber modified concrete, *Composites: Part B* 2004; 35: 305-31.
- [4] Acevedo J, Torres M, Gómez L. Compressive strength prediction of building blocks from lightweight raw materials: A neural network approach, In: *Proceeding of the Electronics, Robotics, and Automotive Mechanics Conference*, Mx, 2006, 1: 184-187.
- [5] Kim J, Kim DK, Feng MQ, Yazdani F, Application of Neural Network for Estimation of Concrete Strength, *Journal of Materials in Civil Engineering*, 2004; 16 (3): 257-264.
- [6] Kim DK, Lee JJ, Lee JH, Chang SK, Application of Probabilistic Neural Network for Prediction of Concrete Strength, *Journal of Materials in Civil Engineering*, 2005, 17 (3): 353-362.
- [7] Ince R, Prediction of fracture parameters of concrete by Artificial Neural Networks, *Engineering Fracture Mechanics*, 2004, 71: 2143–2159.
- [8] Bai J, Wild S, Ware JA, Sabir BB, Using neural networks to predict workability of concrete incorporating metakaolin and fly ash, *Advances in Engineering Software*, 2003; 34: 663–669.
- [9] L. Torres, Master D. Thesis, UASLP (1998).

Semi-automatic Codification of Economical Activities with Support Vector Machines

Fabiano Rogerio Correa
Escola Politecnica da Universidade de
Sao Paulo
fabiano.correa@poli.usp.br

Jun Okamoto Jr.
Escola Politécnica da Universidade
Sao Paulo
jokamoto@usp.br

Abstract

The correct attribution of codes to economical activities is of ultimate importance for fiscal and administrative purposes. In a joint effort involving federal, state and city business regulating offices, the unification and automation of this codification process is in progress. As the input information is in the form of free textual entries, techniques used in multi-class, multi-label text categorization are well suited. Due to the fact that the possible codes are in the order of one thousand, this problem is under investigation through an approach based on a hierarchical use of Support Vector Machines (SVM). The hierarchical organization of several SVMs split the problem into smaller ones and allows a fine tuning of the code attribution obeying the very hierarchy present in the table describing the codes. This paper will introduce the problem, describe the proposed approach based on SVMs and show some results that validate the approach.

1. Introduction

More than ten years ago, the Brazilian government started a project to standardize the codes used in economical activities categorization across the whole country. At the present time, the codification, for fiscal and administrative purposes, of the economical activities executed by a company or an individual is handled by the business owner himself or a representative. Experts only interfere in this process in the most difficult cases. As this classification is of ultimate importance for the Brazilian government

regulating offices, they started, in the middle of 2006, some projects to propose solutions to initiate the automation of the codification process.

Due to the lack of references on this specific subject, some guidance was obtained studying similar problems. Giorgetti and Sebastiani (2003) dealt with the problem of survey coding, where one set of pre-defined codes was attributed to an open-ended question. This is a difficult task, because the attributed code is based on a specialist knowledge and is a matter of subjective judgment. The authors formulated this problem as a text categorization problem, using data to learn a mapping between answers in free texts and code categories. Following this approach here, the problem of automatic codification is also formulated as a text categorization problem. Although there are many similarities between the automatic codification of economical activities and the survey coding, as both have as input a free textual entry, there are some differences. In the automatic codification case, a single document or entry receive in general more than one code, related to different economical activities carried out by the business owner. Here the problem is a multi-class, multi-label text categorization.

In a nutshell, text categorization is the task of automatically attributing one or more pre-defined categories to each document of a given set. In the last decade, the interest in text categorization has witnessed a boom [10]. Part of this booming is due to the successful application of machine learning techniques to that task. In supervised learning, a machine learning technique, it is used an amount of data labeled manually to learn a mapping between inputs and outputs [8]. Discriminative models like SVMs are one of the machine learning models that several authors point out as achieving better

performance [11]. An initial tutorial in using SVMs to pattern recognition problems is presented in [1].

The codification problem has as input a short textual description of the economical activities that a person or company wants to perform. One of the main problems in treating this as text categorization is the amount of categories (or codes) that an input could be classified in. The input could receive a dozen codes out of one thousand. Furthermore, until now, a representative amount of data to train a classifier is not available. The data set gathered so far has a very poor quality, and there are few examples of most of the codes. To use machine learning techniques in this context is not trivial, but it was decided to tackle these difficulties with a hierarchical SVM approach.

The solution proposed makes binary classifications in each level of the hierarchy present in the table of codes, as the upper levels are general activities or process and the lower levels represent specific economical activities. The hierarchical separation helps to deal with the large number of possible categories to be handled, so that in any level, the number of categories is about twenty. Although the table has five levels in its hierarchy, in this paper it is presented the methodology up to the second level, as it is virtually easy to extend to other levels. The decision of considering only the first two levels is due to the amount of data representing those levels. There are not sufficient data to represent the other levels properly. Maybe it will be almost impossible to apply the same approach in the last level, because many economical activities predicted in the table encounter few realizations in the country, and some not even a single realization.

Because all the difficulties of this problem regarding the data set available, it is not expected to obtain classifiers with enough accuracy to implement a fully automatic codification. So it is being proposed, as a first step, to implement a semi-automatic codification process that includes some probability measure attached to each result, indicating the belief in the accuracy of the results based on training quality measures. This approach would help human experts by producing a small subset of possible codes with probabilities assigned to each code to guide a final decision.

2. The problem of codification of economical activities

The economic activities codes are described in a table named CNAE (National Classification of Economic Activities) and its elaboration follows international standards. The table is organized in a hierarchy of Sections, Divisions, Groups, Class and Sub-classes, totalizing 1,301 sub-classes or codes in its most recent version, 2.0 (See Table 1 that pictures the 1.1 version of this table, used in the experiments presented in this paper). Each section contains some specific group of economical activities.

In the official document describing each subclass or code, it is presented some short description of the economical activities under this category, followed by fields as *'This subclass contains'*, *'This subclass contains also'* and *'This subclass does not contains'*. This sort of structure in the CNAE subclasses description makes it difficult to use any sort of pure logic modeling without handling in full the complex semantic involved in textual descriptions. Thus, machine learning techniques seem to be more appropriated for this problem, formulated as a text categorization problem dealing only with the words presented in the input textual description.

Table 1. Version 1.1 of CNAE table of codes.

Sections	Divisions	Groups	Classes	Subclasses
Totals	59	222	580	1178
A	2	7	25	91
B	1	1	2	11
C	4	7	14	42
D	23	104	286	395
E	2	4	7	8
F	1	6	16	43
G	3	19	72	223
H	1	2	7	16
I	5	14	29	77
J	3	11	27	65
K	5	24	38	74
L	1	3	10	10
M	1	4	10	17
N	1	3	9	35
O	4	11	26	69
P	1	1	1	1
Q	1	1	1	1

One of the difficulties in machine learning problems is the availability of data to train properly the classifier. Because of the large number of possible codes in this problem, in both table versions, there is a necessity for a large dataset of examples where all these codes appear many times. Other difficulty is that the number of words per text description is reduced to a few words in the majority of the cases (See Fig. 1). The average count of words per text description is 6, representing 8% of all the cases in the corpus.

Still, another problem related to the short description of economical activities using a few words is that in some cases there are not enough words to characterize all economical activities involved. In these cases, the codes were attributed by a human that obtained additional information by asking more questions to the business owner. That information was not registered in the textual description available for training and testing. As an example, consider a textual description containing only the word “bar”. Two codes were attributed to it relating the business activities: “bar activities” and “fast-food activities”. This is one of the causes that contributed to the poor quality of the data available and restrict the accuracy of the final classifier.

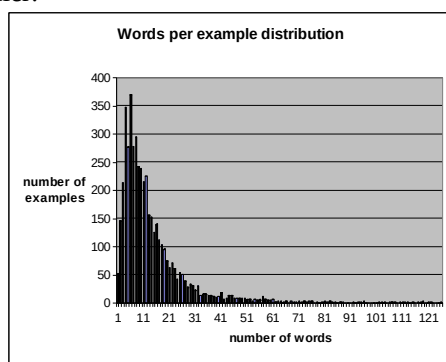


Figure 1. Word's per example distribution.

To handle many of these difficulties in trying to improve the performance of a single layer of binary classifier, it is proposed some features in the model: a hierarchical approach to reduce the number of categories in each level and a metric to be applied to the results of the SVMs both to recover some misclassified input in higher levels and to give a probability measure of the correctness of the results.

3. Support vector machine implementation

The basic implementation of SVMs follows Joachims (1998) pioneering approach to problems of text categorization. Each dimension of the input vectors is correspondent to a word of a specific vocabulary. The quantities in the vectors amount to the frequency of that word in the textual input, multiplied by a weight measuring the importance or relevance of that word in the training corpus. This measure is calculated as $\log(m/N)$, where N is the number of documents in the corpus, and m corresponds to the number of documents of the corpus in which that word appears. This measure takes into account the due weight of

each word improving generalization [12]. SVMs are trained using polynomial kernels and the input is normalized.

As a result, SVM outputs the margin or a measure associated to the distance of a document from the hyper-plane that separates the space in two volumes. A positive margin indicates that the document belongs to the category been tested; otherwise, it does not belong. Not just the signal is important in that result, the value of the margin, as it is related to the Euclidean distance from the hyper-plane, could be used to measure the certainty of the classification. Documents with margin between -1 and 1 have lower probability of being correct than documents with a margin higher than 1 or lower than -1.

As there is no reference in the problem of multi-class text categorization dealing with more than one hundred labels [6], it was decided to handle the initial stage of the semi-automatic codification with a hierarchical use of SVMs to handle this dimension of different codes, considering only the words present in the textual descriptions.

3.1. Hierarchical approach

The hierarchical approach was proposed due to the large number of different categories (1,301 codes) that could be attributed to a single document. Obeying the hierarchical structure of the CNAE table, initially it was decided to treat the first two levels: Section and Groups. Dealing only with sections, the possible categories decrease to 21 (in the 2.0 version of the CNAE table). In the division level, each section contains less than 24 categories (this number corresponds to the C Section; however, there are 4 divisions per section in average). This reduced order in the number of labels can be easily treated by SVMs. Still, the hierarchical categorization approach brings more benefits than just reducing the order of the problem. Each section is expected to contain a particular collection of terms because it specifies a particular economical activity, facilitating the detection of false positives coming from upper levels. Thus an important feature of the hierarchical approach is to filter out part of the cases misclassified in that section. To develop this filtering property, in the division level, all divisions of a given section are trained with only the examples that belong to that section. Doing so, the classifier is trained only to distinguish between divisions of a section and not between all the possible cases of other sections,

concentrating in a particular subset of words or dimensions of the input vector.

With this filter feature, each level of the hierarchy must receive more cases from the upper levels to solve the misclassifications. This is obtained with a metric to deliver more cases beyond those with positive margin to the lower levels of the hierarchy.

3.2. Metric to recover false negatives

The function of the metric is two-fold: to recover cases misclassified as not-belonging to a given section, and to attribute a probabilistic measure of the quality of the classifier's responses that will give to the human specialist a hint of the correctness of the possible suitable codes.

Besides the margin, there are some other values that characterize the quality of the trained classifier: recall and precision in the training set. The recall value is related to the relative quantity of cases belonging to a section that was recovered by the classifier, among all that really belong. The precision value is related to the relative correct classification of cases that belong to a section. These two values are dependent on each other and it is difficult to improve one without lowering the other during training.

The proposed solution is to use a metric to recover misclassified cases that truly belong to a category in order to improve the recall measure. Furthermore, the hierarchical approach is applied to filter out extra cases from the ones that are misclassified as belonging, thus improving the precision measure. In this implementation the metric used to recover misclassified cases is different than the metric used to attribute the probability measure.

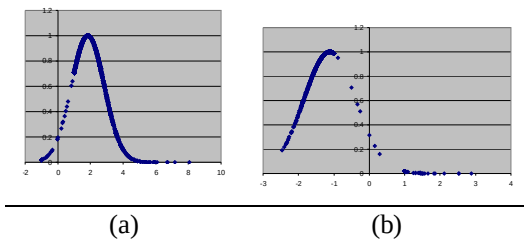


Figure 2: Classifier's results from G section belonging documents

The goal of the attribution of probabilistic measures is to aid a human specialist to better interpret the results of the machine, in the case of semi-automatic classification. Also, it reflects the quality of the data

handled by the algorithm. With better data, the results tend to have higher probabilities associated to it.

In Figure 2 (a) there is an approximation of the distribution of documents belonging to G section with regard to the classifier responses. It depicts that in training, the classifier don't have zero error as a few documents are classified with negative margin. Figure 2 (b) shows the case of documents that do not belong to G section. Some are misclassified with positive margin. In the hierarchical implementation it is of interest to recover cases that belong to a given section but that received a negative margin by the classifier. The region between -1 and 0 of Figure 2 comprehend those cases.

Here it is considered four distinct possibilities and a different metric to each one of them. The measures of recall, precision and error rate at the training are used to compose specific metrics.

The first case encloses all documents that receive a negative margin less than the mode of the distribution of cases not-belonging to the given section. In the case of a normal distribution, the mode and average are the same.

$$Metric(m) = error * \exp\left(-\frac{(m - \mu_N)^2}{2 * \sigma_N^2}\right)$$

The second case deals with documents with negative margin higher than the mode.

$$Metric(m) = precision - \left(B * \exp\left(-\frac{(m - \mu_N)^2}{2 * \sigma_N^2}\right) + D \right)$$

$$B = (precision - error) / \left(1 - \exp\left(-\frac{\mu_N^2}{2 * \sigma_N^2}\right) \right)$$

$$D = (precision - error - B)$$

The third case deals with documents with positive margin but with values less than the mode of the distribution.

$$Metric(m) = A * \exp\left(-\frac{(m - \mu_p)^2}{2 * \sigma_p^2}\right) + C$$

$$A = recall + C$$

$$C = \left(precision - recall * \exp\left(-\frac{\mu_p^2}{2 * \sigma_p^2}\right) \right) / \left(1 - \exp\left(-\frac{\mu_p^2}{2 * \sigma_p^2}\right) \right)$$

Finally, the fourth case considers the documents with positive margin with values higher than the mode of the distribution.

$$Metric(m) = 100 - (100 - recall) * \exp\left(-\frac{(m - \mu_p)^2}{2 * \sigma_p^2}\right).$$

4. Experiments

The experiments presented bellow were a preliminary test using SVMs in a cascade of sets of binary classifiers to solve the problem of classification of free-text in codes belonging to the hierarchical CNAE table. All the results were obtained using SVM-Light software package, developed by Joachims (1999).

Until now, we have access to two datasets, one with approximately 88,000 descriptions from the city of Belo Horizonte and another with 21,000 descriptions from the city of Vitoria. Here the data used was from Belo Horizonte, coded in CNAE 1.1. The software handled very well this dimension of examples. However, only a small sample was used in the experiment because each example needed to be manually corrected before it could be used as input to the software due to the fact that many words were typed incorrectly.

The resulting training corpus had 4,747 cases which distribution in sections of the CNAE's table is presented in Figure 3. The set of examples to the tests were made with 1,406 cases. The input vector had 4,198 dimensions, but the vectors were very sparse.

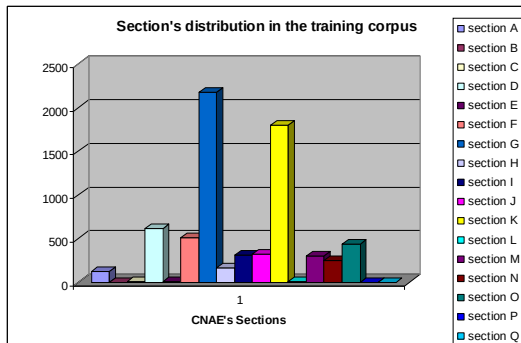


Figure 3: Distribution per sections in CNAE table of training data

Due to the number of cases of each section inside the training corpus, it was considered only sections A, D, F, G, H, I, J, K, L, M, N and O. The proposed experiment consisted in a binary-classification of all cases related to the sections in the CNAE table that contained more than 100 examples in the training dataset.

In Table 2, the results are shown for the training and the test, considering the sub-set of sections. Considering individual discrimination of sections, as it could be noted in the Accuracy column of Table 2, the results could be considered very good. It must be considered, in analyzing these values, that in cases

like Section A with few test examples, the accuracy value could be misleading.

Table 2 - Training and test results.

Section	Training			Test		
	Error (%)	Recall (%)	Precision (%)	Accuracy (%)	Precision (%)	Recall (%)
A	1.96	53.17	66.34	99.36	40.00	25.00
D	14.79	33.66	41.32	92.32	76.87	57.22
F	7.37	55.62	69.29	95.16	83.33	70.51
G	14.03	82.13	86.66	93.24	94.54	92.34
H	2.25	63.98	67.76	98.79	94.59	70.00
I	5.03	53.87	63.50	98.29	93.10	72.97
J	5.33	54.91	62.81	97.44	95.95	68.27
K	18.62	71.92	77.42	91.32	91.72	84.38
M	5.10	49.50	61.92	97.80	86.11	74.70
N	3.79	60.00	65.22	98.01	77.42	77.42
O	7.67	42.49	61.54	95.87	80.00	51.06

In the first layer of classifiers, the section level, the overall accuracy was of 68.70%. By overall accuracy we mean that in a correct classification a particular document receive a positive margin from the binary classifiers of all the sections to which it belongs, and negative margin of all the other sections. This is a very harsh criterion. Sections that are not considered in the test does not influence the overall accuracy. For example, as there is not a binary classifier to section B, if a example belongs to this section and to section F that possess a classifier, a correct classification means a positive margin in the Section F classifier, and a negative margin in the other classifiers. Without considering section D, that in CNAE version 1.1 presents a major number of divisions including more than 20 codes and in training appears with low levels of recall and precision, the overall accuracy raises to 73.19%. Not considering sections D (poor recall and precision in training) and K (poor error in training), the overall accuracy raised still more to 78.10%. The main reason for this behavior is the poor quality of data used in training, principally in cases that belong to particular sections like D and K.

The training of the division level considered only cases belonging to that section, so that the classifier could focus on discriminating only between divisions of the same section. The results obtained are shown in the table below.

In the second layer of binary classifiers, if a example is considered by the system as belonging to a given section, a sequence of classifiers of all the division of that section are tested. The overall accuracy, as defined above, is 57.75% without considering the metric (58.11% considering the filtering of the hierarchical approach and 58.32% with the filter and the metric approach). Disregarding the divisions of D and K sections, the overall accuracy raised to 70.20%.

Table 3 - Training results in the division level of the hierarchy.

Section	Division	Error (%)	Recall (%)	Precision (%)
A	01	8.73	100.00	91.13
	02	9.52	14.29	100.00
D	15	3.90	50.00	87.50
	17	5.69	22.22	16.00
	18	8.94	62.38	78.75
	19	2.76	23.08	30.00
	22	10.41	76.02	84.97
	28	6.20	20.69	30.00
	29	25.04	51.40	57.86
	33	7.64	17.24	17.86
	35	0.33	33.33	100.00
	36	6.02	27.27	40.91
G	50	4.49	73.73	82.86
	51	30.51	61.51	57.60
	52	28.45	78.82	80.03
I	60	20.65	73.19	78.91
	63	26.13	61.86	66.97
	64	17.74	66.67	69.05
J	65	11.35	93.75	86.39
	66	11.96	6.67	3.85
	67	19.33	77.07	81.76
K	70	9.32	64.81	60.34
	71	13.82	50.00	48.19
	72	13.65	63.90	65.01
	74	24.47	83.75	82.93
	90	4.16	21.05	57.14
O	91	7.39	83.22	94.66
	92	12.47	87.08	87.08
	93	5.08	74.24	90.74

However, as the initial interest is to use the classification as an aid to human classifiers, each output has a probability measure associated to it. Consider the following economical activities description:

“Selling at retail films, photographic cameras and electronic equipment. Installation and maintenance of electronics equipment and revelation of photographic films”. That description has the following codes: 5144-6/02 (section G, division 51), 5242-6/02 (section G, division 52) and 7491-8/03 (section K, division 74).

Table 4 - Result of the codification.

Sections	Divisions	Margin	Certainty measure (%)
D		-0.78	20.78
F	45	-0.87	13.55
G		3.52	95.35
	50	-0.95	
	51	-0.52	
	52	0.21	
I		-1.52	6.12
J		-1.55	6.14
K		0.24	68.80
	70	-1.20	
	71	-1.32	
	72	-1.46	
	73	-0.89	
	74	0.34	
M	80	-1.75	4.35
N	85	-2.55	0.71
O		-1.73	7.8

The margins and probability measures attributed to that input are shown in Table 4.

The sections G and K have higher probabilities assigned to them and are the only with a positive margin demonstrating that the metric is suitable for treating this problem. This leads to the testing of the next level, the division level for these two cases. In the division level, the classifier assigned correctly the two divisions, 52 of section G and 74 of section K. However, it failed in finding division 51 of section G, although the margin of division 51 was in the recoverable region between -1 and 0.

5. Conclusions and future works

The results presented here are still preliminary but they already provide an aid to the human codifier in selecting some possible codes to a given economical activity description (actually the codes correspond to section and division levels of the CNAE table) by assigning a probability measure that reflects the degree of belief in the results of the classifier.

Although the amount and overall quality of data is not sufficient to produce a set of classifiers completely accurate for the task of automatic codification, the results demonstrate that the proposed solution is suited to this class of problem, providing a way to further development.

Regarding the data used to train the classifiers, the Brazilian government is undertaking a huge effort inviting selected companies to participate in a survey which will collect a large amount of data followed by a rigorous manual classification. With the availability of this data, it is expected to obtain a large improvement in the quality of the classifiers.

The next step in improving the classification results is to work with different representation of the textual entries. One possibility is the use of *bag of concepts* representation [9], including some terms more directly associated with a product of an economical activity or with verbs related to the production of goods or services. Another alternative, still in an early stage of implementation, is the incorporation of semantics in the description of the text according to the approach by [3] to integrate *latent semantic analysis* [4] with the kernel in the SVM.

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7. References

- [1] C.C.J Burges, “A Tutorial on Support Vector Machines for Pattern Recognition”, *Data Mining and Knowledge Discovery*, 2 (2), 1998, pp. 121 – 167.
- [2] N. Cristianini, and J. Shawe-Taylor, “An Introduction to Support Vector Machines and Other Kernel-based Learning Methods”, Cambridge University Press, 2000.
- [3] N. Cristianini, J. Shawe-Taylor, and H. Lodhi, “Latent Semantic Kernels”, *Journal of Intelligence Information Systems*, 18:2/3, 2002, pp. 127-152.
- [4] S. Deerwester, S.T. Dumais, G.W. Furnas, T.K. Landauer, and R.A. Harshman, “Indexing by Latent Semantic Analysis”, *Journal of the American Society for Information Science*, 41(6), 1990, pp. 391–407.
- [5] D. Giorgetti, and F. Sebastiani, “Automating Survey Coding by Multiclass Text Categorization Techniques”, *Journal of the American Society for Information Science and Technology*, 54, 14, 2003, pp. 1269-1277.
- [6] T. Joachims, “Text Categorization with Support Vector Machines: Learning with Many Relevant Features”, *Proceedings of 10th European Conference on Machine Learning*, number 1398, Springer Verlag, 1998, pp. 137-142.
- [7] T. Joachims, “Making Large-scale SVM Learning Practical”, *Advances in Kernel Methods - Support Vector Learning*, B. Schölkopf and C. Burges and A. Smola (ed.), MIT-Press, 1999.
- [8] T. Mitchell, “Machine Learning”, WCB/McGraw-Hill, 1997.
- [9] M. Sahlgren, and R. Cöster, “Using Bag-of-concepts to Improve the Performance of Support Vector Machines in Text Categorization”, *Proceedings of the 20th International Conference on Computational Linguistics*, Geneva, Switzerland, 2004.
- [10] F. Sebastiani, “Text Categorization”, *Text Mining and its Applications to Intelligence, CRM and Knowledge Management*, WIT Press, Southampton, UK, 2005, pp. 109-129.
- [11] V.N. Vapnik, “The Nature of Statistical Learning Theory”, Springer Verlag: Heidelberg, 1995.
- [12] Y. Yang, and J.O. Pedersen, “A Comparative Study on Feature Selection in Text Categorization”, *Proc. 14th International Conference on Machine Learning*, Morgan Kaufmann, 1997, pp. 412-420.

Adaptation of Surface Roughness Models based on AI in Changing Environments

J.V. Abellan, F. Romero, C. Vila
Universitat Jaume I
Dept. of Industrial Systems Eng. and Design
12,071 Castellón, Spain
{abellan,fromero,vila}@esid.uji.es

R. Morales-Menendez, H.R. Siller
Tecnológico de Monterrey, Campus Monterrey
Avenida Garza Sada 2501 Sur.
64,849 Monterrey, NL, Mexico
{rmm,hector.siller}@itesm.mx

Abstract

Machining empirical models for surface roughness based on statistical or artificial intelligence (AI) approaches have been intensively studied during last decades. However, there is no industrial methodology to be applied in industry due to the time consuming and costly experimental procedures required. Furthermore, continuous changes in the cutting process such as cutting-tool replacements or changes in material workpieces make them unfeasible. This paper presents two methodologies to develop empirical models based on few experimental runs based on design of experiments, Taguchi's array, and AI techniques to learn the model from experimental data. The first methodology models the system from the scratch. The second methodology adapts a previous AI model with few experimental runs, defining two possible methods for adaption according to model change. A case study is presented, where a face milling operation with two cutting-tool changes is analysed. The results show a steady prediction error less than 20% along process changes.

1. Introduction

After one century from machining systems, there is not enough understanding about machining processes to describe accurately the surface roughness generation. Several decades ago, many researches moved from an analytical point of view to a more empirical approach in order to define practical models for surface roughness prediction. These models can be applied to optimize cutting parameters, assure part specifications, decrease inspection activities and maximize the use of cutting-tool. However, these models require a tedious experimental procedures and in some cases it also requires costly and impractical sensor systems. Although these models' efficiency have been suc-

cessfully proved, they are still far away to be applied in industry. Therefore, industry still demands a practical, effective and reliable procedure to model surface roughness accurately. Recent developments in machine learning field promise future empirical surface models which are able to adapt its prediction ability according to changes in cutting tool, machine dynamics, workpiece material, etc.

This paper describes a methodology to model surface roughness from the scratch with few experimental runs in a specific machining operation, and a methodology to adapt any change in the cutting process with few experimental data without any sensor systems. Both methodologies are based on Design of Experiments, Taguchi's arrays and Artificial Intelligence (AI) models. These approaches can provide a practical and efficient methodology to be applied in industry. The paper is organized as follows. First, section 2 introduces briefly the empirical models applied in machining for surface roughness prediction and the adaptability problem in changing environments. Section 3 describes step by step the two methodologies proposed. A case study is presented in section 4 where the results of applying the proposed methodologies are discussed. Finally, section 5 concludes the paper.

2. Empirical predictive models for surface roughness based on AI

During the last decades, artificial intelligence techniques have been intensively applied in machining processes. Their ability for dealing with uncertainty, missing data and high numbers of variables as well as their ability for learning and adapting models with new experimental data makes them a more effective approach than traditional statistical methods [6]. The most common artificial intelligence techniques applied in machining systems are Artificial Neural Networks (ANN) and Neuro-Fuzzy systems, although other approaches such as genetic algorithms, Markov hidden mod-

els and Bayesian models have been also applied successfully. Chien and Chou [4] developed three ANN models for predicting not only surface roughness, but also cutting forces and cutting-tool life in turning operations. Tsai [8] predicted the surface roughness in milling operations using several multilayer backpropagation neural networks. He compared his results on regression models and pointed out the best performance of ANN models (error around 4%) versus regression models (error around 8.7%). Similar studies where ANN models presented better performance than regression models were reported by Ozel [6] and Lee [5]. Benardos [3] applied ANN techniques and Taguchi's design of experiments in order to minimise the experimental runs. He applied a L-27 orthogonal array to conduct the experiments and the cutting parameters considered in the design of experiments were depth of cut (a_p , 3 levels), feed rate per tooth (f_z , 3 levels), cutting speed (V_c , 3 levels), engagement (a_e , 3 levels), cutting tool wear (V_B , 2 levels) and cutting fluid (C_f , 2 levels). The accuracy error of the ANN model learnt was around 1.86%. Neuro-fuzzy systems have been also applied for surface roughness prediction with good results. Abburi y Dixit [2] applied an Adaptive Neuro-Fuzzy Inference System (ANFIS) for predicting surface roughness in turning operations. The learnt model can be applied for predicting the surface roughness and also for estimating the best cutting parameters according to the rules learnt.

In spite of the huge number of research studies around empirical surface roughness models, there is no methodology applied in industry to model effectively surface roughness based on few experimental data. The costly and time consuming experimental procedures required for empirical modeling prevent the application of most of the models presented in the literature. Furthermore, any change of the process with respect to the initial process where the experiments were conducted implies an additional prediction error which difficulties the use of the model in the current process. Few works have presented methods for adapting empirical models due to changes in production (cutting-tools, workpiece material, etc.). Westkamper [10] and Vanluttervelt [9] exposed the adaptation problem, but they did not deal with any physical example. Risbood [7] modeled surface roughness and dimensional quality in turning using an ANN and tested the use of the model in different machining conditions which were not modeled. He showed that it was necessary to apply two different ANN models for modeling the cutting process depending on the presence of coolant, as the cutting process changes considerably due to coolant effects. In order to adapt the surface roughness model when a cutting-tool change occurred, Risbood proposed a proportional compensation. This compensation assumes that the model changes slightly, so a straight line between the actual and predicted data can be fitted, and this linear relationship can be used for adapting the initial ANN model to the new

process with a new cutting-tool geometry. This adaptation showed a prediction error similar to the initial ANN model, around 20%. However, his investigation does not provide any methodology for minimising the experimental runs, and it is not explained how to adapt sever process changes.

3. Methodology for modeling and adapting empirical surface roughness models based on AI

The methodology presented in this paper for surface roughness modeling and adaptation is divided in two parts. First, it is presented a methodology to model surface roughness through minimal experimental runs, depending on the number of factors that are added into the model. Secondly, it is presented a methodology to adapt this model in case a process variation is presented, such as a cutting-tool geometry change. The methodology relies on statistical and artificial intelligence techniques to learn from experimental data, and fractional factorial design of experiments and Taguchi's array to minimise the experimental runs acquiring the most important relationships among machining variables.

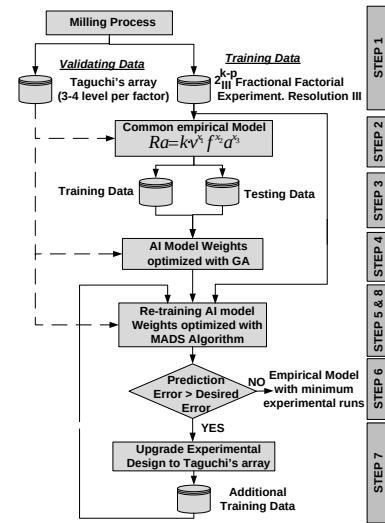


Figure 1. Methodology (a) for surface roughness modeling with few experimental data.

3.1. Modeling an empirical surface roughness model with few experimental data

The methodology (a) proposed for modeling surface roughness is shown in Fig. 1 and it consists of eight steps as follows:

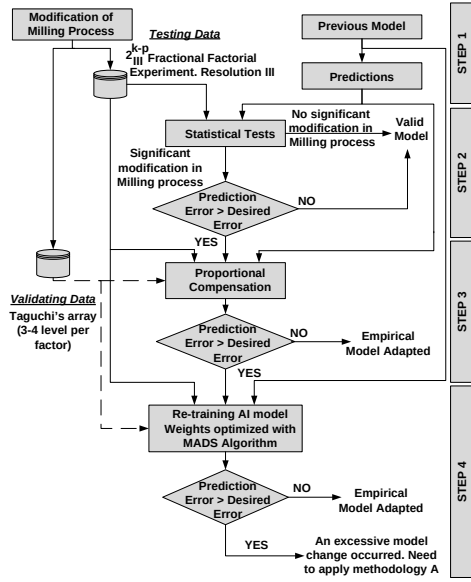


Figure 2. Methodology (b) for adapting surface roughness models with few additional experimental data.

1. Step 1: Experimental design. In order to minimise experimental runs, a 2^{k-p}_{III} fractional factorial experiment with resolution III is conducted, where k is the number of factors in the process and p is chosen to get an experiment with resolution III. With this experiment, the main effects are aliased with the first interactions. Additionally, at least one replication is conducted. For the validating set an experimental design based on Taguchi's array is conducted, in order to minimise the experimental runs. Taguchi's arrays such as L-9, L-18 and L-16 can be applied depending on the number of levels of each factor.
2. Step 2: Fitting an empirical model. The common empirical model in Eq. (1) is fitted by a least squares regression using training data. This model explains part of the process variability, but it is usually not accurate enough to be applied in practice.

$$Ra = kV_c^{x_1} f_z^{x_2} a_p^{x_3} \quad (1)$$
3. Step 3: Generating training data. The empirical model learnt in step 2 is used for generating fictitious experimental data which are utilised in the next step to learn an artificial intelligent model.
4. Step 4: The artificial intelligent model. An artificial intelligent model is built based on the data generated in step 3. Here, two artificial intelligent models can be applied: artificial neural networks (ANN) and Adaptive Neuro-Fuzzy Inference Systems (ANFIS). Due to

the reduced experimental data (although step 3 generates many data, the actual experimental data conducted is only based on the first fractional factorial experiment), ANN is preferred to ANFIS, as ANN requires less data to learn the model than ANFIS. However, ANN cannot explain with rules the model learnt, and difficulties the understanding of the model as it can be done through ANFIS. The ANN model structure should be chosen according to three considerations: a high number of neurons and layers require large quantity of data for training; a very limited number of neurons prevents the ANN model to learn the actual model behavior; although the machining processes are non-linear, the process is not highly non-linear, and only effects such as built up edge, or chip formation defects and chatter provide a non-linear behavior. Therefore, one can assume that for modeling purposes with few experimental data, it is not necessary to apply complex ANN models with large number of neurons and layers. On the other hand, the learning process uses as training data the data generated in step 3 whereas the validating data is defined in step 1. Due to the scarce data training, a Genetic Algorithm is applied to optimise the neural weights initialization and avoid the convergence at a local minimum. In order to avoid over-training and improve generalization, the validating data is used for stopping the training process when the error in validating data increases (early stop criterium).

5. Step 5: Re-training the AI model. In step 4, the AI model only learns the variability of the process explained by Eq. (1). Now, a re-training process is conducted to use the initial training data from the fractional factorial design in order to incorporate this information into the AI model. Now, instead of using GA it is applied a Mesh Adaptive Direct Search algorithm (MADS) to refine the initial neural network weights using as the first mesh iteration the neural network initialization provided by the GA.
6. Step 6: Performance model evaluation. The AI model is used to test the prediction ability on the validating data. If the error is higher than the desired error, an additional experimentation must be conducted. If not, an adequate surface roughness model has been trained. A high error can be due to scarce experimental data to model the process, stochastic behaviour of the machining process or noisy experimental data.
7. Step 7: Increasing experimental data with Taguchi's arrays. As the initial training data is based on a simple fractional factorial design where factors are evaluated with two level, it may be necessary to increase the experimental design in order to get more meaningful data for modeling purposes. The 2^{k-p}_{III} is upgraded

to another kind of fractional factorial design, called Taguchi's array. From the initial two levels, the most influencing factors increase their levels up to three or four levels, and a Taguchi's array is accommodated according to the number of factors and levels.

8. Step 8: Re-training the AI model. The new training data from Taguchi's array experimentation is used for retraining the AI model, where the early stop technique is again applied. After training, the steps 6 and 7 are again conducted.

Please note that the effectiveness of steps 3, 4 and 5 are validated later in the case study.

3.2. Adapting an empirical surface roughness model with few additional experimental data

To adapt any change in the machining process such as cutting tool, workpiece hardness, etc. the methodology (b) presented in Fig. 2 is proposed. The methodology (b) is divided in four steps as follows.

1. Step 1: Experimental design. A similar experiment is conducted as in methodology (a), based on a 2_{III}^{k-p} fractional factorial Design of Experiments with resolution III. At least one replication is conducted.
2. Step 2: Statistical tests. Statistical tests are conducted to infer if the new machining process is statistically equivalent to the previous one within a certain confidence level. Two tests are analysed: the Levene's test to check if the prediction errors in both machining processes have equal variances, and the two-sample t -test to determine if the mean of the prediction errors are equal [1]. The results of both tests indicate if statistically the machining processes are equivalent or not. If they are equivalent, the roughness model can be applied without adaption. Otherwise, if a significance process variation occurs the prediction error is quantified. If the variation produces a prediction error less than the desired one, the previous model can be applied. In case the prediction error is higher than the desired prediction error, a model adaption is required.
3. Step 3: Proportional compensation. A first step for model adaptation proposes a proportional compensation as it is shown in Ref. [7]. This compensation is based on the assumption that the modification of the machining process produces a lineal deviation of the expected surface roughness, for example a higher roughness due to a smaller nose radius. In this case, the training data obtained from step 1 is used for fitting a lineal regression model where the response variable

is the actual R_a and the predictor is the predicted R_a by the AI model. For validation purposes, additional experiments based on Taguchi's array are conducted, with at least three or four levels for each factor. If the prediction error is less than the desired one, the model is assumed to be adapted and is valid to be used. Otherwise, the surface roughness model is not able to predict the actual machining process, and more than a proportional compensation is required.

4. Step 4: Re-training AI model. In case the proportional compensation is not valid, a re-training process of the AI model is required. Re-training consists of adapting the roughness model of the previous machining process using the training data of the current machining process. During training, the weights of each neuron are modified in order to minimise the prediction error. Similar to the previous methodology, a MADS algorithm is applied in the re-training process to avoid a local minimum convergence. The final prediction error of the adapted model is calculated based on validating data. If the prediction error is higher than the desired one, it is considered an excessive change in the process machining behavior, and the methodology (a) should be conducted in order to model surface roughness from the scratch. Otherwise, the model has been adapted successfully and it can be applied on the modified machining process.

4. Case Study

In order to validate the methodology proposed, a face milling operation of AISI D3 workpieces with three different cutting tools was studied. First, the face milling operation was carried out with a cutting tool with 25mm diameter and two inserts with ISO code denomination RDHW1002MO and uncoated grade. The methodology (a) was conducted to model the surface roughness. Later, the effect of two cutting-tool changes on the face milling process were analyzed. In the first cutting-tool change the machining operation was conducted with a different cutting-tool insert geometry defined by the ISO code RDHW0802MO and 20mm diameter, and the grade and the number of inserts was identically to the previous cutting operation. In a second cutting-tool change, the cutting tool used consisted of a five inserts with ISO code RDHX12T3, with 52mm diameter and grade TiAlN/TiN-PVD. The experimental setups for each face milling modification are shown in Fig. 3. The cutting parameters recommended for finishing and applied in the DoE are: $180m/min \leq V_c \leq 220m/min$ and $0.04mm \leq f_z \leq 0.2mm$. Table 1 shows the main characteristics of the different AI approaches applied in both methodologies (a) and (b) for the case study.

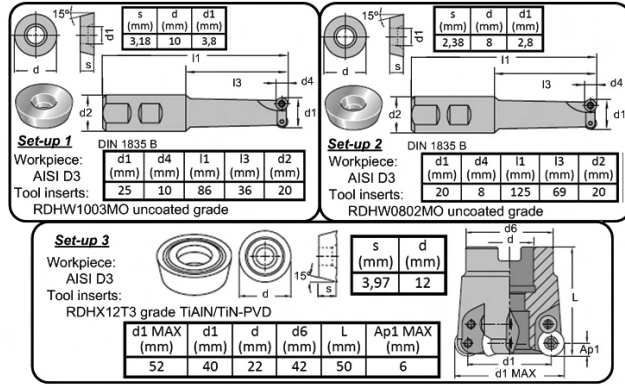


Figure 3. Setups for the machining process studied.

4.1. Empirical Surface Roughness Model

The empirical surface roughness model was obtained through the methodology (a) and the setup 1, and the factors modeled were cutting speed (V_c), feed rate per tooth (f_z) and depth of cut (a_p). Initially, a fractional factorial design 2^{3-1}_{III} based on two levels per factor were conducted with two replicates to provide the training data (step 1). The validating data was obtained through a design of experiment based on Taguchi's array $L - 9$ in order to evaluate three levels per each factor with few runs. The training data was used to fit the Eq. (1) by a least squares regression (step 2). The coefficients fitted were $x_1 = -1.51$, $x_2 = 0.21$, $x_3 = 0.44$, $k = 3.58$, and the prediction error on validating data was $E_{step2} = 21\%$ (Fig. 4 (a)). The resulting regression was used for generating 200 fictitious experimental data (step 3). These fictitious data was used to train an ANN model, using GA for optimising the neural network weights (step 4). In order to choose the most simple ANN structure for surface roughness prediction, a brief experimentation was conducted. This experimentation showed that structures with less than 3-4 neurons seemed to be unable to model the machining process, whereas higher structures (with one or more hidden layers and neurons) seemed to have a similar performance but required more experimental data to be learnt. Therefore, an ANN network with 1 hidden layer and 4 neurons was chosen. After training the ANN model with the fictitious experimental data (step 4), the ANN model was re-trained with the training data obtained in step 1 (step 5). The ANN model was evaluated with the validating data after re-training, using a MADS algorithm for optimising the neural network weights (step 6), with a final prediction error of $E_{step6} = 12.83\%$. In order to improve the model accuracy, the design of experiments was upgraded using a Taguchi's array L-9. The additional

experimental data were used to re-retrain the ANN model using again a MADS algorithm (step 7), and the final prediction error was $E_{step7} = 9.63\%$ (Fig. 4 (b)).

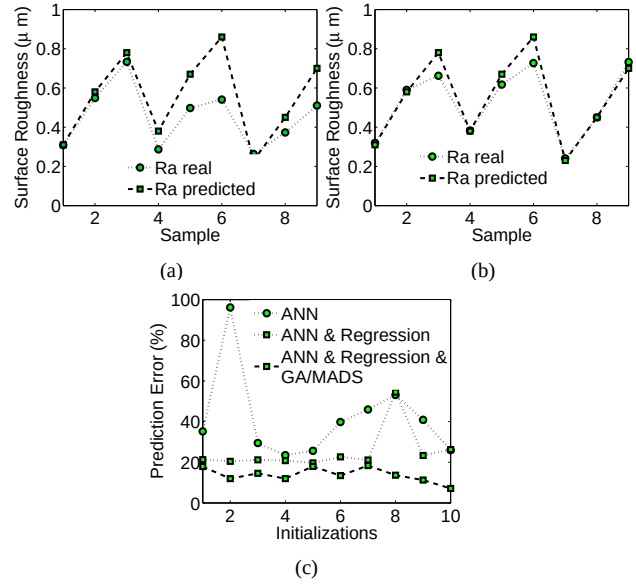


Figure 4. (a) Setup 1. Ra real versus empirical regression model after methodology (a), step 2. (b) Setup 1. Ra real versus ANN predictions after methodology (a), step 7. (c) Comparison of ANN prediction errors on validating data using: the fractional design data directly for training (legend ANN), and the training data generated by the regression without GA/MADS algorithms (legend ANN & Regression) and the proposed methodology (legend ANN & Regression & GA/MADS).

The methodology (a) conducted in this case study is based on using a previous regression model fitted by training data and applying GA and MADS algorithms to minimise a local minimum convergence. In order to test the feasibility of this approach, this methodology was compared with two other possible methods for training. The first comparison consisted of applying the training data from the fractional design directly in the neural network without GA/MADS algorithms. A second comparison consisted of applying the training data generated from the fitted regression without GA/MADS algorithms. Fig. 4 (c) shows the results of each method after conducting the training 10 times. As it is shown, the proposed methodology reports a lower prediction error on validating data than the other two methods because GA/MADS algorithms tend to prevent a local minimum convergence when few experimental data are used.

Table 1. Characteristics of AI techniques applied.

ANN model			
Type Network	MLP	Training alg.	Lev.-Marq.
Hidden layers	1	Epochs	200
Neurons	4	Stop criterium	Early Stop.
Map. functions	Logsig	Learning rate	0.05
Genetic Algorithm			
Num. variables	16	Crossov. Frac.	0.8
Population Size	10	Elite Count	2
Stall Generations	Inf	Mutation func.	Gaussian
Stall Time	Inf	Selection func.	Stochastic
Generations	8	Initial Ranges	[-1.5,1.5]
Mesh Adaptive Direct Search			
Num. variables	16	Expansion	2
Initial Mesh Size	0.01	Contraction	0.5
Max. Mesh Size	15	Poll Method	Pos. 2N
Max. Func. Eval.	40	Polling order	Consec.

4.2. Adapting the Empirical Surface Roughness Model

The machining process was modified to analyze model adaption for setup 2 (slight cutting-tool geometry change) and 3 (strong cutting-tool geometry change).

4.2.1. Adapting a slight cutting tool geometry change (Setup 2). The experimental set-up was modified to setup 2 as it is shown in Fig. 3. The methodology (b) was applied based on the previous ANN model. Firstly, a fractional factorial design 2^{3-1}_{III} was conducted to provide the testing data (step 1). Using the previous ANN model, the prediction error on testing data was statistically compared with the prediction error in setup 1. The statistical test Levene's test with a $p - value = 0.006$ indicated a change in the variance of the prediction error whereas the two-sample t -test with a $p - value = 0.082$ did not indicate a change in the mean error. Since a variance change was detected, the surface roughness model obtained previously did not reflect the actual machining process. Fig. 5 (a) shows the ANN predictions by the previous ANN model and the actual surface roughness, where the prediction error was $E_{step2} = 57.8\%$. In order to improve the surface roughness model, a proportional compensation was conducted (step 3). The testing data were applied to fit the proportional compensation and the data obtained through a design of experiment based on Taguchi's array $L - 9$ were used to validate the model. The adapted roughness model based on the proportional compensation improved the prediction error to $E_{step3} = 33.1\%$.

Although proportional compensation improved the model performance, a model re-training was conducted to increase model accuracy (step 4). The re-trained model had a prediction error of $E_{step3} = 22.6\%$. Fig. 5 (b) shows the ANN model predictions with proportional compensation and re-training.

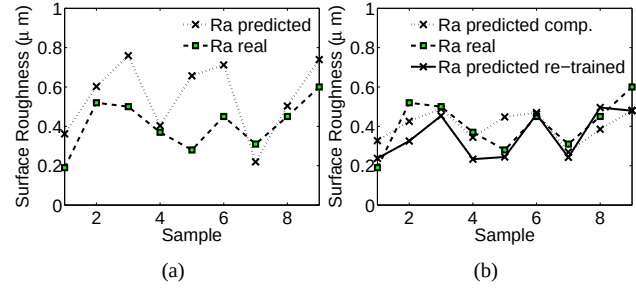


Figure 5. (a) Setup 2. Ra real versus ANN predictions from the ANN model trained in setup 1. (b) Ra real versus ANN predictions after proportional compensation (step 2) and re-training (step 4).

4.2.2. Adapting a strong cutting-tool geometry change (Setup 3). The experimental set-up was modified to setup 3, as it is shown in Fig. 3. Again, the methodology (b) was applied based on the initial ANN model. For this set-up, a fractional factorial experimental design 2^{3-1}_{III} was conducted to provide testing data (step 1). The statistical analysis results showed a $p - value = 0.079$ and $p - value = 0.0001$ for Levene's test and two-sample t -test respectively (step 2), indicating a change on mean error and therefore a change on the machining process. Fig. 6 (a) shows the ANN model predictions without adaption and the actual surface roughness, where the prediction error was $E_{step2} = 45.9\%$. In order to adapt the model, the proportional compensation was conducted with an L-9 Taguchi's array (step 3). However, this time the machining process change is more complex than the previous one, and the adapted roughness model based on setup 1 and the proportional compensation even greatly increased the prediction error ($E_{step3} = 80.1\%$). Therefore, the initial ANN model was re-trained using the testing and validating data obtained at step 1 and 3 (step 4). The re-trained ANN model decreased the prediction error to $E_{step3} = 32.8\%$, but it was still non-acceptable. As a result, it was concluded that this machining setup was far away from the initial one, and the surface roughness model had to be modelled from the scratch. Finally, following the methodology (a), the prediction error for setup 3 was decreased to $E_{step7} = 10.5\%$. Fig. 6 (b) shows the final ANN predictions results using both methodology (a) and (b).

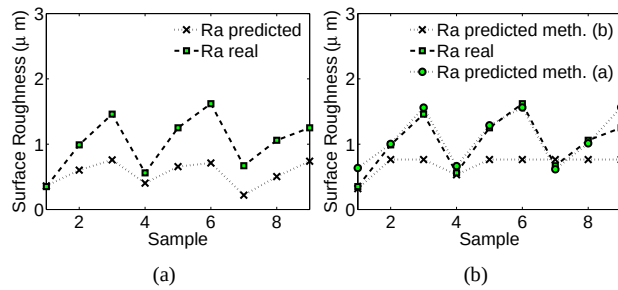


Figure 6. (a) Setup 3. Ra real versus ANN predictions from the ANN model trained in setup 1. (b) Ra real versus ANN predictions after conducting methodology (a) and (b).

4.3. Discussion

The case study shows how a cutting-tool geometry change from setup 1 to setup 2 and 3 increases the prediction error from 9.63% to 57.8% and 45.9% respectively. Therefore, surface roughness models can not be applied from one setup to another one without an adaption process in the milling operation studied. These errors seem to be caused by runout effects and built up edge effects. Setup 2 presents the best surface roughness ($0.2\mu\text{m} < R_a < 0.6\mu\text{m}$) since it is the smaller cutting-tool diameter with less number of inserts. However, setup 3 with 5 inserts and bigger cutting-tool diameter tends to produce a higher surface roughness ($0.4\mu\text{m} < R_a < 2\mu\text{m}$) due to runout effects among others. The adaptation procedure conducted in setup 2 through proportional compensation improves surface roughness prediction from 57.8% to 33.1%, so runout effects are partially compensated. After re-training the ANN model, other non-linear effects are partially incorporated into the model and the error prediction is reduced to 22.6%. In the case of setup 3 the adaptation procedure only decreased the prediction error to 32.8% due to the high cutting differences between setup 1 and setup 3. To improve the model in setup 3, the methodology to model surface roughness from the scratch is conducted and the prediction error is improved up to 10.5%.

5. Conclusions

In order to facilitate the application of empirical surface roughness models in industry, two methodologies have been proposed. A first methodology deals with modeling surface roughness from the scratch using few experimental runs. The methodology applies fractional factorial design of experiments to generate training and validating data respectively. An ANN model is learnt using the training data and GA to improve network convergence. In case ANN per-

formance is not adequate, the ANN model can be re-trained with additional experimental data through a Taguchi's array. This re-training process is conducted searching the optimal initial neural network weights through MADS algorithms instead of GA. The second methodology deals with adapting surface roughness models when a process change occurs. The methodology applies two possible adaptations. A first adaptation, called proportional compensation, is conducted if the process change produces a lineal error with respect to the previous model. A second adaption is conducted in case a proportional compensation is not accurate enough. This adaptation is based on re-training the ANN model, based on the training data and using a MADS algorithm for optimising the initial neural network weights. The methodologies were proved on a case study, where two cutting-tool geometry changes were analysed. The performance of the surface roughness model and the adapted model for each cutting-tool change were 9.63%, 22.6% and 10.5% respectively, whereas without adaption, the prediction error in setup 2 and 3 were 57.8% and 45.9% respectively.

References

- [1] Nist/sematech e-handbook of statistical methods. Electronic Citation, 2006.
- [2] N. R. Abburi and U. S. Dixit. A knowledge-based system for the prediction of surface roughness in turning process. *Robotics and Comp.-Int. Manuf.*, 22(4):363–372, 2006.
- [3] P. G. Benardos and G. C. Vosniakos. Prediction of surface roughness in cnc face milling using neural networks and taguchi's design of experiments. *Robotics and Computer-Integrated Manufacturing*, 18(5-6):343–354, 2002.
- [4] W. T. Chien and C. Y. Chou. The predictive model for machinability of 304 stainless steel. *Journal of Materials Processing Tech.*, 118(1-3):442–447, 2001.
- [5] S. S. Lee and J. C. Chen. On-line surface roughness recognition system using artificial neural networks system in turning operations. *International Journal of Advanced Manufacturing Technology*, 22(7-8):498–509, 2003.
- [6] T. Ozel and Y. Karpat. Predictive modeling of surface roughness and tool wear in hard turning using regression and NN. *Int. J. of Mach. T. & Manuf.*, 45(4-5):467–479, 2005.
- [7] K. A. Risbood, U. S. Dixit, and A. D. Sahasrabudhe. Prediction of surface roughness and dimensional deviation by measuring cutting forces and vibrations in turning process. *J. of Materials Processing Tech.*, 132(1-3):203–214, 2003.
- [8] Y. H. Tsai, J. C. Chen, and S. J. Lou. An in-process surface recognition system based on NN in end milling cutting operations. *Int. J. of Mach. T. & Manuf.*, 39(4):583–605, 1999.
- [9] C. A. van Luttervelt and J. Peng. Symbiosis of modelling and sensing to improve the accuracy of workpieces in small batch machining operations. *International Journal of Advanced Manufacturing Technology*, 15(10):699–710, 1999.
- [10] E. Westkamper, J. Pirron, and T. Schmidt. Simulation based on learning methods. *J. of Int. Manuf.*, 9(4):331–338, 1998.

Probabilistic Virtual Sensor for On-Line Viscosity Estimation

Pablo H. Ibargüengoytia, Alberto Reyes, Mario Huerta
Instituto de Investigaciones Eléctricas
Av. Reforma 113, Palmira
Cuernavaca, Mor., 62490, México, {pibar,areyes, mhuerta}@iie.org.mx
Jorge Hermosillo
Universidad Autónoma del Estado de Morelos, jhermosillo@uaem.mx

Abstract

Virtual sensors or software sensors are computer programs that estimate the value of difficult variables, using a model of the process and the measurements of the related variables. These difficult variables can be measurements in inaccessible places for hardware sensors, expensive instruments or instrumentation difficult to calibrate and maintain. Examples of variables are temperature measurements in turbines, emissions in chimneys, or viscosity of oil fuel. The estimation of the value of a variable utilizes a model and real readings of related variables. Several types of sensors can be built according to the type of model implemented. This paper presents a probabilistic viscosity sensor that utilizes probabilistic relations between the variables of the process. The model is based on Bayesian networks that can be learned using historical data and some expert advice. Experiments are presented where two data set are used. One is used for training the model and the other is used for validating the model. Promising conclusions are also presented.

1 Introduction

Real applications of computer programs require the larger possible set of physical variables for taking the optimal decisions. This includes some variables that are very difficult to read. Current trend is the use of virtual sensors, soft sensors or inferential models. They are computer programs that *estimate* the value of a variable given a model of the process involved and the measures of the related variables. Some examples for the reasons of the necessity of virtual sensors can be:

- measures are required in inaccessible places where sensors can not be installed, like the combustion chamber of gas turbines.

- physical sensors are difficult to install and operate, like emissions sensors in chimneys.
- hardware instruments are expensive and difficult to calibrate and maintain.
- redundancy with hardware sensors for fault detection and tolerance.

The basic idea is to create a model of the process and the information provided by the accessible variables. For example, temperature in the combustion chamber of a gas turbine may be inferred using the temperature of the escape gases, temperature of the turbine exterior and others [7]. As another example, the control emission monitoring system (CEMs) are expensive equipment that has to be installed at the top of chimney but also require calibration in short periods of time. Literature reports the use of predictive emission monitoring system (PEMs) as a common solution for contaminant emissions.

Different kind of virtual sensors exist according to the kind of models [5]. One approach used in traditional chemical processes is analytical. This includes the development of complex differential equations that related the virtual variable with other easier to read parameters and that are related in some way to this value [1]. However, this is sometimes very difficult to obtain. Also, any small change in the process may represent huge changes in the analytical models that relate these variables. Additionally, analytical models require the participation of high experimented experts of the process. This also, is difficult to find.

Computational intelligence methods have been used in this estimation. For example, neural networks, fuzzy logic and genetic algorithms are used in the estimation of one variable after a training phase. Sometimes, a combination of these methods are utilized in specific environments [4]. However, these combinations result in unique prototypes difficult to apply in similar problems.

Another approach consists in the use of artificial intelligence techniques in the development of virtual sensors. If the estimation is based on probabilistic relations of the variables, then Bayesian networks mechanism can be used. This mechanism includes robust and efficient automatic learning algorithms that provide the models given real data from the process. This approach requires the acquisition of all the variables while the process is operating at full range. This means that, obtaining and calibrating the costly instrument for a few days, it is possible to create a data base with all the information. Later, a probabilistic model can be built using one of the varieties of learning algorithms. Finally, estimation of the virtual sensor can be made through probability propagation once that the related values have been read.

This paper proposes and demonstrates the use of probabilistic reasoning, i.e., Bayesian networks, in the creation of virtual sensors. In a previous paper, it was shown the basic mechanism for constructing virtual sensors using Bayesian networks [7]. This paper presents the application of that technology in a real sensor for a real domain: A prototype was constructed and tested with viscosity of oil fuel in power plants.

This paper is organized as follows: we start describing the Bayesian network mechanism used traditionally in uncertainty management. Section 3 explains the application domain where this virtual sensor is developed. Then, we present our approach in the construction of probabilistic models that will be used to create the virtual sensors. Then, section 5 discusses the experiments reported of the oil viscosity virtual sensor. We finally conclude giving new directions of this work.

2 Probabilistic modeling using Bayesian networks

A Bayesian network (BN) is a graphical representation of dependencies and independencies of random variables for probabilistic reasoning in intelligent systems [9]. Fig. 1 depicts an example of a simplified BN representation of five variables, and their relationships. In a BN, each node represents a discrete random variable and each arc a probabilistic dependency. The variable at the end of a link is dependent on the variable at its origin. Thus, the following considerations were taken in the construction of the BN of Fig. 1. The viscosity V is caused by the temperature of the oil fuel tmp and the supply pressure Pr before the combustion. This pressure also causes changes in the flow of fuel into the boiler flw . In this example, variables tmp and V depend probabilistically of variable ext . This network can be taken as representing the joint probability distribution of

the variables ext , V , Pr , tmp and flw as:

$$P(ext, V, Pr, tmp, flw) = P(tmp | ext)P(Pr | ext) \quad (1) \\ P(V | tmp, Pr)P(flw | Pr)P(ext)$$

This equation is obtained by applying the chain rule and using the dependency information represented in the network. Notice that all nodes are probabilistically dependent on their parents, except the root node ext .

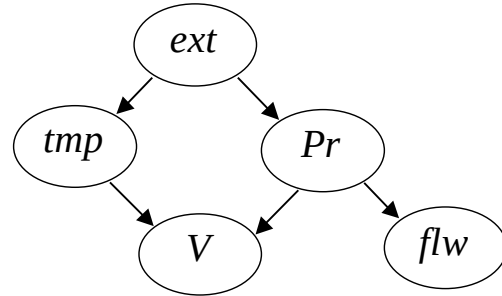


Figure 1. Example of a network representing the causal relation between variables.

Given a knowledge base represented as a probabilistic network, it can be used to reason about the consequences of specific input data, by what is called *probabilistic reasoning*. This consists in assigning a value to the input variables, and propagating their effect through the network to update the probability of the hypothesis variables. The updating of the certainty measures is consistent with probability theory, based on the application of Bayesian calculus and the dependencies represented in the network. For example, in the BN in Fig. 1, if ext and flw are measured and V is unknown, their effect can be propagated to obtain the posterior probability of V given ext and flw . For singly connected networks, such as trees or polytrees, there is an efficient algorithm for probability propagation [9]. It consists on propagating the effects of the known variables through the links, and combining them in each unknown variable. This can be done by local operations and a message passing mechanism, in a time that is linearly proportional to the diameter of the network. The more complete Bayesian network representation is multiply connected network as the example in Fig.1. For this kind of networks, there are alternative techniques for probability propagation, such as clustering, conditioning, and stochastic simulation [9].

Bayesian networks can be used to represent the dependency relations between the measurements, and obtain their posterior probabilities given the evidence of other measured variables. The next section briefly describes the problem that combustion processes face with the viscosity of oil fuel.

3 Viscosity of oil fuel

Viscosity is a property of the matter that offers a resistance to flow. Fossil oil needs to flow to the burner's boilers in order to heat water and produce steam. Fossil oil is provided to the electric generation plants from different sources and different qualities. This means that the viscosity of the oil fuel is almost an unknown variable.

The key question is: why is so important a viscosity sensor?

More than 45 % of the electricity produced in México comes from oil fuel based thermoelectric power plants [3]. Briefly, water is heated in huge boilers that produce saturated steam that feeds turbines that are coupled to electric generators. The calorific energy of the steam is transformed in mechanical work at the turbines and this work is transformed to electricity on the generators. While more gas is consumed in the boiler, more steam is produced and hence, more power is generated. This generation process is measured by an index called thermal regime. This index relates the Mega Watts produced by liter of oil burned.

To increase the thermal regime index, the combustion in the boiler is an important process to control. Usually, control systems regulate the aperture of the gas valve to feed more or less oil to the burners in the boiler. However, the optimal combustion depends on several factors. One of these factors is the oil atomization at the burners. If the droplet of oil is too big, only a small portion of it will be burned and the rest is expelled through contaminant smoke. If the droplet is too small, the mixture of fuel and air is incomplete and produces also contaminant residues and low combustion performance. Thus, in order to have a good oil atomization, an exact viscosity is required in the flow of oil to the burners. The viscosity changes mainly with the temperature. Thus, an optimal combustion control includes the determination of the viscosity of the input oil and its optimal heating, so the viscosity can be driven to the required value. This produces a good combustion that generates steam for power generation.

Measuring viscosity on-line is therefore essential in order to achieve good combustion. However, an on-line hardware fuel viscosity sensor is expensive. Moreover, maintenance costs are high since it has to be cleaned perfectly every short periods of time given the nature of the object: a dense flow of raw oil [10]. Since it is difficult and expensive to maintain a hardware viscosity sensor, a virtual sensor is naturally an attractive solution.

In practice today, technicians obtain a small sample of oil when it is supplied to the plant. This sample is analyzed in chemical laboratories to identify several chemical properties and the viscosity at current temperature. This is a process that takes more than one hour and that is repeated a few times during a month. Once the viscosity has been deter-

mined, the oil supply temperature is fixed at certain value, usually not the optimal one. For this project we had access to several analysis sessions so we could build up a database with the content of different components apparently related to viscosity. Some examples of these components are: water, carbon, hydrogen, oxygen, nitrogen, nickel, and sulphur. Table 1 displays the identified variables with their assigned ID.

Table 1. Variables obtained

ID	Variable	Units
μ_0	Viscosity in laboratory	<i>cSt</i>
T	temperature	$^{\circ}C$
H_2O	water	%
Sed	sediment	%
SWt	specific weight	<i>g/ml</i>
Ash	ash	%
CC	Conradson carbon	%
CV	calorific value	<i>kcal/g</i>
ASF	asphaltene	%
IT	inflammation temperature	$^{\circ}C$
S	sulfur	%
Fe	iron	<i>ppm</i>
Mg	magnesium	<i>ppm</i>
Na	sodium	<i>ppm</i>
Ni	nickel	<i>ppm</i>
V	vanadium	<i>ppm</i>
C	carbon	%
N	nitrogen	%
H	hydrogen	%

The units refer to centistokes (cSt), Celsius degrees ($^{\circ}C$), grams per milliliter (*g/ml*) percent in the sample (%), kilo calories per gram (*kcal/g*), and part per million (*ppm*). This information is utilized in the construction of the virtual sensor as described in the next section.

4 Constructing the virtual sensor

Having access to a database containing the results of hundreds of analysis, we can utilize the main advantage of the proposed method for virtual sensor, namely, automatic learning algorithms. Several learning algorithms have been developed for constructing the network structure and the numerical parameters. Some algorithms work autonomous and some allow human expert participation. For example, the K2 algorithm [2], receives as input, a table with variables as columns, and all the cases as rows. Thus, the order of the variables in the table, indicates the known relation between all the variables. Additionally, K2 limits the number of parents that a single node may have.

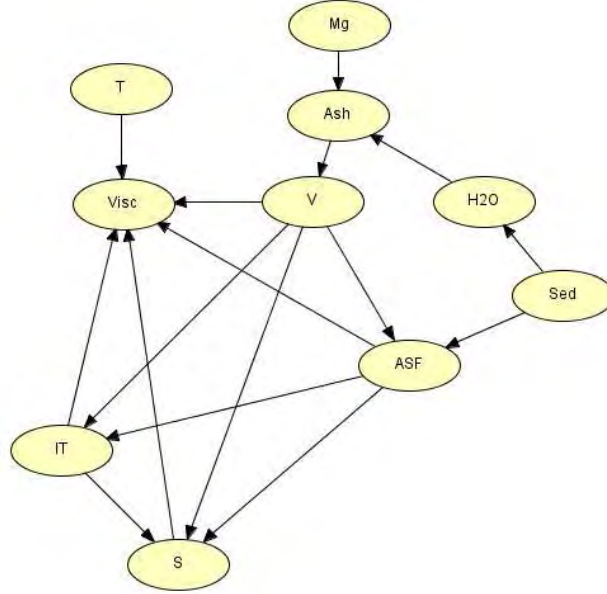


Figure 2. Probabilistic model obtained with *PC* automatic learning algorithm.

The *PC* algorithm available in the *Hugin* software package [6] also receives a table with variables and cases. However, this algorithm questions the user for known relationships between certain variables that may not be discovered in the data set. In this step, we know that some of the most important relations to the viscosity are the temperature of course, plus sulfur and asphalthe. Thus, completing the algorithm, the network of Fig. 2 is obtained.

This network structure corresponds to the following expression of joint probability function.

$$\begin{aligned}
 P(Vars) = & P(T)P(Sed)P(Mg)P(Ash | Mg, H2O) \\
 & P(H2O | Sed)P(V | Ash)P(ASF | V, Sed) \\
 & P(IT | V, ASF)P(S | IT, V, ASF) \\
 & P(Visc | V, ASF, S, IT, T)
 \end{aligned}$$

where *Vars* represent all variables. Notice that many of the chemical components were unnecessary according to the learning algorithm. This means that there exists no probabilistic relation between these elements and others.

In order to validate the performance of the virtual sensor, we utilized measurements of chemical analysis from different locations at different times. Table 2 shows some examples of the experiments carried out with the validation data set. The first column indicates the parameter measured and supplied to the model as evidence. The following four columns specify four types of oil fuel measured at four different locations.

Data from each column was supplied to the model as evidence. The first row corresponds to the viscosity measured

at the laboratory. The following ten rows are provided as evidence case by case, i.e., row by row. Since Bayesian networks process discrete value variables (or nodes), the continuous value quantities have to be discretized. The viscosity for example, was discretized in ten intervals. Thus, in the first experiment shown (column of location 1), nodes were instantiated with the value corresponding to the discretization of every variable. Probability propagation is executed, and the posterior probability of the viscosity is calculated. Figure 3 shows the graph of the posterior probability of viscosity, given the evidence. Vertical axis represent the posterior probability with lines every 20%. Horizontal axis represent the interval (states *S*) of the viscosity after the discretization process. Notice that in some cases, evidence is missing, i.e., information is incomplete.

In location 1 and 2 experiment2, the posterior propagation indicates that there is a 100% probability that the viscosity is in the interval of state *S*₁. Location 3 and 4 resulted with wider distributions but the real value always corresponds to the higher probability interval. In all cases, the estimation resulted correct.

5 Discussion

The preliminary experiments described in Fig. 3 look promissory. Most of the examples have accuracy higher than 90%. In the rest of the cases, the posterior probability distribution was wider. However, the real value coincides with the interval of higher probability. One aspect to consider is the lost of precision caused by the discretiza-

Table 2. Variables measured and supplied to the model

Variable/ Experiment	Location 1	Location 2	Location 3	Location 4
Viscosity	49.6	523	462	526
H_2O	0.6	0.1	0.1	0.1
Sed	0.4	-	-	-
ASF	11	16.18	15.98	16
S	-	3.64	3.55	4.16
C	85.2	84.77	84.35	84.06
Ash	0.04	0.081	0.07	0.07
H	11.2	10.56	11.27	9.66
N	-	0.32	0.3	0.36
CV	10.112	10.142	10.219	10.024
IT	-	74	80	78

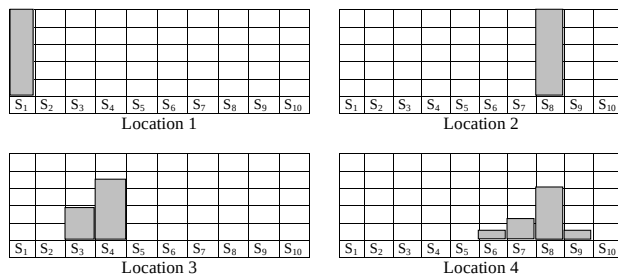


Figure 3. Probability of the viscosity given the evidence.

tion process. Five intervals may be few for variables with large variation, but the computational cost is maintained low. If higher precision is required, better discretization approaches can be utilized, but the computational cost would increase. This cost is with respect to memory storage and the propagation time. In a Bayesian network, a conditional probability table is defined with all the values of a node, given all the combination of the values of its parents. Thus, a node with 5 possible values, and 8 parents with 5 values each, represents a table with 1,953,125 entries. In this example, node *Viscosity* has 5 parents (see Fig. 2).

For a better performance, it is advisable to prove with different discretizations schemes or more intervals. With this, models result in more exact representation of the signals behavior in the range of operation of the process.

The solution of this problem is to keep the probabilistic model with the lowest interconnectivity possible. The problem is not the number of nodes but the number of arcs between them. The simplest case is to learn a tree structure, i.e., a network where all the nodes can have at most one parent. In this case, discretization can be carried out with larger number of intervals. Notice that the structure learned in this experiment, shown in Fig. 2 is very interconnected,

so the number of intervals had to be maintained low. However, in the present days, dealing with models that utilize mega bytes of memory is a common task.

6 Conclusions and future work

This paper has shown the construction of virtual sensors using probabilistic reasoning. Borrowing a real sensor, readings can be collected while the process is being executed. Later, a model is learned automatically using the PC algorithm. Finally, the virtual sensor is executed on-line, estimating the corresponding value. This paper utilized real data from chemical analysis of oil fuel. Even if a single hardware viscosity sensor is needed for obtaining historical data, the virtual sensor can be replicated in all the locations where similar combustion processes take place.

Future work includes the utilization of alternative mechanisms for making probabilistic models. In the framework of Bayesian Programming [8], the notion of probability is not only used in its statistical sense, where we consider the frequency of occurrence of a variable, but rather the state of knowledge including equations or empirical knowledge that the expert may have.

References

- [1] D. A.W. and L. J.H. Building inferential prediction models of batch processing using subspace identification. *J. Proc. Control*, 13:397–406, 2003.
- [2] G. Cooper and E. Herskovits. A bayesian method for the induction of probabilistic networks from data. *Machine Learning*, 9(4):309–348, 1992.
- [3] C. F. de Electricidad. Generación termoeléctrica. <http://www.cfe.gob.mx/es/LaEmpresa/generacionelectricidad/termoele> 2008.
- [4] H. et. al. Analysis of initial conditions for polymerization reaction using fuzzy neural network and genetic algorithm. *Com. Chem. Eng.*, 27:1011–1019, 2003.

- [5] L. Fortuna, S. Graziani, A. Rizzo, and M. G. Xibilia. *Soft Sensors for Monitoring and Control of Industrial Processes*. Advances in Industrial Control. Springer, Glasgow, Scotland, U.K., 2007.
- [6] S. H. *Constrained-Based Structural Learning in Bayesian Networks Using Finite Data Sets*. Ph.d. thesis, Institut für der Informatik der Technischen Universität München, München, Germany, 2001.
- [7] P. H. Ibargüengoytia and A. Reyes. Constructing virtual sensors using probabilistic reasoning. In A. Gelbukh and C. Reyes-Garcia, editors, *Advances in Artificial Intelligence - MICAI 2006, LNAI 4293*, pages 218–226, Berlin Heidelberg, 2006. Springer-Verlag.
- [8] O. Labeltel, P. Bessiere, J. Diard, and E. Mazer. Bayesian robot programming. *Auton. Robots*, 16(1):49–79, 2004.
- [9] J. Pearl. *Probabilistic reasoning in intelligent systems: networks of plausible inference*. Morgan Kaufmann, San Francisco, CA., 1988.
- [10] A. I. S.A. Medida en línea de viscosidad en blending de fuel oil. Technical report, Aplein Ingenieros S.A., 2006.

Improving the accuracy and reliability of wireless location systems: a case study

Russell Beale
Advanced Interaction Group
School of Computer Science
University of Birmingham
Edgbaston, Birmingham
B15 2TT, UK
R.Beale@cs.bham.ac.uk

Abstract

Wireless location is of increasing importance in context-aware systems. We have developed a wireless location positioning system that can use multiple algorithms for providing estimates of device location, and we present a case study of the system in use with a building. We show that combining two algorithms allows us to provide improved accuracy over one algorithm alone, and draw some general purpose conclusions regarding location determination in real-world environments.

1. Introduction

The widespread deployment of wireless local area networks and the increasing popularity of light-weight mobile computing devices has lead to an increased interest in location-aware applications and services [1, 2, 3]. Location-aware applications make use of the user's location in the network, and a number of practical benefits can be obtained by being able to determine the location of the user in a wireless network. Service and infrastructure uses, such as wide-area security, inventory control, emergency response, and event management are potential applications, as are automatic call forwarding to the user based on location, helping shoppers through stores based on their locations, and assisted tours in libraries and museums [4, 5]. Location-specific information also allows us to achieve user interface improvements through adaptive menus and navigation that save time and are more convenient and compact. Adaptive menus improve the productivity and interactivity of user-interfaces by providing the user only a few menu items to click and less data to input depending on location of the user. Network performance improvement is another area that can benefit by knowing the location of the wireless device in the network; this can be derived from pre-emptive catching of content based on upon better spatial

understanding of user location. Location information can also be used to improve system operations of networks by including the spatial distribution of users and assets for communications (e.g., routing) and data storage organization. Another driving force for wireless network user location determination is that it often exists in spaces where other traditional location determination systems such as Global Positioning System (GPS) fail, or at least require specialized hardware to work which prove to be expensive for implementation [5].

2. Related work

The field of location-aware computing has attracted enormous interest amongst researchers. Different techniques for user location determination are being proposed and systems built to tackle the problem of determining and tracking user position, but currently there are no standard frameworks for building wireless LAN location determination systems [6].

Ekahau, a prominent commercial system, is a location determination system based on over 10 years of research by the Complex Systems Computation Group at the University of Helsinki. The Ekahau system has been specifically developed to provide GPS-like positioning inside buildings and it works over different Wireless LAN standards (802.11a,b,g), using an fingerprinting technique [7]. Research systems that provide location-based services over wireless networks include LOCATOR[8], HORUS[9], and RADAR[10, 11].

There are three principal techniques for wireless location: proximity, triangulation, and scene analysis. Proximity determines the location of the nearest wireless access point and gives the user location as near that: it only requires one access point but is relatively coarse. Triangulation uses distances (lateration) or angles (angulation) of known signals to compute the location of the mobile device, and provides a more accurate determination but is

affected by environmental factors such as fading, multipath interference and other interference effects that add error. Scene analysis uses features observed from a particular vantage point to infer the location of objects, and is most commonly found in fingerprinting approaches, in which an offline learning phase collects location fingerprints by conducting a site survey of received signal strengths from multiple access points. In the online, location determination phase, clients collect signal strength data from the access points and compare the values to those collected in the first phase to determine their position.

We were interested in whether an intelligent combination of approaches would provide a more accurate or reliable estimate of position than relying on one approach on its own.

3. System

We created a middleware software layer between the wireless receiver and the operating system to allow us to capture and process raw signal strength values and access point identifiers, and used this to provide the input into the positioning algorithms. For deployment in positioning systems, this provides a simple position value that can be accessed by other software without knowing the details of how it is calculated or created, and can be interfaced with other location providing systems such as GPS. It therefore provides a flexible, adaptable positioning solution.

4. Experiment

Our experimental aim was to improve the accuracy of the calculated location by combining more than one algorithm so that we could gain additional precision when possible and still retain the breadth of coverage in poorer signal areas. To achieve this we combined two existing algorithms, each described below.

4.1. Weighted nearest neighbour Algorithm

The weighted distance, L_p , between a measured received signal strength for N access points $[X_1 X_2 \dots X_n]$ and a fingerprint value in the database $[X'_1 X'_2 \dots X'_n]$ is

$$L_p = \frac{1}{N} \left(\sum_{i=1}^N \frac{1}{w_i} |x_i - x'_i|^p \right)^{1/p}$$

We choose the location to be the measured fingerprint point closest to the measured one, using either the Manhattan ($p=1$) or Euclidean ($p=2$)

distance metrics, with a default weight $w_i=1$ for all entries. Alternatively, we can choose the M closest ones and estimate the position based on the average of these points. In this case the weight w_i can be used to bias the values if some measure of reliability can be used (e.g. signal strength).

4.2. Maximum likelihood algorithm

The Maximum Likelihood method, also like the Nearest Neighbour approach, works in two phases. In the Offline training phase, a radio map is built which tabulates the sampled signal strength values received from access points at selected locations. The radio map is considered as a conditional probability distribution consisting of finite location spaces and a finite observation space.

During the online phase, the location determination phase, a posterior probability of real-time signal strength samples is computed using Bayes' rule

$$P(\bar{S}/l) = \prod_{i=1}^n P(s_i/l)$$

i.e. the probability of a particular location l

given a received signal $\bar{S}$ can be computed from the previously computed probabilistic map of existing signal strengths s_i observed at a particular location. The location with the maximum posterior probability is estimated to be the location of the user (more precisely, the wireless LAN device).

4.3. Training

Initial experiments with Netstumbler, an application that measures the signal strength of access nodes in a wireless network, showed that in a typical office building, signal strength varied between points >1m apart. During training, we therefore did the fingerprinting phase using grid points notionally spaced 1 meter apart. Note that a fully regular grid could not followed due to presence of obstacles such as walls, stationary furniture and other objects. The origin of the coordinate system (0,0) was designated to be the left bottom corner of the floor plan and the third coordinate for each floor, making a three dimension coordinate system (x,y,z). In our trial building, testing was performed on the ground floor, which had few wireless access points, and on the first and second floors, which have a large number of access points compared to the ground floor. Our aim was to study how the location determination system accuracy varied in different radio signal coverage using the two algorithms individually or together.

During training at each grid point, the signal strength was measured facing east, west, north and south, and then we stored the averaged signal strength value in the radio map. The reason for this was to mitigate the effect of signal variations with changing orientation, which were observed during preliminary tests to be at most 5dB. This could have an impact during the location determination phase because the orientation of the user cannot be assumed and it is likely to change. We collected a total of 50 scans for each floor. The test was performed on different hours of the day, early morning, afternoon and late evening to investigate the variation of signal strengths with varying environmental conditions such as temperature, movements of people etc. and their effects on location determination.

5. Experimental test results

For the building investigated, the number of access points covering a particular grid point was observed to vary in an unpredictable manner. The accuracy of the location-providing application increases with the number of audible access points and size of training data, which gives more samples of signal strength for comparison. The first and second floor have large number of access points compared to the two lower ground floors - most positions were covered by at least two access points and some positions with three access points audible at a position – see Figure 1. Very few positions on the first and second floor were simultaneously covered by four access points.



Fig. 1. Schematic diagram showing wireless access point location and expected coverage – second floor of the trial building

5.1. Independent algorithm results

The two algorithms were first tested independently to determine their accuracy before deploying them together in the system prototype.

The presence of other users and changes in furniture or its location all impacted the accuracy of the system, and summarizing the results is not trivial.

We have decided to provide the results in terms of two values to describe each result, rather than a single accuracy figure and associated standard deviation. This allows us to mitigate for extreme events and to provide more of a picture of the systems performance. There are two factors we are particularly interested in – getting an *accurate* position, and getting a *reliable* one. For accuracy, we want to sensibly calculate where the user is most likely to be. For reliability, we want to know the area within which we can reliably find them – thus the reliability figures is greater than or equal to the accuracy one. We chose to use the 50th-percentile and 80th-percentile values: the 50th-percentile value is equivalent to the modal value, and can be seen as giving the variability of the most *accurate* position. The 80th-percentile value gives a value for the precision of a *reliable* position. The 80th percentile value provides an accuracy figure, and within that figure of the current location the user will actually be, 8 times out of 10 – it trades off ultimate accuracy for improved certainty.

During the online location determination phase, the system was used at randomly chosen points. Using 20 test samples, the weighted nearest neighbour algorithm achieved 50th-percentile and 80th-percentile location estimation accuracy of 1.4 meters and 2.3 meters.

In comparison, the maximum likelihood algorithm achieved 50th-percentile and 80th-percentile location estimation accuracy of 1.6 meters and 2.2 meters respectively.

The result showed that with averaging of received signal strengths, the weighted nearest neighbour algorithm was more accurate in most locations where there was good coverage by access points. The maximum likelihood method, though less accurate compared to the weighted nearest neighbour algorithm, performed better in areas where there was poor coverage of radio signals from the access point; this can be attributed to its ability to consider prior known location in estimating the current user location.

5.2. Combination algorithm

Combining the algorithms requires the use to appropriate heuristical techniques. Both algorithms can in principle be analyzed to determine the potential accuracy of their location; for example, in the maximum likelihood method, the presence of many other locations with almost as much probability as the chosen one would suggest a less reliable answer. Our initial plan was to use this to drive the weighting of the algorithms, but empirical observations demonstrated that this approach was unreliable and relatively complex to compute, and a simpler approach would be required. To combine our algorithm's estimates, we followed a simple weighting heuristic: in areas with better coverage of

radio signals (more access points audible and stronger signal strength values observed), the weighted nearest neighbour algorithm is given higher priority in estimating user locations. Where there is poor coverage, the location estimated by the maximum likelihood algorithm is given higher priority.

By combining the estimates of the algorithms using a weighted function in this way, we achieved a 50th-percentile and 80th-percentile location estimation accuracy of 1.5 meters and 2.2 meters respectively.

The combined approach by using both algorithms slightly increases the reliability of the location determination system although the accuracy remains about the same.

6. Discussion and Lessons Learnt

The results of the location determination phase varied from one location to another and from one floor to another, depending on access point coverage, with better-covered locations yielding high accuracy. The following were identified as factors that impacted the accuracy of the location estimation:

- Size of the training data and distance between points where measurements are recorded were observed to be the main contributing factor to the accuracy of the location determination system *regardless* of the location determination technique used.
- The weighted nearest neighbour method fared well compared to maximum likelihood method especially in areas where there was a decent coverage of the radio signal. On the other hand for areas with poor radio signal, the maximum likelihood method was occasionally better than the weighed nearest neighbour method. However, this does not significantly increase the accuracy of the system, but it makes the location determination system more robust and reliable to changing radio signal coverage.
- In large rooms, there was marginal or no change at all in signal strengths in some cases at the grid spacing of 1m. Also, most large rooms in our building have only one access point audible, which makes it even worse for location determination.
- The placement of access points in the building is designed only to achieve maximum coverage. They are not optimised for location determination systems; some places are not covered at all or covered by only one access point.
- Multipath interference and refraction of wireless signals on walls and other objects in the building occurs often. This could potentially be mitigated by proper path-loss modelling of wireless signals and careful choosing of points for training.
- Although there were few cases of errors in multi-floor location determination (i.e. locating the user on the wrong level of the building), there is a potential for such errors if the access points on other floors have strong signals observed in adjacent floors. We were able to correct for this by retraining the system on those locations in which there was an initial issue, which largely solved the problem. These tended to occur at the outside walls of the buildings. The same approach can be applied in different environments with carefully chosen training points.
- Other minor factors like number of people in the building or room, doors closed or open, atmospheric condition changes etc. affect radio signals in a difficult to predict manner. For example, the same point exhibited minor changes in signal strength in morning and afternoon using the same configuration and hardware.

The results we obtained during testing would be reasonably adequate for use in most location-aware applications because the magnitude of the errors are significantly less than the size of the smallest rooms.

One useful characteristic was observed: in some locations where the noise to signal ratio (a measure of the quality of service) was low to the extent of not being able to get network service, the location determination system was still able to work successfully. This can be used to alert users when they move to areas where there is poor coverage of radio signal.

It was also observed that increasing the number of grid points by reducing spacing between the points, averaging of measurements taken at each point with different orientations, and training the system during the time which it is likely to be used enhanced the accuracy of the system for locations which are well covered by radio signals (in the first and second floor of experimental test bed). However, this resulted into a slight increase of the latency in estimating location due to the increase in the number of samples in the radio map that needs to be searched, and it also increased the system calibration time as more samples need to be taken. For locations which have poor coverage of radio signals, reducing the grid point distance did not result in any noticeable change in the observed signal strength values, which resulted into the same accuracy in the system but with more

time spent on calibration (in the lower and ground floor of experimental test bed).

We appreciate that the weighting factors used in this system are determined heuristically, but when calculated for the building they can then be set and left unchanged for application usage. The approach therefore does transfer well to other buildings, though the same accuracy and reliability figures should not be expected, since the propagation of signals and placement of access points vary considerably. In particular, the lessons learned are transferable: for locating people using wireless networks, as well as simply achieving wireless coverage in the building it is helpful to have multiple access points audible in any one location; daily variations including movement of people make it impossible to achieve ultimate accuracy in real-world situations; adding training points in areas of error improves accuracy.

7. Conclusion

Combining location estimation algorithms can provide increased reliability of position determination, though it has little impact on the accuracy of the estimate. Combining them using a weighted sum with the weights set heuristically based on signal strength and access points audible provide an effective real-world solution. In areas of strong signal coverage, the weighted nearest neighbour algorithm performed better than the maximum likelihood method, but this was reversed in areas of weak coverage: combining the algorithms gives a slightly more reliable result. Accuracy is most affected by the number of access points visible at any location, a consideration not often at the forefront of decision-making in the positioning of access points.

8. References

- [1] A. Dix, T. Rodden, N. Davies, J. Trevor, A. Friday, and K. Palfreyman, "Exploiting space and location as a design framework for interactive mobile systems," *ACM Transactions on Computer-Human Interaction*, vol. 7, 2000.
- [2] G. Giaglis, A. Pateli, K. Fouskas, P. Kourouthanassis, and A. Tsamakos, "On the Potential Use of Mobile Positioning Technologies in Indoor Environments," presented at 15th Bled Electronic Commerce Conference. e-Reality: Constructing the e-Economy, 2002.
- [3] S. Wang, M. Green, and M. Malkawi, "Mobile positioning technologies and location services," presented at Radio and Wireless Conference, 2002. RAWCON 2002. IEEE, 2002.
- [4] P. Lonsdale and R. Beale, "Using context awareness to enhance visitor engagement in a gallery space," presented at Proceedings of HCI 2005, Edinburgh, 2005.
- [5] J. Hightower and G. Borriello, "Location systems for ubiquitous computing," *Computer*, vol. 34, pp. 57-66, 2001.
- [6] P. Prasithsangaree, P. Krishnamurthy, and P. K. Chrysanthis, "On Indoor Position Location with Wireless LANs," presented at The Thirteenth IEEE International Symposium on Personal, Indoor and Mobile Radio Communications (PIMRC 2002), Lisbon, Portugal, 2002.
- [7] Ekahau, "Website," vol. 2005.
- [8] A. Agiwal, P. Khandpur, and H. Saran, "LOCATOR - Location Estimation System for Wireless LANs," presented at Proceedings of the Second ACM International Workshop on Wireless Mobile Applications and Services on WLAN hotspots (WMASH'04), Philadelphia, Pennsylvania, US, 2004.
- [9] M. Youssef and A. Agrawala, "Handling Sample Correlation s in the Horus System," presented at IEEE Infocom, Hong Kong, 2004.
- [10] P. Bahl and V. Padmanabhan, "RADAR: an in-building RF-based user location and tracking system," presented at INFOCOM 2000. Nineteenth Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings. IEEE, 2000.
- [11] P. Bahl, V. Padmanabhan, and A. Balachandra, "A Software System for Locating Mobile Users: Design, Evaluation, and Lessons."

Classification based on fuzzy inference systems for artificial habitat quality in shrimp farming

J. J. Carbajal¹, L. P. Sánchez¹

¹Department of Real Time Systems and Modeling, Centre of Computer Researches -IPN, Mexico D.F., Mexico
Phone (55) 57296000 E-mail: juancarvajal@sagitario.cic.ipn.mx

Abstract

Nowadays, the methods based on fuzzy inference systems (FIS) have demonstrated to be useful in the treatment of biological problems. An ecological abstract model for classifying artificial habitat quality in shrimp farming has been developed, based on a water quality index calculated with fuzzy reasoning. The potential application of the fuzzy index has been tested with a case of study proving the importance of the artificial intelligence in this area. The results show a good response obtaining four classifications of the status of the water quality; excellent, good, regular or poor. Therefore, this model emerges as a suitable and alternative tool to be used in the effective treatment of the water management in shrimp aquaculture.

Keywords — Classification, fuzzy logic, aquaculture

1. Introduction

The water quality in oceanic research is a problem that affects daily the activities of many people that practice fishing activities. The quantity of biological data is increasing each day and it is needed to create models and use some functional features.

The shrimp farming is an important economical activity in many countries. The shrimp production is determined by two main factors: 1) the capacity of maturing in the organisms and 2) the capacity of the environment [1], [2].

There are variables that have more impact into the habitat and they can affect the growing and the surviving of the organism [2], [3]. These variables have the most part of information and the water quality index could be evaluated only analyzing the patterns of their behaviors in the habitat. In general all the environmental variables have different ranges of measurement, each of them represents a specific situation in the habitat (Table 1) and the combination among them could disestablish the ecosystem, however, controlling the environment maintain equilibrated the biodiversity [4].

2. Scientific Problems and Hypothesis

2.1 Environmental Problems

There are not methodologies that asses the water quality giving an index about its status.. Some standards have been established as the NOM-001-ECOL-1996 (maximum limits

of pollutants in water discharges on coastal water in Mexico) and the CE-CCA-001/89 (water quality criteria) [2], [3]. However, these standards do not establish an index of the degradation status in fresh water or coastal water. International standards has been created giving a solution in this area as the ACA and NSF, however, the methodologies given by them can be only applied by fresh water bodies [5], [6]. The Canadian Council of Ministers of the Environment proposes a method that can be used in coastal water, it is based on calculate the number of failed value tests in a set of environmental variables, obtaining a water quality index (CWQI) [7]. However the ACA, NSF and CWQI indices have a number of weak points, where the lack of a reasoning process in the classification of environmental patterns is the main problem.

In general, the environmental variables have nonlinear relations, which have been observed and proven experimentally, the equations that represent them, have been formulated, which is very hard to do [1], [2], [8].

2.2 Hypothesis.

Fuzzy logic is based on a natural language and it is used for creating mathematical approximations, for modeling non-linear functions and for building expert systems. These advantages have been used to resolve environmental problems related with artificial shrimp habitats. Since there are not methodologies for ocean water quality in shrimp farming, we hypothesize that a FIS can provide an indicator of the environmental conditions in an artificial shrimp habitat [9].

2. Methodology

2.1 Environmental behaviors

Physical-chemical variables contain particular behaviors, and their combinations can disestablish the artificial habitat. In order to classify their negative impact into the shrimp habitat it is important to define the maximum limits that allow having optimal conditions [1], [3]. A particular variable is classified in various levels, each one has a different impact into the habitat, and the Table 2, shows the physical-chemical variables and their classification levels.

When the value of a test is close to the limit, the level classification will deteriorate its status. This behavior can be expressed using the following concepts:

- Tolerance: Maximum value when the desired level analyzed is recognized
- Limit: Maximum value when the desired level does not deteriorate its classification condition.

2.2 Classifications status

The environment always changes and the physical-chemical variables can disestablish the habitat. The classification of the habitat can be expressed in the following status:

1. Excellent: All physical-chemical variables are in the optimal ranges.
2. Good: One variable is out of the optimal level.
3. Regular: some of the variables are out of the optimal level or one variable of high impact is in a problematic level, however this do not represents an impact highly negative into the pond.
4. Poor: the combination of the variables or the level of high impact could generate a lethal problem for the shrimp.

2.3 Data Collection

Data collection was made on the farming shrimp ponds from Sonora, Mexico. Almost 9000 measurements were obtained in a farming period using conventional techniques, and it represents the 90% of time of the period. For each pond, the physical-chemical variables measured were pH, temperature, salinity, and dissolved oxygen.

Table 1. Ranges of classification of the environmental variables.

Variables	Hyp Acid	Low	Normal	High	Alk
Temp (°C)	-----	0 - 23	23 - 30	31 - ∞	-----
Salt (mg/L)	-----	0 - 15	15 - 25	25 - ∞	-----
DO (mg/L)	0 - 3	3 - 6	6 - 10	10 - ∞	-----
PH	0 - 4	4 - 6.5	6.5 - 9	9 - 10	10 - 11

Physical-chemical variables were measured in a frequency of 15 minutes with an error of 0.25%. While the shrimp farming is in process, the parameters can suffer some variations because the weather (Fig. 1). Due these variations, the variables are always in change and the combination of the whole set could generate a situation in the habitat [10]; the Fig. 1 shows this process.

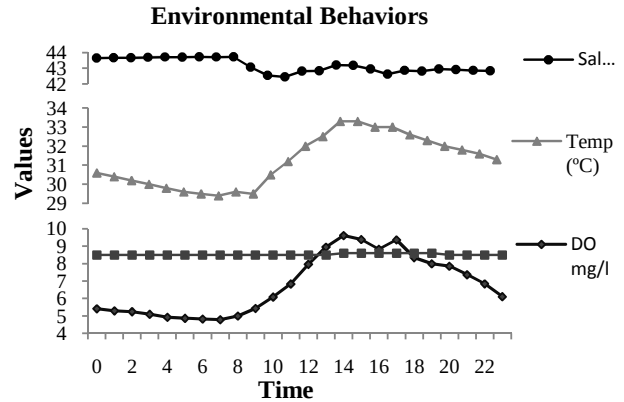


Fig. 1. Measurements of the environmental variables in one day in a shrimp pond in a farming period. Each variable shows different changes that generates different situations into the habitat.

2.4 Fuzzy Inference Systems for WQI (FWQI)

The Fuzzy inference systems (FIS) theory was applied in this study providing a non-linear relationship between input sets (Physical-chemical variables) and output set (Water Quality Index) [9], [4]. A set of four variables were used in the FIS: temperature, dissolved oxygen, salinity and pH (Fig. 2).

2.5 Physical-chemical Bioindicators

Measurements of physical-chemical variables are used as an input vector, however before to be analyzed by the biological rules, it is needed to transform the real values in fuzzy inputs, that process is known as fuzzification and it is made developing membership functions [9].

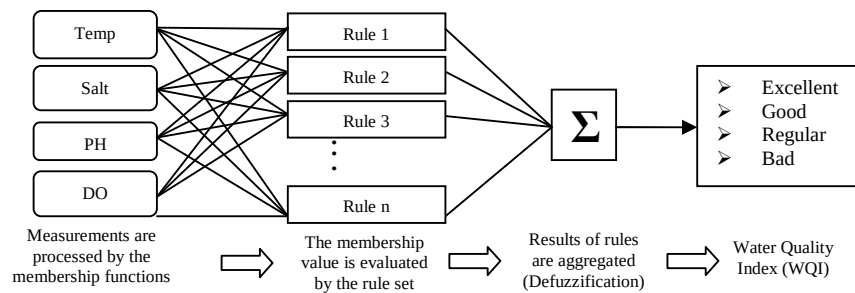


Fig. 2. Architecture of the Fuzzy Inference System applied to the water quality problem in shrimp farms.

A Bioindicator is a membership function, and it describes in a [0, 1] range how a variable belongs to a desired level. A Bioindicator depends of the physical-chemical variable analyzed. Following the ranges, tolerances and limits of the variables, it is possible determine four bioindicators for the farming of the p. vannamei shrimp, for dissolved oxygen we chosen “anoxia”, “low” and “normal”, for the temperature and salinity variables we chosen “low”, “normal” and “high”, and for the pH variable we chosen “acid”, “low”, “normal”, “high”, and “alkaline”. For the output fuzzy set we chosen “poor”, “regular”, “good” and “excellent”.

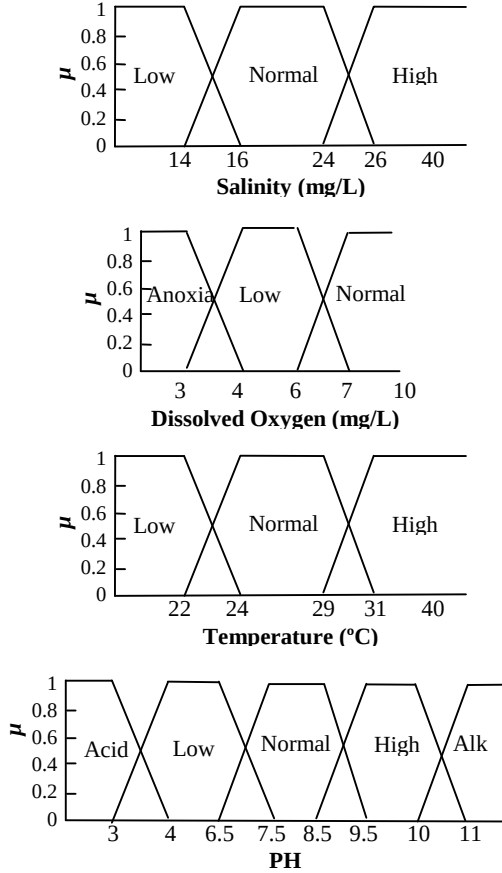


Fig. 3. Bioindicators of temperature, salinity, dissolved oxygen and pH.

2.6 Water Quality Bioindicator

The output membership function is the water quality bioindicator (WQI). The output of the FIS is expressed in a [0, 1] range, where 0 means “poor” and 1 “excellent”, the “good” and “regular” status are involved within the range, this is represented in the Fig. 4.

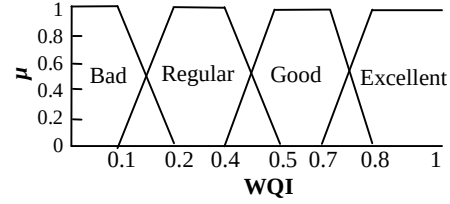


Fig. 4. Bioindicator for WQI

2.4 Reasoning Process

The reasoning process is the core of the inference system, due that it establishes the criteria that allow having a good performance in the assessment of the environment. This process is based in a rule set that is built with physical-chemical knowledge [9], [4].

There are some expressions that are frequently used by experts in water quality, that expressions will be helpful for the construction of the FIS. This kind of expressions built the fuzzy language of the FIS and it is represented as follows:

Rule 1: If Temp is normal and Salt is normal and pH is normal y DO is normal then WQI is Excellent

Rule 2: If Temp is normal and Salt is normal and pH is normal y DO is low then WQI is Good

The size of the set rule depends of the number of rules that are involved in the environment; a total of 51 rules have been used in this case. When the final membership function is obtained $\mu_{out}(x)$, the gravity center is calculated using the centroid method equation [9]:

$$WQI = \frac{\int x \mu_{out}(x) dx}{\int \mu_{out}(x) dx} \quad (1)$$

If we assume that it is necessary to evaluate the WQI in a shrimp pond using the rule 1 and 2, having the next fq vector:

$$\vec{v} = [temp, Sal, pH, DO] = [25, 20, 0.8, 6.3] \quad (2)$$

Using the membership functions proposed in the Fig. 4 and 5, for “R1” and “R2” we can calculate:

$$R_1: \mu_{out}(x) = \min\{\mu_{Temp-n}(x), \mu_{Salt-n}(x), \mu_{pH-n}(x), \mu_{DO-n}(x)\} \\ \mu_{out}(x) = \min\{1, 1, 1, 0.3\} = 0.3$$

$$R_2: \mu_{out}(x) = \min\{\mu_{Temp-n}(x), \mu_{Salt-n}(x), \mu_{pH-n}(x), \mu_{DO-l}(x)\} \\ \mu_{out}(x) = \min\{1, 1, 1, 0.7\} = 0.7$$

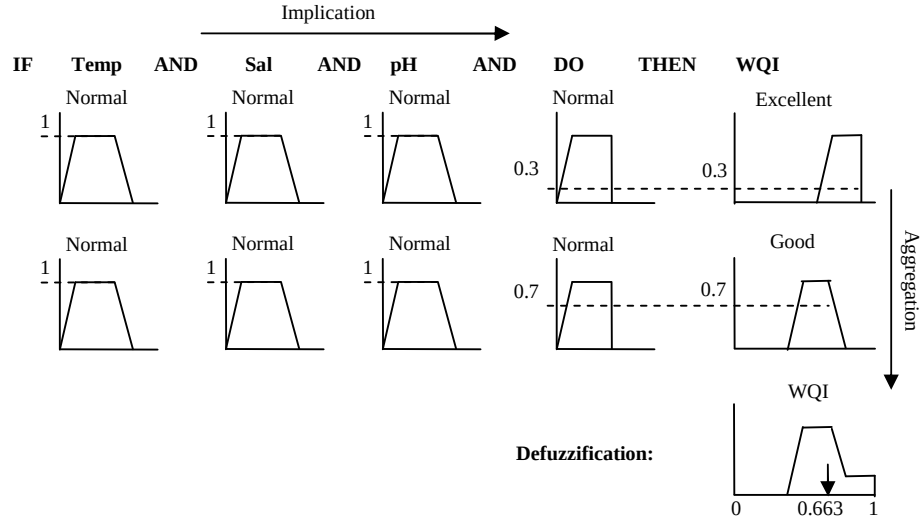


Fig. 5. The final membership functions are obtained using the aggregation of the fuzzy outputs.

Where n is normal, l is low and μ_{out} is the membership value calculated in R1 and R2. Calculating the aggregation functions:

$$\mu_{R1}(x) = \min\{\mu_{out1}(x), \mu_{excellent}(x)\} = \{0.3, \mu_{excellent}(x)\} = 0.3$$

$$\mu_{R2}(x) = \min\{\mu_{out1}(x), \mu_{good}(x)\} = \{0.7, \mu_{good}(x)\} = 0.7$$

Calculated the output membership functions (fuzzy outputs), the aggregation of these functions will generate a one membership function, this is showed in the Fig. 5.

The WQI is then evaluated using the equation 3 replacing the membership functions:

$$WQI = \frac{\int x \mu_{out}(x) dx}{\int \mu_{out}(x) dx} = 0.663$$

3. Results

A set of measurements have been evaluated for proving the FWQI. Input data extracted from shrimp farms databases have been used to assess water quality between the one farming in 2007.

The validation of an index is not an easy task; although indexing processes can miss information their benefits are significant when measuring environmental impacts. The real validation process is done when the model is proved with real values; however, the most relevant aspect to highlight here is the methodology used in this work to develop de index. A comparison of the performance between deferent methodologies is proposed in this investigation. In Fig. 5 and 6, the FWQI index is compared with the CWQI index, which is used by the CCME and to benchmarking the WQI

index. A comparison of the performance for the FWQI and the CWQI used in environmental assessment could remark some interesting things. In fig. 5 and 6 the CWQI index is compared with the FWQI index.

The treatment of the information within the FIS influences the final score; CWQI and FWQI are always over a regular and good water quality score. However the CWQI index is above of the FWQI index; although the final values are similar they have some differences, they are done because the FWQI is calculated with reasoning cases that affects the status of the water, and there are some especial situations that the CWQI does not consider, as an example we can refer the hypoxia situations with the DO. The best way to validate the performance of the proposed index is comparing it with indicator of similar conditions in sea water.

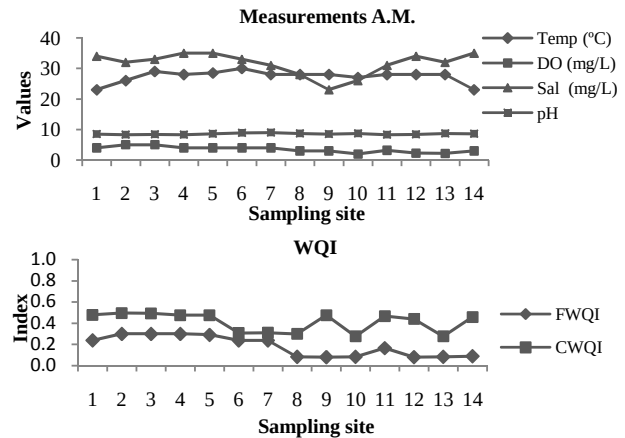


Fig. 5. Results for the WQI index calculated in measurements at morning.

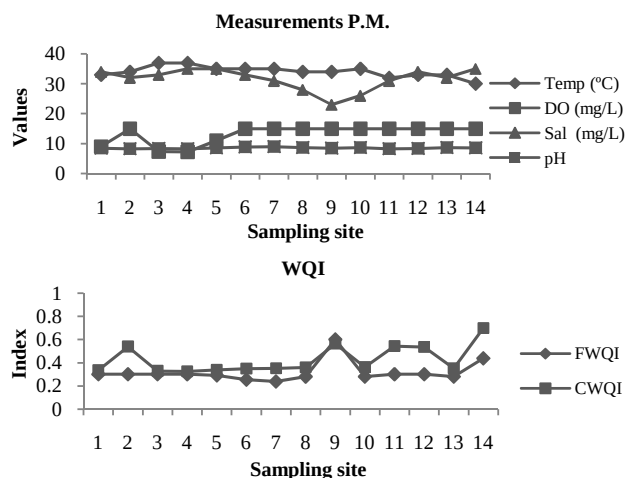


Fig. 6. Results for the WQI index calculated in measurements at afternoon.

4. Discussion

Our results support the hypothesis that a fuzzy inference system could model the environmental behaviour in shrimp ponds, this is done because the environment could be modelled using a reasoning process.

There is a lack of methodologies that help to regulate, control and assess the marine water quality. The indices that have been developed do not satisfy the needed of the coastal water bodies and their limitations were showed. Nowadays, there is not a legal regulation in the area, the standards do not evaluate environmental problems and they only present monitoring data on individual parameters, without interpret the problems that a set of parameters could generate in the environment. To resolve this problem various water quality indices have been developed. Comparisons with other indicators of WQI have been done showing that in some cases these indices have not a good performance. The NSF and the ACA developed some indices, however they only assess fresh water bodies and occasions the measurements have temperature restrictions. The CMME methodology could evaluate marine water bodies, however it is based in failed measurements, and it lost important information when any variables have a significant deviation. The lack of reasoning processes in water indicators is the main problem.

5. Conclusion

An original methodology to create an ecological abstract model that describes the artificial habitat in shrimp farming based on pattern recognition. This research establish an index that assess the status of the artificial

shrimp habitat based on four levels; excellent, good, regular and poor.

In this paper, we present a robust tool for water management in shrimp ponds; the methodology is presented in the form of a fuzzy inference system and it is based on a reasoning process implemented to evaluate information data. The necessity to link fuzzy systems and reasoning methodologies to evaluate water bodies has been showed. Studies are now needed to tune the different cases that are presented in the environment.

This model can be adjusted by assessing other aquaculture marine systems that have the necessity of analyze the status of their ecosystem. Nowadays some researchers are evaluating the possibility of use this model for embedding in expert systems or for creating predicting models that allows recognizing undesirable status into the ponds.

References

- [1] Martínez Córdova Rafael M., "Cultivo de Camarones Pendidos, Principios y Practicas", Ed. AGT Editor S.A., 1994.
- [2] [INE] Instituto Nacional de Ecología: La calidad del agua en los ecosistemas costeros de México. 2000.
- [3] [SEMARNAP], Secretaría de Medio Ambiente, Recursos Naturales y Pesca.: NOM-001-ECOL-1996
- [4] Brown, D.: A qualitative ecological model to support mariculture pond water quality management, *Bioinformatics*, Vol. 11 (1995) 595-602.
- [5] [ACA] Agencia Catalana del Agua (Catalonia, Spain). 2005. Available at: <http://www.mediambient.gencat.net/aca/ca/inici.jsp> [Accessed August 2007].
- [6] [NSF] National Sanitation Foundation International. 2005. Available at: <http://www.nsf.org> [Accessed August 2007].
- [7] [CCME] Canadian Council of Ministers of the Environment (Canada). 2004. An assessment of the application and testing of the water quality index of the Canadian Council of Ministers of the Environment for selected water bodies in Atlantic Canada. National indicators and reporting office. Available at: <http://www.ec.gc.ca/soer-ree/N> [Accessed August 2007].
- [8] Hernández, J., Zirino, A., Marione, S., Canino, R., Galindo, M.: PH-density relationships in seawater. *Ciencias Marinas*, Vol. 29 (2003) 597-508.
- [9] Ocampo, W., Ferré, N., Domingo, J., Schuhmacher, M.: Assessing water quality in rivers with fuzzy inference systems: A case study. *Environment International*, Vol 32. Elsevier (2006) 733-742.
- [10] Li, Y., Li, J., Wang, Q.: The effects of dissolved Oxygen Concentration and Stocking Density on Grown and Non-Specific Immunity in Chinese Shrimp, *Fenneropenaeus Chinensis*. *Aquaculture*, Vol. 256. Elsevier (2006) 608-616.

An Alternate Downloading Methodology of Webpages

Anirban Kundu¹, Alok Ranjan Pal¹, Tanay Sarkar¹, Moutan Banerjee¹, Subhendu Mandal¹,
Rana Dattagupta² and Debajyoti Mukhopadhyay³

¹Netaji Subhash Engineering College (West Bengal University of Technology),
West Bengal-700 152, India
{anik76in, chhaandasik, tanay.sarkar, moutanbanerjee, subhendu.mnd1}@gmail.com

²Jadavpur University, West Bengal-700 032, India
rdattagupta@cse.jdvu.ac.in

³Calcutta Business School, Diamond Harbour Road, Bishnupur, West Bengal-743 503, India
debajyoti.mukhopadhyay@gmail.com

Abstract

We propose an advanced method for downloading Webpages from the internet. In this technique, the whole system is considered as a bundle of crawlers which have been created dynamically at execution time. Numbers of crawlers are used depending on the requirement of downloading Webpages. The software module which interacts with WWW to search one or more Webpages is known as crawler. The numbers of crawlers are generated using the hierarchy structure of the Web server from which the data would be downloaded. Webpage downloader is an important issue for downloading Web documents from the internet to facilitate a Web user in terms of knowledge gathering. This type of downloaders are very popular in the 'Information Technology' field. All kinds of public data, accessible throughout the world without any authentication, can be retrieved any time from any geographic location using the downloading methodology. Typically, a downloading technique has been utilized to accumulate Webpages of different domains within a single computer machine one at a time. So, our aim in this paper is to show an advanced technique for downloading a lot of related Webpages with a minimum effort and time using Hierarchical Downloader consisting of several dynamic crawlers.

Keyword - Multi-downloading, Hierarchical downloading.

1 Introduction

In recent years, it has become important for perform downloading operation efficiently in terms of information retrieval [1] due to the enormous growth of World Wide Web (WWW). The world at present generates near about 1 to 2 exabytes of unique information each year, and also translates to about 250 megabytes for every man, woman and child on earth (an exabyte is a billion gigabytes). The World Wide Web Worm (WWWW) was one of the first Web Search Engines, and was basically a storage of huge volume of information [2]. With the advent of the WWW, users are now trying to propagate the information to a much wider audience more quickly via some medium of communication. In this information & technology era, if somebody wishes to gather information, he/she can find a lot of data related to the topic through WWW from any location in the world using some Web browser [3]. Web browser helps people to reach the desired information with an ease instantaneously over the internet. In practical scenario, a typical Web browser invokes Webpages one at a time. One has to check all the links available on a Webpage for downloading more than one Webpage for collecting overall information on a particular topic [4]. For example, if the user wishes to read a tutorial of a specific subject, all the hyperlinks should be checked in a trial and error basis. So, all of the information is specified within a Webpage in terms of their URLs [5]. A typical Webpage consists of a set of URLs that possibly contain the sought information. So, it will take a longer time to retrieve complete information using the available methods.

This paper is organized as follows: Section 2 shows Related Work in a brief. Our Approach is described in Section 3. In Section 4, Experimental Results are shown. Finally, Section 5 depicts the conclusion of the work.

2 Related Work

A huge volume of research works have been carried out in the past few years in Web Technology field [6]. Serial crawling has been proposed initially to download Web documents from WWW. In this technique, the Webpages are fetched and downloaded one after another. So, the available bandwidth of Internet connection has not been fully utilized.

In serial crawling, Bandwidth utilized is calculated by dividing 'Size of a Webpage to be downloaded' by 'Time taken to download'. Then, parallel crawling technique came into picture. Here, all the Webpages have been downloaded at a time in a parallel fashion to reduce the time for downloading as well as to utilize the bandwidth of Internet connection more with respect to serial crawling[7-9].

In parallel crawling, Bandwidth utilized is calculated by dividing 'Total size of Webpages to be downloaded' by 'Time taken to download all the Webpages'.

From the above discussion, it has been observed that parallel crawling is too better than serial crawling with reference to resource utilization and time complexity as a theoretical view [10]. But in practical sense, there are some limitations in parallel crawling also. The major limitation is that somebody has to fix the number of crawlers before starting the download operation. In some cases, it is not feasible to calculate number of crawlers in advance. Bandwidth utilization is not 100% in all the scenarios, since the number of crawlers are predefined [11].

To overcome this problem, our crawling technique is introduced in this paper for client-side operation. By our approach, 100% bandwidth utilization is always achieved. In best case, the time complexity is better than parallel crawling and in worst case both are same. In addition, a user needs not to know all the details and also needs not to follow all the links (URLs) on each Webpage. So, our approach is further time saving method with respect to a user. Duplicate Webpages are not downloaded through our approach.

3 Our Approach

In this section, the detailed view of Hierarchical Downloader is shown using algorithms and snapshots. Before going into details, following definitions should be remembered.

Definition 1: Seed queue - A storage space with the property of a queue (First-Come-First-Serve) is defined as Seed queue. The URLs which come first into the queue are re-

trieved first by the crawler module to download the corresponding Webpages from WWW.

Definition 2: Seed url - The url which is stored within the seed queue as the initial seed for beginning the downloading process from WWW, is defined as Seed url.

Definition 3: Life Cycle of Dynamic crawler - Life Cycle of Dynamic crawler refers to time period between its creation & destruction for a particular job to be executed.

Definition 4: Hierarchical Downloader - Hierarchical Downloader is a set of dynamically created crawlers whose number of instances is dependent on the number of seed urls. One crawler can download only one Webpage using its assigned URL from the seed queue in its life time. After downloading the Webpages, these crawlers are destroyed immediately.

Definition 5: Depth - Level of searching through WWW for downloading the required Webpages in a Hierarchical fashion.

This paper is based on multi-downloading of Webpages from WWW in an extended form of parallel crawling. Threaded programming has been used to maintain a reasonable number of crawlers at runtime. In this client based approach, the crawlers are considered as dynamic in nature. So, crawlers are destroyed automatically after completion of its execution. Algorithm 1 describes the life time of dynamic crawler. Number of URLs of seed queue are being checked for producing same number of crawlers in real time to download specific Webpages from WWW at a time. This concept of life time is utilized in this work. Each crawler downloads a particular Webpage.

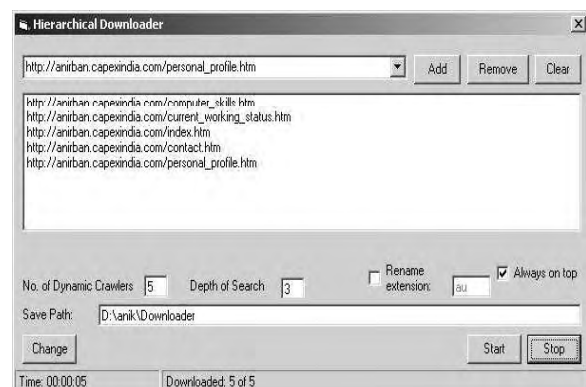


Figure 1. Hierarchical Downloading Snapshot

In our approach, the Webpages are downloaded from WWW using Algorithm 2. All visited Webpages have been discarded where as new Webpages are being saved within the storage of the computer machine itself. These Webpages are further parsed using 'Parser' tool for next level of downloading of the Webpages. The software module,

which extracts all the URLs from the downloaded Webpage, is known as Parser. So, our ‘Parser’ module only concentrates on the external links of the Webpages and not on the internal links. Internal link means a link to the different sections of same Webpage; where as external link means a link to the different Webpage. The depth level of searching / downloading is predefined and mandatory for stopping the downloading procedure such as set to a threshold value. After parsing, extracted URLs are being checked with the URL log entries within the system. Newly found / unique URLs are stored within seed queue for further downloading at the next level of search where as visited URLs are discarded as discussed in Algorithm 2. Algorithm 2 calls Algorithm 1 for creating several dynamic crawlers for next level of downloading whenever required.

In the 1st phase, the user has to submit the value of depth level while downloading through the software interface. Figure 1 shows Hierarchical downloading. There is a field named Number of Dynamic crawlers on the graphical user interface (GUI) of the package. This field shows number of active crawlers at any depth level of search. So, this is dynamic in nature. In every level of search, this value may be changed depending on number of URLs within seed queue at runtime.

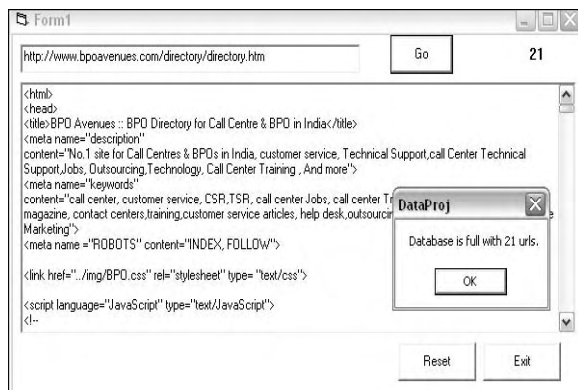


Figure 2. URL extraction from a downloaded Webpage using Parser

In the 2nd phase of our approach, the downloaded Webpages are parsed using ‘href’ tag of ‘html’ programming within Webpages as shown in Figure 2. The crawler & Parser module work in tandem, which dramatically reduces response time for data-intensive operations on large databases typically associated with decision support systems. To handle the Parser module, Fast Lexical Analyser Generator (FLEX) has been used running in the background. It is basically a tool for generating programs that performs pattern-matching on text. Only valid URLs are stored for next level of downloading after extraction. All other URLs are discarded through the software. Here, valid

URLs mean the URLs which are required for downloading either the full Web site or a tutorial or data sets, etc. as per the requirement of the user / client. No irrelevant URLs are stored within URL log file. For pointing out the related URLs, previously visited URLs are required. Consider, a user wishes to download a “C Tutorial” Webpages from “http://xyz.com/c_tutorial/index.html”. In this case, all the valid URLs should contain “xyz.com/c_tutorial/” as a part of its URL followed by some other characters. If some extracted URLs do not contain “xyz.com/c_tutorial/”, these URLs are irrelevant for download of this particular “C Tutorial” Webpages.

urls : Table	
id	url
1	http://www.macromedia.com/go/getflashplayer
2	http://www.Freshersworld.com/BPOavenues
3	http://www.bpoavenues.com/directory/directory.htm
4	http://www.bpoavenues.com/placementweek/career.htm
5	http://www.bpoavenues.com/placementweek/english.htm
6	http://www.bpoavenues.com/jobs/jobs.asp
7	http://www.Freshersworld.com/MBAJunction
8	http://www.mbajunction.com/overview/overview.htm
9	http://www.mbajunction.com/prepare/prepare.htm
10	http://www.mbajunction.com/career/career_advice.htm
11	http://www.mbajunction.com/career/mbajobs.asp
12	http://www.mbajunction.com/career/project_reports.htm
13	http://www.waveaxis.com/freshersworld.html
14	http://www.KochiCalling.com
15	http://www.google-analytics.com/urchin.js
16	http://www.BPOavenues.com
17	http://www.bpoavenues.com/BPO/ads/ad468x60img.asp
18	http://www.BPOavenues.com
19	http://www.Freshersworld.com/directory/colleges.asp
20	http://www.Freshersworld.com/directory/sourcecodes.asp

Figure 3. An instance of URL database created through our approach

In the 3rd phase of our approach, newly extracted URLs are checked with the existing URL database. Unvisited / unique URLs are being saved as depicted in Figure 3 using MS-Access database. The following snippet shows how the checking took place within the programming.

```

If IsNull(DatabaseLookUp("URL", "SQL Query")) Then
    Add the record
Else
    Discard the URL
End If

```

In the 4th (final) phase of our approach, the downloaded Webpages are finally saved within the computer as shown in Figure 4 & Figure 5 respectively.

Algorithm 1 Life Time of Dynamic crawler

Input : A set of seed urls

Output : Downloaded Webpages

Step 1 : Check number of URLs within seed queue

Step 2 : Generate ‘N’ number of crawlers in runtime based on number of URLs in seed queue

Step 3 : Assign each seed (URL) to a specific crawler

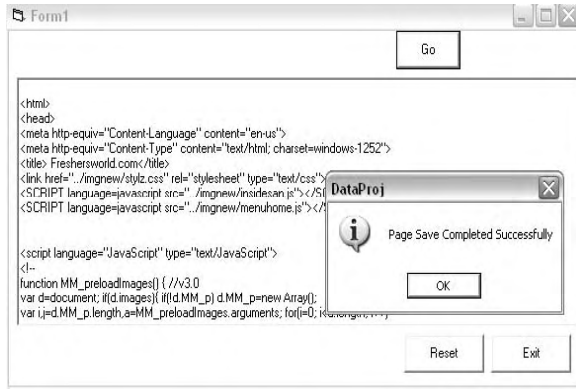


Figure 4. An instance of Webpage saving

$\{c_j < -s_i\}$

Step 4 : Search for Webpages in the Web using the specific crawlers

Step 5 : Download Webpages which are found through searching

Step 6 : Kill the crawlers after successful download of Webpages

Step 7 : Stop

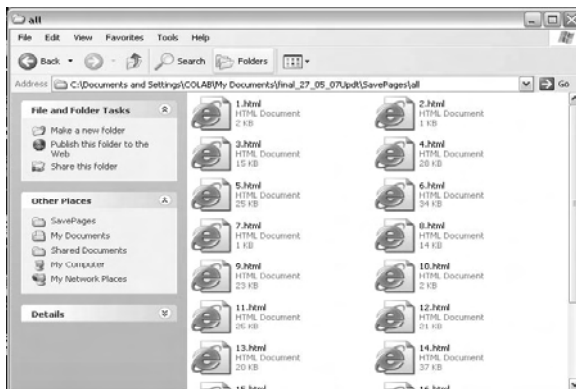


Figure 5. Webpage Repository after downloading from WWW

Figure 6 shows internal structure of Hierarchical downloading. Hierarchical technique means a technique with a hierarchical view. A number of crawlers have been utilized based on the requirement of the level of hierarchy for this purpose. The 'number of crawlers' is calculated using the directory structure of the concerned Web directory. In this paper, an advanced technique, for downloading a lot of related Webpages with a minimum effort and time using Hierarchical Downloader consisting of several real-time crawlers, is depicted using Algorithm 2. The search

limit, for downloading Webpages, depends on the 'Depth' value. Until the 'Depth' value is reached, our method downloads all the related Webpages as per schedule. All the required Webpages of each level have been tried to download at a time in a parallel manner using dynamically created crawlers utilizing all the available bandwidth of Internet connection dedicated for the particular client machine.

Algorithm 2 Hierarchical Downloading

Input : A set of seed urls within seed queue and Depth level of searching

Output : Storage of downloaded Webpages

Step 1 : Initialize i with 0

Step 2 : Continue loop until $i > \text{Depth}$

Step 3 : Call Algorithm 1

Step 4 : Check whether the URLs of downloaded Webpages are already visited or not

Step 5 : Discard already visited Webpages

Step 6 : Save new Webpages

Step 7 : Hyperlinks of all saved Webpages are extracted using Parser

Step 8 : Check whether the extracted URLs are already visited or not

Step 9 : Save those extracted hyperlinks(URLs), which are still not visited, within seed queue as well as in storage

Step 10 : Increment i by 1

Step 11 : Stop

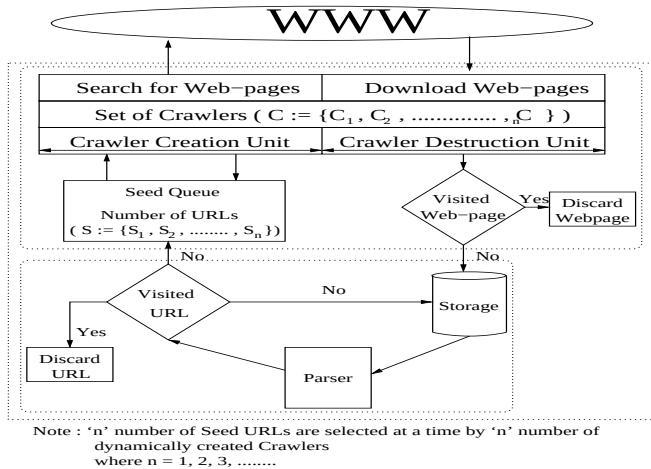


Figure 6. Internal Architecture of Hierarchical Downloading

Question: Why any parallel approach would be slower than our Hierarchical methodology?

Answer: In a typical parallel approach, the number of crawlers are fixed before downloading the documents from WWW. So, a user can have two options for downloading all the Webpages.

1st option: The user has to approximate the amount of number of crawlers before the download operation has taken place. This approximation is not always correct. Sometimes, number of crawlers are greater than the available URLs; and in some cases, it is vice-versa. So, the practical usage is not 100% effective.

2nd option: The user has to supply all the URLs as seeds for fixing the number of crawlers accurately. But, it is pretty difficult to know all the details of URLs at starting point (i.e., in case of a tutorial download).

To overcome these 1st & 2nd options (loop holes), our Hierarchical methodology is a perfect solution. In our approach, the user needs not to know all the details of URLs to be downloaded, since the number of URLs would be calculated at runtime dynamically. After downloading the Webpage (using the initial seed url), the Parser module parses all the URLs from the Webpage. These parsed URLs are then submitted to the seed queue for downloading further Webpages. The number of crawlers are decided using number of URLs within seed queue at that time instance. So, there would be no approximation on creating number of dynamic crawlers. After downloading the specific Webpages, these crawlers would be killed / destroyed at that particular level. So, the 'number of crawlers' for the next level is again dependent on the number of parsed URLs for the next level. So, in our methodology, the total number of crawlers is not fixed rather dynamic in nature. So, all the URLs can be downloaded in a parallel manner depending on the available bandwidth of the system. No process would be queued if the bandwidth permits. So, our approach is faster than any other typical parallel approach.

Example: Consider, there are 'N' number of crawlers in a parallel downloader and let 'P' number of Webpages have been downloaded.

Case 1: If ($P < N$); then 'P' number of Webpages can be downloaded in parallel.

Case 2: If ($P == N$); then 'P' number of Webpages can be downloaded in parallel.

Case 3: If ($P > N$); then 'N' number of Webpages can be downloaded in parallel and the remaining 'P-N' Webpages would be downloaded in the next iterations.

From this example, it can be said that our approach is better in case of $P > N$, since 'P' number of crawlers would be created dynamically for downloading in parallel fashion. That means, our method is faster than other typical parallel methods.

4 Experimental Results

In this section, experimental results are shown based on the following configuration of a computer system:

Processor - Pentium 4 (P4); Processor speed - 2.8 GHz; Primary memory (RAM) - 512 MB; Hard disk - 80 GB; Inter-

Table 1. Time required for downloading 100 URLs using Hierarchical Downloader

Sl. No.	Website Name	Number of downloaded URLs / Webpages	Time needed for downloading URLs (hh:mm:ss.ms)
1	www.freshersworld.com	100	0:1:29:98
2	www.indiagsm.com	100	0:0:34:79
3	www.cocacola.com	100	0:0:16:39
4	www.yahoo.com	100	0:0:39:90
5	www.gmail.com	100	0:0:32:83
6	www.pepsi.com	100	0:0:46:80
7	www.anandabazar.com	100	0:0:49:11
8	www.telegraph.com	100	0:0:30:07
9	www.times.com	100	0:0:18:66
10	www.bollyextreme.com	100	0:0:38:26

Table 2. Comparison between DAP & Our Approach

Sl. No.	Set of downloaded Webpages	Time taken by DAP (sec.)	Time taken by our approach (sec.)
1	http://capexindia.com/anirban/computer_skills.htm	3	1
2	http://capexindia.com/anirban/computer_skills.htm http://capexindia.com/anirban/current_working_status.htm	5	2
3	http://capexindia.com/anirban/computer_skills.htm http://capexindia.com/anirban/current_working_status.htm http://capexindia.com/anirban/index.htm	7	2
4	http://capexindia.com/anirban/computer_skills.htm http://capexindia.com/anirban/current_working_status.htm http://capexindia.com/anirban/index.htm http://capexindia.com/anirban/contact.htm	9	3
5	http://capexindia.com/anirban/computer_skills.htm http://capexindia.com/anirban/current_working_status.htm http://capexindia.com/anirban/index.htm http://capexindia.com/anirban/contact.htm http://capexindia.com/anirban/personal_profile.htm	11	5

net connection speed - 512 kbps, 1:1 RF line;

Table 1 enlists the time requirement for downloading 100 URLs through our crawling methodology as a sample study. It has shown better result than Download Accelerator Plus (DAP). In Table 2, the comparison between DAP & our approach has been shown. In this part of the experiment, one or more Webpages have been downloaded all at a time as a concurrent process.

DAP configuration used - 8.5.5.3

Maximum simultaneous downloads possible by DAP - 20

In this paper, our approach is compared with DAP as it is available free of cost (trial version) in the Internet and also DAP is very much popular software for the internet users. If somebody wishes to download the Web documents using DAP, it is possible only for maximum 20 documents. Since, our DAP version supports maximum 20 simultaneous downloads. But, in case of our approach, there is no upper limit for parallel connections. Theoretically, the number of connections are infinite and practically, the number of connections are dependent to the available bandwidth of the Internet connection. In Table 3, the time taken by our method in Single, Parallel & Hierarchical mode of crawling has been shown. From the experiment, it is established that the result would be better in the case of Hierarchical mode of crawling, since the number of crawlers are not pre-defined.

Table 3. Comparative Study on Time taken by Hierarchical Downloader in Single, Parallel & Hierarchical mode of crawling

Note: Number of Parallel crawlers = 2 for this experimentation.

Website Name	Total Number of Webpages downloaded	Time taken by Single crawler	Time taken by Parallel crawler	Time taken by Hierarchical crawler
freshersworld.com	4994	9 hr. 30 min.	4 hr. 45 min.	58 min.
theatrelinks.com	469	44 min.	22 min.	10 min.
indiagsm.com	34	3 min.	1 min. 30 sec.	30 sec.
rediff.com	163	3 min.	1 min. 30 sec.	20 sec.
w3.org	2333	6 hr.	3 hr.	37 min.
indiafm.com	7087	7 hr.	3 hr. 30 min.	53 min.
nokia-asia.com	193	2 hr.	1 hr.	17 min.
amazon.com	349	2 hr. 58 min.	1 hr. 29 min.	19 min.

5 Conclusion

In this paper, an advanced method for downloading Webpages has been proposed. In case of a typical downloading software, Single crawling or Parallel crawling are used for downloading the Webpages of selected URLs. In our proposal, an enhanced methodology is discussed to minimize time requirement while crawling through WWW using Hierarchical crawling. The main advantage of this type of crawling system is to generate & kill any number of dynamic crawlers depending on the number of URLs present within seed queue at any depth level of the concerned hierarchy. These crawlers are generated at runtime based on the number of URLs present within seed queue at any depth level of concerned Web hierarchy. After downloading the specific Webpage, the respective crawler would be destroyed automatically. At any time instance, maximum number of Webpages available for Hierarchical downloading depends on the allowable bandwidth of the system.

References

- [1] Hongfei Yan, Jianyong Wang, Xiaoming Li, Lin Gu, "Architectural design and evaluation of an efficient Web-crawling system," *The Journal of Systems and Software*, 60(3): 185-193, 2002
- [2] Soumen Chakrabarti, Byron E. Dom, Ravi Kumar, Prabhakar Raghavan, Shidhar Rajagopalan, Andrew Tomkins, David Gibson, Jon Kleinberg, "Mining the Web's Link Structure," *IEEE Computer*, (32)8: August 1999, pp. 60-67
- [3] Sergey Brin, Lawrence Page, "The Anatomy of a Large-Scale Hypertextual Web Search Engine," *Proceedings of the Seventh International World Wide Web Conference, Brisbane, Australia, April 1998*
- [4] Arvind Arasu, Junghoo Cho, Hector Garcia-Molina, Andreas Paepcke, Sriram Raghavan, "Searching the Web," *ACM Transactions on Internet Technology*, Volume 1, Issue 1, August 2001
- [5] Debajyoti Mukhopadhyay, Sajal Mukherjee, Soumya Ghosh, Saheli Kar, Young-Chon Kim, "Architecture of A Scalable Dynamic Parallel WebCrawler with High Speed Downloadable Capability for a Web Search Engine," *The 6th International Workshop MSPT 2006 Proceedings, Youngil Publication, Republic of Korea, November 2006, pp.103-108*
- [6] M. Burner, "Crawling towards eternity: Building an archive of the world wide web," *Web Techniques*, 2(5), 1997
- [7] Marc Najork, Allan Heydon, "High-performance web crawling," In J. Abello, P. Pardalos, M. Resende, editors, *Handbook of Massive Data Sets*, Kluwer Academic Publishers, Inc., 2001
- [8] Marc Najork, Janet L. Wiener, "Breadth-first search crawling yields high-quality pages," In *Proc. of 10th International World Wide Web Conference, Hong Kong, China, 2001*
- [9] Martijn Koster, "The Robot Exclusion Standard," <http://www.robotstxt.org/>
- [10] Paolo Boldi, Bruno Codenotti, Massimo Santini, Sebastiano Vigna, "Ubicrawler: A scalable fully distributed web crawler," In *Proc. AusWeb02, The Eighth Australian World Wide Web Conference, 2002*
- [11] Junghoo Cho, Hector Garcia-Molina, "Parallel crawlers," In *Proc. of the 11th International World-Wide Web Conference, 2002*

Prediction of Velocities on Motorways by k-Means Clustering

Johannes Asamer and Kashif Din

arsenal research

Giefinggasse 2

1210 Vienna

Austria

johannes.asamer@arsenal.ac.at, kashif.din@arsenal.ac.at

Abstract

In this paper a method for the prediction of vehicle velocities is described that can be used for any point of time in the future. The approach is based on a two step clustering which uses toll transaction data of the training of the model. The results are different clusters for each road segment, containing the velocity value and its probability of occurrence. Furthermore the results of the method have been compared to a statistical method as well as to a neural network.

1 Introduction

A prior objective of logistic applications is to arrange trips in a manner that route planning through congested roads is avoided. If information of current congestions is available at the time of route planning, the departure time or route of a trip can be adapted accordingly. For pre-trip planning, as requested in logistics, a reliable prediction of congestions is necessary. Moreover the time horizon of the prediction should be as high as possible, in order to improve the planning of trip. Current prediction models mostly provide a short term prediction [4] [6] [12] [3] [8] based on current traffic information. In this paper a method is presented, that allows the prediction of travel times for any given timestamp in the future.

In Austria all vehicles with a maximum permissible gross weight exceeding 3.5 t are charged for the usage of the motorways and expressways by mileage. Therefore these vehicles have to carry an on board unit (OBU) which makes an identification possible on toll gantries. A centralized system collects all toll transaction data on gantries, out of that it is possible to calculate travel times between two gantries (sections) [6]. The method described in this paper has been developed by using the data basis of the toll system.

Table 1. Definitions of the pre-cluster

Description	Possible values
Time of day	00:00-00:15, 00:15-00:30 ,..., 23:45-00:00
Day of week	Sunday, Monday-Thursday, Friday, Saturday
Holiday time	Yes, No

2 Approach

In this approach the prediction is based on two steps of clustering. First the data will be divided into groups, whereas the selection of the groups has been described in different works [13] [5] [15]. The second step is to cluster all group members inside one group by an online k-means cluster algorithm. The result are different clusters i.e. different velocities (or travel times) for each section, whereas each cluster possess a probability for the true velocity belonging to the cluster.

2.1 Pre-clustering

On most of the motorways recurrent traffic situations can be observed, but the occurrence of different situations depend on different factors. Obviously for the prediction of traffic the influencing factors have to be known in advance. For instance a traffic accident has doubtlessly a negative impact on the traffic situation, but it can't be used for the prediction, because the factor itself cannot be predicted. Therefore the aim of the pre-classification was to use significant factors that can be determined for any point of time. Some work [15] [1] [14] [13] showed that the time of day, day of the week and holiday times have significant impact on the traffic situation. Furthermore the impact of weather conditions has been excluded for this method due to little sample rates in some clusters. In table 1 the different factors respective levels of factors are listed.

Out of the number of factors and levels 768 combina-

tions are possible ($2 \times 4 \times 96$). Basically the number of classes should be kept low in order to avoid impacts of the curse of dimensionality [2]. Fortunately the data are evenly distributed in our case, which means data points are available for each of the 768 different combinations.

2.2 k-means clustering

After all data have been assigned to one cluster, a k-means cluster algorithm [2] has been applied to the data points inside each cluster. For the usage of this algorithm the number of k clusters has to be predefined. In our case we have chosen four clusters, whereas the initialization was not done randomly, as it is done usually, but with four different increasing steps of velocities (15, 25, 45 and 65 km/h). The number of clusters resulted from an agglomerative hierarchical clusteranalysis. Thereby at the beginning one cluster is represented by one datapoint, during the process the clusters are merged together successively. Between each merging step a quantisation error for the current clusters is calculated. By this analysis the optimal number of clusters is found if the quantisation error and the number of clusters is low (see Fig. 1).

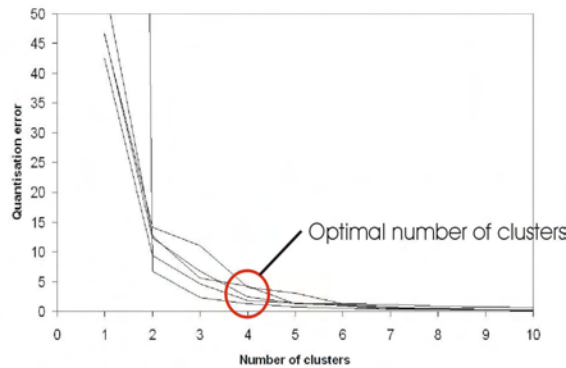


Figure 1. Quantisation error during hirachial cluster process

After the initialization a data point is selected, randomly and the closest cluster to it, is determined. Closeness is defined by the euclidian distance. If the closest cluster has been found, it is moved into the direction of the assigned data point. How far the cluster is moved, is set by the learning rate r whereas r lower or equal to 1 (1 means the cluster moves completely to the selected data point). For the termination of the process either a fixed number of steps is defined or the learning rate is decreased after each step (or each epoch). The latter is comparable to simulated annealing [9]. In our case the learning rate has been kept constant, whereby the number of learning steps was limited to 1000. After this process the number of data points for each cluster

Table 2. Coding of the day of the week

Sunday	1	0	0
Monday-Thursday	0	1	0
Friday	0	0	1
Saturday	1	1	1

has been assigned. Compared to the total number of data points, this value indicates the probability of belonging to the according cluster.

2.3 Comparable Methods

For the purpose of comparison two methods have been applied to the same data. For a statistical approach a histogram over all data inside one precluster has been calculated, whereas the size for one class is 10 km/h, beginning from 0 km/h up to 120 km/h. The prediction of the velocity for one precluster is the arithmetic mean value of all velocities inside the histogramm class with the largest amount of according values.

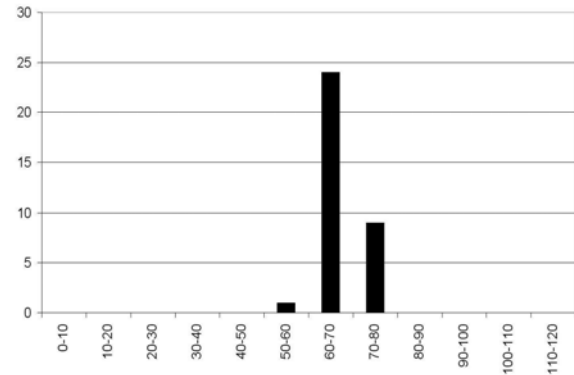


Figure 2. Histogram for one precluster

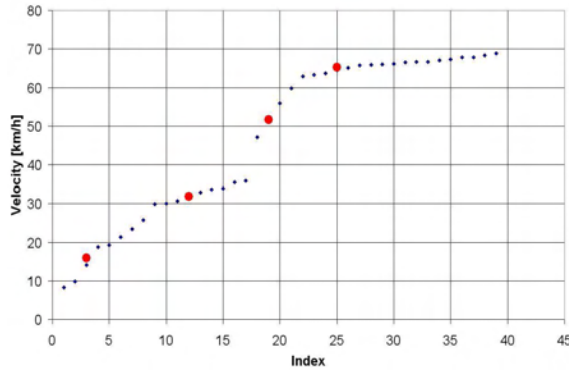
Furthermore the data have been applied to an adaptive neural network (ANN), whereas networks with one and two hidden layer have been investigated. Moreover the number of neurons per layer has been varied between one and ten. The same values out of which the ID of the precluster is calculated, are used here as inputs for the ANN built with a different coding. Holidays have been coded by a value of 0 (no holiday) or 1. The time of the day is a continuous value out of the interval $[0,1]$ and the day of the week has been coded by three binary inputs 2. For the accomplishment of the supervised learning procedure the desired output value (velocity of the vehicles) has been added to the five-dimensional input vector.

Table 3. Result of the k-means clustering

Cluster center	Assigned data points	Probability
65.3 km/h	19	0.49
31.9 km/h	10	0.26
15.9 km/h	7	0.18
51.6 km/h	3	0.07

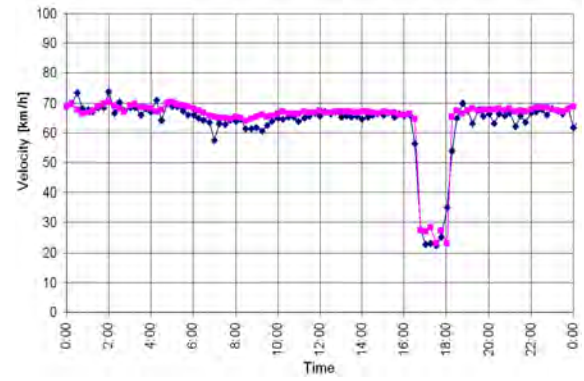
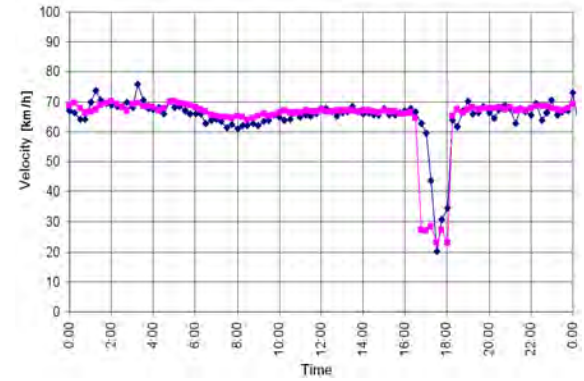
3 Application

As already mentioned the velocities calculated by the toll collection system have been used for the application of the method. A set of data of almost half a year has been divided into training and test data, which is the minimum requirement for the evaluation of a machine learning algorithm [7]. Afterwards the method has been applied only on the test data. The pre-cluster-process resulted in minimum 20 and up to 40 data points inside one cluster, which was assumed to be a sufficient number for a k-means cluster process. In figure 3 all data points of one pre-cluster are illustrated in ascending order.

**Figure 3. Visualization of data points with k-means cluster**

Furthermore the result of the k-means clustering is visualized in figure 3 by the four larger dots. In table 3 the value of each cluster centre, as well as the number of assigned data points, are listed. Out of this number, a probability can be calculated, which expresses how likely a new data point (or a future value) inside the same pre-cluster will be assigned to this cluster.

The procedure for making a prediction is first to select the point of time for which the prediction should be done. Then it is necessary to determine the day of the week and if this time is within holiday time or not. By this information the cluster with the highest probability is selected out of the according pre-cluster. The value of the cluster centre represents the prediction for the selected time.

**Figure 4. Visualization of true velocities (dark) and k-means cluster (light)****Figure 5. Visualization of true velocities (dark) and k-means cluster (light)**

4 Evaluation

The evaluation has been applied to the previously selected train data. Therefore the true value is compared to the predicted one, first visually and afterwards by calculating the root mean square error (RMSE) as described in formula 1.

$$\sqrt{\frac{1}{N} \cdot \sum_{n=1}^N (x_{true,n} - x_{est,n})^2} \quad (1)$$

In figure 4, 5 and 6 three examples are illustrated to describe the prediction quality for the k-means cluster method. In figure 4 the velocities during one day for one section are visualized. It is clearly visible, that the predicted velocities fit very well with the true values. Also in figure 5 the prediction is good, but the congestion in the afternoon (at about 5 p.m.) was predicted about half an hour to early. Finally in figure 6 a failed prediction is visualized: The congestion

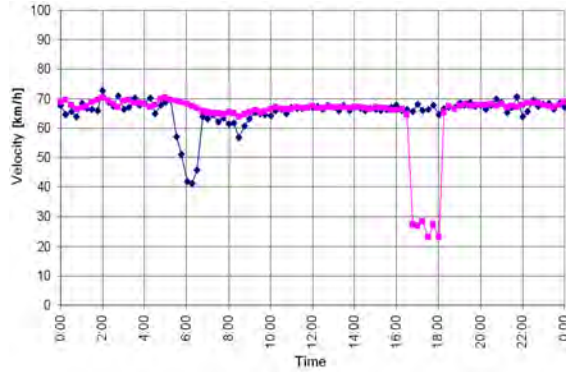


Figure 6. Visualization of true velocities (dark) and k-means cluster (light)

Table 4. Result of the evaluation

	k-means	ANN-1	ANN-2	Stat. Meth.
all data	8.21	7.03	6.76	8.39
≤ 50 km/h	7.11	5.1	4.07	7.26
> 50 km/h	27.76	31.85	35.59	28.46

in the morning was not detected, but therefore a traffic jam was predicted for the afternoon which did not occur.

For the investigated section three different RMSE-values have been calculated for each of the three methods. First for all, data and afterwards (true) values over (respectively) under 50 km/h. It is mentioned in different works that the prediction of congested situation is much more difficult than for free flow situation [11] [10]. Therefore, a separate calculation of the RMSE should increase the possibility to detect improvements of different methods. For example it is possible that the RMSE over all data changes little, while the prediction of congested situations (which occurs rarely, compared to free-flow situations) could be improved significantly. To reduce the influence of random on the evaluation results, the k-means method as well as the ANN has been applied ten times, always with a random initialization. The arithmetic mean value over all runs has been used for the comparisons. For the ANN beside the hidden layers also the number of neurons per layer has been varied, but only the architecture with the best results has been used for the comparison of the RMSE.

In table 4 the evaluation results for each investigated method are listed.

In table 4 the assumed difficulties in the prediction of congested traffic situations is visible, because the RMSE for velocities lower 50 km/h is about five times compared to non-congested situations. The results for the k-means and the statistical method are quite similar, but with slightly better values for the RMSE of the k-means. The results of

the two ANN approaches are interesting because the error is smaller for values over 50 km/h respectively for all data, but it is much higher for velocities below 50 km/h. Because the prediction of velocities is interesting especially in critical situations like congested traffic, the focus of this paper is to evaluate the ability of the methods for predicting these situations.

Furthermore, table 4 is thought to provide compare values for other methods.

5 Conclusion and Future Works

In this paper a method for the prediction of velocities on motorways has been introduced and compared to other methods. For the evaluation of the methods, the predicted values were compared to the true ones. The main advantage of this type of prediction is the large time horizon. While other methods are only able to predict the traffic up to one hour, here it is possible to make a forecast for even every point of time in the future. Furthermore the k-means method showed the best results for values below 50 km/h and is because of its simplicity easy to implement. The prediction of the velocity is a sharp value, but the result of the k-means cluster process provides probabilities for each class. Indeed the class with the highest probability is taken for the prediction, it is also possible to take the class with the second highest probability if required. This could be done by a combination of a short term prediction and switch between two (ore more) classes with high probabilities.

References

- [1] J. Asamer, K. Din, and W. Toplak. Self organizing maps for traffic prediction. *Proceedings of the 25th IASTED Conference on Artificial Intelligence and Applications*, 2007.
- [2] C.M. Bishop. *Neural Networks for Pattern Recognition*. Oxford University Press, 1995.
- [3] R. Chrobok, O. Kaumann, J. Wahle, and M. Schreckenberg. Three categories of traffic data: Historical, current, and predictive. In *Proceedings of the 9th IFAC Symposium Control in Transportation Systems, Braunschweig, Germany*, pages 250–255, 2000.
- [4] L. Chu, Z. Yang, and J.-S. Oh. Development of a synthetic link travel time prediction model for atmis applications. *Proceedings of Annual Meeting of the TRB*, 2001.
- [5] K. Din. Analyse und clusterung von verkehrsdaten. Master's thesis, FH Technikum Wien, 2003.

- [6] N.-E. El Faouzi, A. Ben Aissa, J. Sau, and O. De Mouzon. Highway travel time estimation and prediction from toll collection system. *Journal of Intelligent Transportation Systems - Revised Version (submitted 01/2006)*, 2006.
- [7] A. Flexer. Statistical evaluation of neural network experiments: Minimum requirements and current practice. In *Proceedings of the 13th European Meeting on Cybernetics and Systems Research*, pages 1005–1008, 1996.
- [8] N. Hainitz. Prognose von reisezeiten. Master’s thesis, FH Technikum Wien, 2006.
- [9] A. K. Hartmann and H. Rieger. *New Optimization Algorithms in Physics*. John Wiley & Sons, 2004.
- [10] D. Helbig. *Verkehrsdynamik*. Springer-Verlag, 1997.
- [11] D. Helbig and A. Hubermann. Coherent moving states in highway traffic. *Nature* 396, 738-740, 1998.
- [12] J. Rice and E. van Zweet. A simple and effective method for predicting travel times on freeways. In *2001 IEEE Intelligent Transport Systems Conference Proceedings*, 2001.
- [13] B. Nowotny, J. Asamer, K. Din, and R. Karim. Classification of traffic data time series by cluster analysis, artificial neural networks and anova. In *Proceedings 10th World Congress and Exhibition on Intelligent Transport Systems and Services, Madrid, Spain*, 2004.
- [14] W. Toplak, J. Asamer, and K. Din. Neural networks support causal reasoning in traffic telematics. *Proceedings of the 25th IASTED Conference on Artificial Intelligence and Applications*, 2007.
- [15] D. Wild. *Prognose von Verkehrsstärken anhand klassifizierter Ganglinien*. Shaker Verlag, Aachen, 1996.

Business Process Mining by Means of Statistical Languages Model

Dafne Rosso Pelayo and Raúl A. Trejo Ramírez
Instituto Tecnológico de Estudios Superiores de Monterrey
drosso@pemex.gob.mx, raul.trejo@itesm.mx

Abstract

The goal of this research is to provide an alternative for business processes evaluation and tracking, based on the analysis of non-structured information generated by such processes within the organization areas. In this article we introduce a method to determine the occurrence probability of a business process within the enterprise's text documents. The proposed method introduces the use of Statistical language model (SLM) [1], as a new technique in business processes mining area [2]. In order to obtain this objective the following is considered: the probability that a sub process or a process part is in the text paragraph; the probability that this text belongs to a business process; the language model of the processes set; and the set of realized activities which is reconstructed according to the processes that gave origin to the analyzed documents.

1. Introduction

Business processes mining is a technique that uses workflows registered within enterprise applications logs [3, 13, 15] to reconstruct business processes. From the earlier works in business processes mining [4] to present day, there has been a development of new heuristic techniques based on intelligent computation that involve genetic algorithms, data mining algorithms, and neuronal networks, in addition to traditional statistical techniques. In [2] there is a summary of these developed techniques. For example, [3], [10], [11], show the reconstruction of a business processes model by making a job workflow modeling, based on the analysis and events log in the period of time in which these happen, nevertheless, these

analyses come mostly from the (structured) logs of enterprise systems like SAP, PeopleSoft, or CRM systems.

The business processes of our interest follow the process classification framework (PCF), this framework is a high level neutral enterprise model, that reflect the activities in which the enterprise incurs to satisfy its business and organizational objectives [14].

The alternative we propose is a novel technique to perform business processes mining. One of the main motivations for this research is that, due the nature of the processes, the non-structured information they generate is typically very generic, vague and complex in structure. Besides, business processes in many cases are not completely automated, because there are activities, analysis of experts and decision making which are not feasible to structure. As [2] indicates, more formal research is required for business processes, being important to look for solutions that allow analysis and obtain knowledge of this type of information.

In this research we focus in text documents generated as a result of business processes execution, instead of starting from the analysis of workflow logs.

The documents analyzed, belong to a dominion of widely dispersed texts, i.e., texts belong to different areas and contain highly dispersed information of the items within the document. To reconstruct the original processes, a statistical language model will be used to classify documents, SLM has been applied for information retrieval in heuristic techniques for document classification [1, 9], in this work, we use SLM for document classification according to process events or activities [15]. This allows for establishing a method of text clustering for the document, which operates by means of a probability precision.

2. Motivation

For organizations, it is of vital importance to make an effective use of the information generated in its processes, by means of schemes of Businesses Intelligence [5]. It is common in enterprise businesses processes that a great variety of the resulting process information is stored in non-structured documents (documents, presentations, spreadsheets, electronic mails, etc.).

More than 80% of the information generated in businesses processes is stored in text documents without structure [6]. Although in most cases it is possible to determine the documents context, generally it is not possible to obtain significant data about the original processes, and it is not possible to generate information about all the process by means of the integration of contained information in a document set.

In order to analyze this type of information, exist solutions of text mining that determine the main subject documents, the relations between documents and the patterns within these texts [7]. However, these solutions, in general cannot, determine the causal variables in the process, nor can they determine in which part of the process the pattern or the found rule is applied to.

The following sections will present the use of SLM to determine the involved processes in the non-structured information. The discovery of patrons and its inference in the process will be made in a following article that of pursuit to this project.

3. Towards Process definition from non-structured documents

Next we present the use of statistical language model (SLM) in the context of document treatment and we extend its use to determine processes and document treatment.

3.1. SLM within the context of process

SLM is a method of categorization that operates by means of a similarity coefficient, on the basis of a probability of precision [9], it represent a probability

distribution over the document text terms. The concept of similarity coefficient of the terms is considered in this model, according to the probability of occurrence of a sequence of independent terms j for query $Q = (j_1, j_2, \dots, j_m)$, according to the SLM generated for the document D_i . SC in probability terms is indicated in the following formula.

For the construction of a SLM, a clustering method is used, which establishes a valuation criterion in terms of probability of the items that integrate the document.

$$SC(Q, D_i) = P(Q, MD_i) \dots \quad (1)$$

An SLM is a probabilistic mechanism to generate a piece of the text [1], throw this language model, it's possible to define a distribution for all the sequences of possible words that represent the presence or absent of the process in the inspected document. The set of words that integrate the query Q , it's the set of activities (a set of pre-defined steps of the process) that reflect the event occurrence [13] or absent in the process.

To obtain the SLM we must calculate the probability that term j is in the document D_i and the probability that it is not in the document D_i . This is expressed in the following formula:

$$SC(Q, D_i) = \prod_{t_j \in Q} P(Q | MD_i) \prod_{t_j \notin Q} (1 - P(Q | MD_i)) \quad (2)$$

In order to apply these formulas in our business processes dominion, instead a query Q we determine the set of activities $A = (a_1, a_2, \dots, a_k)$ of length k , where k is the number of defined activities whose existence in texts allows to identify if the process were carried out, and also we determine a set of elements $B = (b_1, b_2, \dots, b_k)$ which indicate that the process was not carried out.

Each one of the elements of this activity set will be analyzed taking the following formula into account:

$$SC(a_k, D_i) = \text{Max}(\prod_{t \in a_k} P(t | MD_i) \prod_{t \notin a_k} (1 - P(t | MD_i))) \quad (3)$$

Where a_k is an activity pertaining to the set A whose presence (maxim probability for document D_i) indicates that process i was found.

$$SC(b_k, D_i) = \text{Min}(\prod_{t \in b_k} P(t | MD_i) \prod_{t \notin b_k} (1 - P(t | MD_i))) \quad (4)$$

Where b_k is an activity pertaining to the set B whose absence (minimum probability for document D_i) indicates that process i was found.

$P(t|MD_i)$, is the expected probability that the term t occurred in the document D_i . Terms t take the values from the items that integrate the A and B activities sets, this is $t \in a_i \perp a_i \in A$ or $t \in b_i \perp b_i \in B$

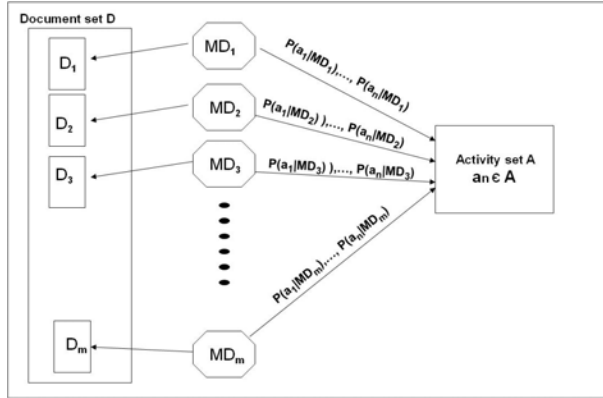


Figure 1. Statistical language model for the processes association

Once that the expected probability for all terms that compose the document is calculated, formula (1) is applied in such a way that the activities A are evaluated to obtain the probability that the activity is present in a document by means of the SLM of each document formula (3) and (4).

A $k \times m$ matrix is obtained from the SLM MD_i of documents where k is the activities set length and m is the document number. Once we find this probability, we continue proceed to analyze the resulted probabilities of the SLM by means of logistic regression. The main reason to apply logistic regression is that of using clustering analysis to find what process belongs to which document, based on the resulting probability (independent values) of the activities. The final result is a model that allows the prediction of the clustering when these activities are present.

When applying this method to the probability matrix of the SLM, the results allow associating the text set with the corresponding processes that generated the information.

4. Preliminary Results

As an exploratory example, the method was applied to 266 documents pertaining to one of 15 probable processes. The similarity coefficients evaluation took place for the 1763 contained independent terms in our text sample.

Once the matrix with the SC was found, we performed the construction of the SLM for documents of the determined set A of 5 elements, this is, $P(Am|Dn)$ with $n= 266$ and $m = 5$, for effects of this analysis.

For example, given a document $D1$ and an activity $a1$:

$D1 =$ “En esta novena reunión del Cockpit de la Dirección General, se presentan los avances acumulados durante el año y el estimado de cierre para agosto en las principales variables operativas de las cadenas de crudo y gas.”;

$a1=$ “estimado de cierre”;

We obtain the process and activities associations shown in Figure 2.

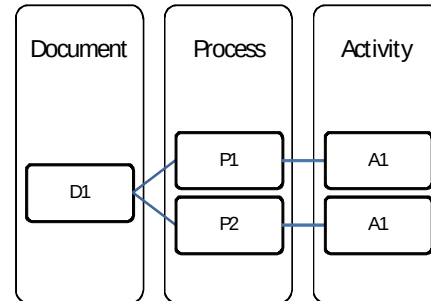


Fig. 2 Association processes result for a document

From the method, we obtain $P(a1|MD1) = 0.0000043344276124$ whereas the average for $n>1$ was of $P(a1|MDn) = 0.0000000000012426328$.

This preliminary result is part of a broader research goal that deals with identification of process failure based on evidence on non-structured text.

5. Conclusions

We introduced a new alternative for the text processes mining of business documents by means of statistical language model and its processes association.

We obtained an approach to reality, associating process activities to text documents. With this advance we demonstrated that it is feasible to apply a new alternative for business processes mining of text documents by means of statistical language models, nevertheless, is necessary to make more tests.

The selection of an SLM as a text mining alternative permits to construct an ad hoc retrieval and assign weights to the terms that have more relevance and represent the occurrence of an event in the process [1].

The classifications and probabilities obtained from SLM become the inputs for the following part in our research in which we will obtain rules, patterns and relations cause-effect of the process.

6. Future Work

Future work will focus will on techniques for finding cause-effect rules, and for mining of evidence of incomplete or fractured processes, by means of obtaining the rules and patterns involved in text documents and determining its inference in these processes.

7. References

- [1] Xiaoyong Liu, W. Bruce Croft, "Statistical Language Modeling For Information Retrieval", 2005.
- [2] A. Tiwari, C.J. Turner, "A review of business process mining: state of the art and future trends", 2008.
- [3] Wil van der Aalst, Ton Weijters, Laura Maruster. "Workflow Mining: Discovering Process Models from Event Logs", 2004.
- [4] R. Agrawal, D. Gunopulos, F. Leymann, "Mining process models from workflow logs", 1998.
- [5] Leymann, W. Altenhuber, "Managing business processes as an information resource", 1994

- [6] Ah-Hwee Tan, "Text Mining: The state of the art and the challenges", 1999
- [7] Josh Froelich; Sergei Ananyan; David L Olson, 2005
- [8] Christopher D. Manning, "Foundations of Statistical Natural Language Processing", 2000.
- [9] David A. Grossman, Ophir Frieder, "Information Retrieval", 2004.
- [10] Leymann, F. y Roller, D., Business process management with FlowMark, 1994
- [11] Wil van der Aalst, A.H.M. ter Hofstede, and M. Weske. Business Process Management: A Survey, 2003.
- [12] William B. Cavnar, John M. Trenkle. "N-Gram Based Text categorization". 1994.
- [13] W.M.P. van der Aalst, H. A., Reijers, A.J.M.M Weijters, B.F van Dongen, A.K. Alves de Medeiros, M. Song, and H. M.W Verbeek. Business process mining: An industrial application. 2007
- [14] APQC, "Process Classification Framework", 2006.
- [15] W.M.P. van der Aalst, "Business alignment: using process mining as a tool for Delta analysis and conformance testing", 2005.

ARES: A Semantic Solution for Business Collaboration

Javier Chamizo
Departamento de Informática
Universidad Carlos III de Madrid
firstName.lastName@uc3m.es

Francisco García-Sánchez
Departamento de Informática
Universidad de Murcia
frgarcia@um.es

Yolanda Hernández
Indra Software Labs
yhernandezg@indra.es

Giner Alor-Hernández
Division of Research and Postgraduate Studies
Instituto tecnológico de Orizaba, México
galor@itorizaba.edu.mx

Juan Miguel Gómez
Departamento de Informática
Universidad Carlos III de Madrid
firstName.lastName@uc3m.es

Ruben Posada-Gómez
Division of Research and Postgraduate Studies
Instituto tecnológico de Orizaba, México
rposada@itorizaba.edu.mx

Abstract

Service industry is a major constituent of developed economies. Providing services is a knowledge-intensive, customer-specific, on-demand business. In the Business Information Systems domain, service-oriented architectures (SOA) and Web Services (WS), as an emerging approach to moving services into the computing world, are two of the most discussed topics in the areas related to Internet research. Today, technical questions of distributed computing, service description, composition, architectures and utilisation prevail, particularly, new forms and dimensions of intra- and inter-enterprise integration, coordination and availability of resources.

In this paper, we present the ARES platform, a new promising approach developed together with Indra Software Labs, one of the biggest Spanish software technology provider, integrates service-centric applications to provide a solution to the development of value-added tasks to be carried out by different organizations.

web-based business collaboration are emerging as a serious approach for carrying out large-scale business interaction, and are considered as a major step towards the next generation of E-Commerce [Aissi et al., 2002]. Therefore, numerous existing Business Process technologies [Van der Aalst et al., 2003] have to be integrated, multi-party collaboration has to be supported, and possibly occurring heterogeneities between the Business Processes of collaboration partners have to be resolved. The ARES project will develop a coherent architecture for this, that will be provided as a Web Service and which is aligned with Semantic Annotations for Web Service Description Language [Verma and Sheth, 2007] (SAWSDL¹).

The remainder of the paper is organized as follows. In section 2, we discuss the State of Art in business process collaboration approaches. In section 3, we describe the base and structure of ARES project; and finally, section 4 concludes the paper and outlines our future work.

2 State of Art

Business Process technologies shall allow organizational entities to formalize specific processes of their business in order to support automation of these, thereby enhancing of production or other business areas. As such technologies promise significant improvements in IT-based business solutions, several techniques for Business Process representa-

1 Introduction

The aim of the ARES project is the development of a generic framework and technologies for semantically driven business collaboration over the Internet by integration of business processes along with mediation facilities for resolving possibly occurring heterogeneities. Techniques for

¹Semantic Annotations for WSDL:
<http://www.w3.org/2002/ws/sawsdl/>

tion and management have been developed. At this point in time, there are around 250 different Business Process technologies and frameworks existing which intend to enable integration and automation of business processes, both within and across organizations [Dayal et al., 2001]. Most of these trends have been developed within joined academic and industrial boards whereof quite a few have become quasi-standards with regard to their usage throughout business solutions.

The problem currently faced within IT-solutions for business support is the Integration Problem [Patil et al., 2003]: this means that techniques are needed that allow making different resources interoperable, thus supporting automated business collaboration that might have heterogenic resources - both data resources as well as different, not a priori, compatible business process specifications and technologies. Another challenge is to make use of the Internet as a world wide information and communication platform for business interaction [Andrews et al., 2003]. Therefore, the Web and Web Services, especially the Semantic Web along with Semantic Web Services are considered as a technology that can solve the problems currently faced within IT business solutions.

The Internet as existing at this point in time provides a world wide information repository and communication infrastructure. Current web technologies around URIs, HTTP, HTML, and XML allow making information accessible world wide, and the Internet has become an important resource for information interchange. For automated information processing of web-content, the information on the Web have to be provided in a structured manner, and, due to the nature of the Internet as a decentralized and heterogeneous system, the information have to be semantically described in order to allow meaningful processing by machines. Thus, Tim Berners-Lee, the inventor of the World Wide Web and director of the W3C, has brought up the vision of the Semantic Web [Berners-Lee et al, 2001] as the next generation of the Internet wherein technologies shall be developed that support automated web-content processing on top of the existing web technology stack. Key technologies have been identified for the Semantic Web: Ontologies [Gruber, 1993] shall provide machine-readable terminology definitions on the basis of a shared domain conceptualization, allowing semantically enhanced and thus meaning-preserving information exchange over the web between different agents (which can be individuals, organizations, or machines); and Web Services as computational functionalities that can be invoked and used over the internet, thus combining the benefits of the Internet as a world-wide information repository with computational facilities.

Research and development activities around the Semantic Web commenced in the late 1990s, resulting in first

specifications of Semantic Web Languages, like RDF² and OWL³ are recommendation of the W3C. Initial Web Services technologies around UDDI, SOAP, and WSDL have not been successful in making the promise of Web Services become reality [Baida et al., 2005]. Because of this, currently major research efforts concentrate on the combination of Semantic Web technologies and Web Services, establishing the notion of Semantic Web Services. In order to provide means for automated discovery, composition, execution, and collaboration of Web Services, the general approach followed in recent research efforts is to develop concise, semantic description frameworks of aspects relevant within Web Services, and to create 'intelligent', i.e. inference-driven techniques to enable automated usage and management of Web Services on basis of the semantic service descriptions. For a summary on the state of the art within Semantic Web Services [Cardoso et al., 2006] we refer to the following section.

Regarding the applicability of Semantic Web and especially Semantic Web Services for IT business solutions with special attention to the Integration Problem [Vinoski, 2003], the general idea is that businesses or organizations provide their functionalities as Web Services and that semantic driven technologies ensure interoperability of these services, thus allow establishing of automated business collaboration [ebXML Commitee, 2002]. Therefore, the challenge for suitable Semantic Web Service technologies is to provide mediation facilities for resolving possible occurring heterogeneities between business entities that ought to collaborate. According to the types of mismatches that might occur, three levels of mediation are distinguished: (1) Data Level Mediation, referring to heterogeneities within the information interchanged within a business interaction, (2) Protocol Level Mediation, concerned with mismatches on the interaction protocols of services that are ought to interact, and (3) Process Level Mediation, that is concerned with the compatibility of the individual business processes of collaboration partners. In order to allow automated interaction and collaboration of several business entities, the mismatches on all mediation levels between their respective services have to be resolved. When this is accomplished, a control functionality for multi-party collaboration on basis of Web Services is needed in order to execute business collaborations over the Web semi-automatically.

There are several approaches existing that serve as starting points for the development of the ARES technology. Apart from existing Business Process technologies [Arkin et al., 2002], there are approaches for mediating between them, there are most recent initiatives that work on multi-party collaboration technologies for Web Services, and there are means for Data Level Mediation under de-

²Resource Description Framework: <http://www.w3.org/RDF/>

³Web Ontology Language: <http://www.w3.org/TR/owl-features/>

velopment in SAWSDL-related research efforts, and approach are proposed for mediation of process descriptions - although on a very rudimentary level. Thus, the research within the ARES project will focus on the further development of existing technologies and their integration into a coherent system for semantically driven business collaboration.

In the following, we outline the state of the art in research fields related to the ARES technology and point out the challenges and risks to be addressed within the project, and we outline the proposed technical solution for the ARES technology.

3 The ARES project

The main problem of Business Information Systems Integration is the fact that vendors such as Oracle, PeopleSoft and SAP build sophisticated, process-centric solutions on complex information structures implemented in relational databases for the business organizations that manage the enterprise resource, as discussed in [Beltz, 2003]. In this same publication, the reason for which information lacks such comprehensively integrated vendor solutions are presented, namely: the concept of information as a resource being new compared to decades-old business processes such as dual-entry accounting and the process discipline imposed by each Information System generating friction in most IT shops, particularly if it involves short-term pain for high-visibility, business-sponsored projects.

Actually, during the ARES project requirement definition phase, a similar outcome was achieved. The problem of integrating different Information Systems was mostly due to the high heterogeneity among technologies, trying to be integrated in the Enterprise Service Bus (ESB) [Schmidt et al., 2005] of the company. Each system constitutes a stand-alone silo with insufficient integration with the overall enterprise IT processes and this problem would remain even worse if it was not because of their exposure as Web Services to the ESB, which significantly improves the interoperability between them.

The ARES project exactly addresses these challenges, providing a web-service based technology for business collaboration along with suitable mediation facilities. The aim is to develop a framework for a Business Process collaboration environment that allows adapting business processes of collaboration partners, defining a valid multi-party interaction protocol, and providing the required Mediation facilities. More precisely, the objectives of the project are:

- Definition of a *Business Collaboration Framework* that specifies the approach for semantically driven business collaboration, the overall architecture of the ARES technology and the core technologies of the system

- A *Business Choreography Module* that allows establishing of valid interaction protocols for multi-party collaborations
- A *Mediation Module* that provides the required mediation facilities in order to resolve possibly occurring heterogeneities on the data, protocol, and process level.
- Specification of *Business Process Adapters* that transform existing Business Process technologies into the ARES process representation format, thus allowing organizational entities to use the ARES
- Specification of the *ARES Architecture* that integrates the ARES components and manages the system functionality.

The ARES technology will be based on semantic technologies currently under development within Semantic Web Services. This means that a coherent technology framework will be developed, with special regard to semantic formalisms that allow mediation on an abstract, general level. The ARES technology will be integrated as a core Mediation Service into the Semantic Annotations for WSDL [Kopecky et al., 2007], the Semantic Web Services research effort of a joined initiative of European research institutions around the Semantic Web that defines an overall framework that covers relevant aspects of Semantic Web Services and subsequently develops technologies for using and managing Semantic Web Services. This at first ensures the compatibility of the ARES technology with emerging Semantic Web technologies and, secondly, it allows aligning the ARES technology with other technical building blocks developed around SAWSDL. The ARES project is a complementary research effort to SAWSDL, providing a significant contribution to ongoing international research on the one hand, and, on the other, applying and integrating other SAWSDL-enabled technologies into a fully fledged technology framework for Semantic Web Services with special support for business collaboration.

Hence, our main goal was to summarize a number of requirements for integration and defining a high-level semantics process language scenario, which could model the heterogeneity of the Information Systems but eventually provide a solution to locate the particularities of each of them, mostly through a semantic markup language. In the next sections, we will discuss our approach in more detail.

3.1 Problem Settings

The initial setting for business collaboration is that there are several organizational entities that can either be independent businesses, sub-departments inside an organization, a group of companies organized in a supply chain, or

any other setting of entities. These entities want to collaborate in order to achieve some common business goal.

Particularly, in our scenario, those are a number of Information Systems connected through an ESB thanks to the Web Service interfaces. The main problem we face is that most of these systems do not know about each other and do not have a centralized means of "getting to know each other".

With the ARES technology, we aim at providing that common conceptual background to automate or at least, ease collaboration between the business entities i.e. the Information Systems. The approach consists in providing a generic solution that works on a higher semantic level and it is thus independent of particular application areas. For that, we follow a three step procedure. First, we model a high-level ontology with the main concepts of the major IT entities involved. Secondly, we generate metadata out of the ontology by simply using a Semantic Annotation technology for Web Services, such as SAWSDL. Finally, we annotate the actual Web Services that represent each of the Information Systems all over the ESB.

3.2 ARES Architecture

The ARES architecture is the actual implementation of the three steps mentioned in the previous section. Firstly, the architecture builds on a conceptual model, an ontology, which encompasses the main building blocks of the architecture (Step 1). The actual ontology builds on a number of concepts related to the architecture, namely: Information System, Process, Web Service Interface, Data Model element and Non-functional properties, such as security, transactivity and fault-tolerance.

Secondly, according to a given system, we generate metadata to describe the main features of the Information System, for example, if the system is an Enterprise Resource Planning (ERP) system, dealing with money, productive capital, people and stock of goods, the metadata should reflect it. This is an iterative process, comprised as Step 2, which ties the logical meta-model with the actual metadata, transforming the physical into the logical.

Finally, we actually annotate the Web Service with SAWSDL. This is done almost automatically since if the process is too complex, the user (the annotator) might become overloaded and abandon the process. However, the user is necessary in the loop in order to avoid misinterpretation of concepts and misalignment which easily emerges. A final view of the distribution of the architecture is shown in Figure 1.

The ARES platform is an ongoing work implementation of the ARES architecture, which integrates and manages the ARES components and provides the metadata for the Web Services and also, a metadata management facility.

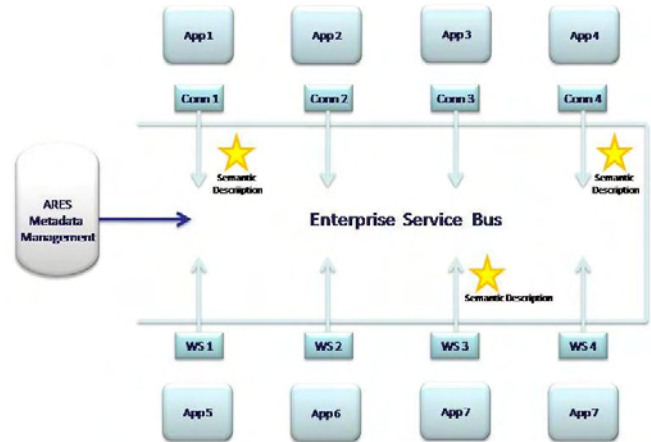


Figure 1. ARES Distribution Architecture Overview

The advantages of using ARES versus conventional mainstream relational database technology are as follows. As discussed in [Beltz, 2003], it is a difficult area to model, since sub-typing hierarchies are deep, recursion or redundancy is rampant and relationships are very complex. Additionally, there is a need to expressively describe those relationships and support abstractions from the physical view to the logical view. A highly ontology-centric approach, emerging as a response to the difficulties of meta-modeling the IT problem domain, can provide a value-added solution. For example, the ITIL concept of a "configuration management database" uses configuration item (CI) as a general abstraction for any internal IT process, whereas ITIL does not seem to call for a robust relationship model i.e. there are no valid relationships between CIs, since they are not formally and strongly defined.

4 Conclusion and Related Work

As the use of Semantic Web Services grows, the problem for searching, interacting and integrating relevant services is becoming increasingly a hurdle for the leverage of existing Semantic Web technologies which have reached a certain level of maturity. In this paper, we have proposed a solution and a proof-of-concept and effective implementation automated business collaboration between existing Semantic Web Services. The forthcoming of our approach are mainly two, namely: striving for a rich and ease-of-use interaction to understand service choreography and the ability to encompass with the existing Semantic Web Services execution environments.

There are several related approaches to the ones pre-

sented in this paper. The idea of integrating complex, stateful interactions among services has been long studied. As a result, several form of business process integration, orchestration, or choreography model has been proposed. These include WSCI, BPML, XLANG, WSFL, WSCL, BPSS, the Web Services Architecture, BPEL4WS, Siebel [Curbera et al., 2002] and most recently WSMX [Haller et al., 2005]. In [Mandell et al., 2003], a adaptation of BPEL4WS to the Semantic Web is proposed since the authors argue that, in order to obtain a seamless interoperability, it is necessary to embrace many approaches derived from the Semantic Web. In [Gibbins et al., 2003], the authors outline their experiences of business process collaboration based on the integration of OWL-S [OWL-S, 2004] based Web Services and an agent communication language (it is done to separate the domain-specific and the domain-independent aspect of communication).

Finally, our future work will focus on finding use cases and real-world scenarios to validate the efficiency of our approach and determine the feasibility of the semantic business collaboration through mediation to stablish a choreography between services.

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References

- [Aissi et al., 2002] Aissi, S.; Malu, P.; Srinivasan, K., "E-business process modeling: the next big step," Computer , vol.35, no.5, pp.55-62, May 2002
- [Andrews et al., 2003] T. Andrews, F. Curbera, H. Dholakia, Y. Goland, J. Klein, F. Leymann, K. Liu, D. Roller, D. Smith, S. Thatte, I. Trickovic, and S. Weerawarana. Business Process Execution Language for Web Services, <http://www-128.ibm.com/developerworks/library/specification/ws-bpel/>, BPEL4WS version 1.1, May 2003.
- [Arkin et al., 2002] Assaf Arkin, Michael Dailey, David Blondeau, Ismael Ghalimi, Wolfgang Jekeli, Stefano Pogliani, Matthew Pryor, Karsten Riemer, Howard Smith, Ivana Trickovic, and Stephen A. White. Business Process Modeling Language. <http://www.bpmi.org/BPML.htm>, November 2002.
- [Baida et al., 2005] Z. Baida, J. Gordijn, H. Akkermans, H. Saele, and A.Z. Morch. Finding e-Service Offerings by Computer-Supported Customer Need Reasoning. *Int. J. E-Business Research* 1(2) 91-112, 2005.
- [Beltz, 2003] Charlie Beltz. Enterprise Resource Planning for IT. *Business Integration Journal*. November, 2003.
- [Berners-Lee et al, 2001] Berners-Lee, T., Hendler, J. y Lassila, O. (2001, May). The Semantic Web. *Scientific American*, 284(5), 34-44.
- [Cardoso et al., 2006] Cardoso, J.; Sheth, A. Semantic Web Services, Processes and Applications. Springer-Verlag New York Inc. (May 2006)
- [Curbera et al., 2002] Curbera, F., Goland, Y., Klein, J., Leymann, F., Roller, D., Thatte, S., Weerawarana, S. Business Process Execution Language for Web Services. <http://www.ibm.com/developerworks/library/ws-bpel/>
- [Dayal et al., 2001] Dayal U, Hsu M, Ladin R (2001) Business process coordination: state of the art, trends, and open issues. In: *VLDB Conference*, pp 311, Rome, Italy
- [ebXML Committee, 2002] ebXML Collaboration Protocol Profile and Agreement Technical Committee, Collaboration-protocol profile and agreement specification version 2.0, September 2002, www.oasis-open.org/committees/ebxml-cppa/documents/ebCPP-2.0.pdf
- [Gibbins et al., 2003] Gibbins, N., Harris, S., Shadbolt, N. (2003). Agent-based Semantic Web Services. In *Proc. of the 12th Int. World Wide Web Conference*, May 2003.
- [Gruber, 1993] Gruber, T. (1993). A translation approach to portable ontology specifications. *Knowledge Acquisition*, 5(2), 199-220.
- [Haller et al., 2005] Haller, A. Cimpian, E. Mocan, A. Oren, E. Bussler, C. (2005). WSMX - a semantic service-oriented architecture. In *Proc. of the 3rd Int. Conference on Web Services*, July 2005.
- [Kopecky et al., 2007] Jacek Kopecky, Tomas Vitvar, Carine Bournez, and Joel Farrell. SAWSDL: Semantic Annotations for WSDL and XML Schema. *IEEE Internet Computing*, 11(6):6067, 2007.
- [Mandell et al., 2003] Mandell, D.J. and McIlraith, S.A. (2005). Adapting BPEL4WS for the Semantic Web: The Bottom-Up Approach to Web Service Interoperation. In *Proc. of the 7th Int. Symposium on Wearable Computers*, October 2003.

- [OWL-S, 2004] The OWL Services Coalition. OWL-S: Semantic Markup for Web Services. Version 1.1, November 2004. <http://www.daml.org/services/owl-s/1.1/>.
- [Patil et al., 2003] Patil, S.; Newcomer, E., "ebXML and Web services," *Internet Computing, IEEE* , vol.7, no.3, pp. 74-82, May-June 2003
- [Schmidt et al., 2005] M.-T. Schmidt, B. Hutchison, P. Lambros, and R. Phippen, The Enterprise Service Bus: Making Service-Oriented Architecture Real, *IBM Systems Journal* 44, No. 4, 781798 (2005, this issue).
- [Van der Aalst et al., 2003] W.M.P. van der Aalst, A.H.M. ter Hofstede and M. Weske, Business process management: a survey, *Proceedings of the 2003 International Conference on Business Process Management (BPM 2003)*, *Lecture Notes in Computer Science* vol. 2678, Springer-Verlag, Berlin (2003), pp. 112.
- [Verma and Sheth, 2007] K. Verma and A. Sheth, "Semantically Annotating a Web Service," *IEEE Internet Computing*, vol. 11, no. 2, 2007, pp. 8385.
- [Vinoski, 2003] S. Vinoski, Integration with web services, *IEEE Internet Computing*, November-December 2003.

Learning of Market Prices based on Temporal Differences for a TAC-SCM Agent

César Flores

Centro de Sistemas Inteligentes y Robótica
Tecnológico de Monterrey
Av. Eugenio Garza Sada #2501
Monterrey, Mexico
a00781557@itesm.mx

Leonardo Garrido

Centro de Sistemas Inteligentes y Robótica
Tecnológico de Monterrey
Av. Eugenio Garza Sada #2501
Monterrey, Mexico
leonardo.garrido@itesm.mx

Abstract

In this work we present a learning method based in temporal differences for learning the market prices for a supply chain manager in the TAC-SCM game scenario. The agent has two learning methods based in temporal differences. On one hand he learns the short term tendency of the market to adjust their prices proportionally to the rise or drop. He also learns a value of probability to elevate the prices proportionally to the excess of real demand.

1. Introduction

The Supply Chain Management is a problem where AI researchers have a special interest due to it contains a big variety of complex problems that have the characteristics for being solved with some of the AI branches.

One of the most fruitful applications of AI in the supply chain exists in the international tournament TAC-SCM (Trading Agent Competition - Supply Chain Management) [4].

The scenario of the game was designed to capture most of the problems wrapped in the implementation of a dynamic supply chain, but maintaining simplicity in the game rules to motivate the participation of people that are not experts in the topic.

The scenario was designed jointly by a team of researchers of the University Carnegie Mellon and the Swiss Institute of Computer science.

2 TAC-SCM Game Environment

The TAC-SCM Game Environment consists in a supply chain of computers. In each game participate six agents; each one represents a manufacturer that produces 16 kinds of computers. The agents compete during 220 days and they

try to maximize their utility. The winner is the agent that in the end of the 220 days has more money in his bank account. All the aspects of the game (bank, clients, suppliers, etc.) are simulated by a server.

The scenario [6] is interesting for the application of uncertainty systems due to the behavior of suppliers, clients and other aspects that are simulated by the server are dominated by randomness. And this randomness is presented not only during the game but also from a game to another game. That is for example that exist games where the client demand is High but others where it is low. At the same time the client demand varies during a same game.

3 Problem Description

One of the principal problems in the supply chain is optimize or solve for the optimal price policy. This is an important aspect because the agent needs knows the marked price in order to sell at the price most expensive of the marked in the client sector.

The TAC-SCM protocol that was implemented for the client sector is the next:

1. At the start of each day the clients send the same bundle of requests to the six agents. These requests specify the computer type, desired quantity, reserve price, due date and penalty for each day of delay.
2. The six agents analyze the requests and based in their capacities and preferences must decide to respond or not to each one of the requests. If they respond, they must send at the end of the day an offer that specifies the sell price. This sell price must be less or equal to the reserve price specified in the request because otherwise the offer is rejected.
3. At the start of the next day the clients finalize the cycle by sending the order to the agent that offered the cheapest price.

4. The winner agent of each order must produce and sent it to the client before de due date. If the agent send the order after the due date, he must pay the specified penalty for each day of delay. After the day 5 of delay, the order is cancelled.

As you can see, is the agent that wins the order is the agent that offer the cheapest price. Nevertheless the agent needs not only offer the cheapest price, but also this price must be profitable for the agent. It is here were a good learning policy of prices plays an important role.

4 Related Work

The problem of choosing the correct bidding strategy has been studied for many TAC-SCM teams. These strategies come from simple to very sophisticated. PackaTAC [5] uses a simple heuristic to find the correct sell price. Is is based only in three bounds that are upper, soft lower and hard lower. These bound are based in some game variables as reserve price and previous days prices.

Mercator [8] for example departs from the hypothesis that all the offers will be accepted. And with the help of machine learning treats this problem. Besides this, it uses a probability scheme to treat the overbidding. Some teams as TacTex-03 [9] study the probability of an order. That is, it tries to find with what probability will be accepted an offer and uses this probability to find the computer marked prices.

Botticelli bidding strategy [2] uses a similar probability to establish the sell price. It uses two sources of information for modeling this relation: the maximum an minimum prices that were paid for the client the previous day and the orders that were resulted from Botticellis offers. A very nice approach of marked prices was made by Jackaroo team [13]. This team uses game theory in order to find the equilibrium production.

5 Temporal Difference Learning

The problem of learning the marked prices can be attacked with a strategy of temporal difference learning (DT). This method has the characteristics described by Sutton [11]. The DT method uses the error or difference between successive predictions for generating a feedback signal according to the changes between them.

These methods have the characteristic of not to require a model for the environment that the agent seeks to learn. This method operates in the states domain, not in the actions domain.

Also, the DT methods are optimums for environments in real time. The general model of TD learning is the next:

$$V(S_t) \leftarrow V(S_t) + \alpha[r_{t+1} + \gamma V(S_{t+1}) - V(S_t)] \quad (1)$$

Where r_{t+1} is the reward observed for the state t+1 and $V(S_t)$ are the observed values for the states t. Finally α and γ are the learning constants. This method has been applied to game problems by Tesauro [12].

In this work we used only the general idea of TD learning. We implemented a model adapted to the particularities of the TAC-SCM game. Learning Price Strategy Each day the agent must determine the assigned price for the client requests. Our agent applies two reinforce learning strategies. The objective is to predict the marked price for each type of computer in the game:

1. To predict the price of the day, searching for taking advantage of the marked tendency.
2. Application of a random surcharge, according with the exceeded capacity of the line production.

In the main strategy of price prediction, the agent tries to determine an acceptable price and to capitalize in the sense of short term tendency of the market. If it is determined that a rising tendency exists, then the price increases. If it is determined that a bear tendency exists then it adjusts its prices to the drop.

For determining the price, the agent calculates a pondered average of the last day based in the maximal price and minimal price offered. The weights for each factor are learned from the marked tendency. The model for determining the price is the next:

$$P(t) = w_l P_{Low}(t-1) + w_h P_{High}(t-1) \quad (2)$$

In the model described in the equation 2, the sense and intensity of the learned tendency by the agent is memorized in the parameters w_l y w_h . These parameters are normalized before enter in the equation. $w_l + w_h = 1$. In this way for example, when a strongly rising tendency exists, $w_h \rightarrow 1$ y $w_l \rightarrow 0$, as consequence the predicted price will be the maximal price of the last day. In a similar way, in bear tendencies, the current price will spread to the minimum of the previous day.

The parameters w_l y w_h are adjusted according to the temporal relative difference $D_p(t) = (P(t) - P(t-1))/P(t-1)$, and they are adjusted with the following learning rule:

$$w_l = w_l + 10|D_p(t)| \quad (3)$$

For $D_p < 0$, that is, when the demand decrease and,

$$w_h = w_h + 10|D_p(t)| \quad (4)$$

For $D_p > 0$, when the demand increase.

In both cases, the factor of 10 is used as scale to make significant the changes in a reasonable period of time. On the other hand, a second learning strategy was introduced to capitalize in a more appropriate way the rise opportunities of the market.

In this strategy the agent increases the price of a type of computers aleatorily according to the change in the gain margins and to the surplus of production cycles. In this way, for each type of computer, the price is In this way, for each type of computer, the price rises in $q/15$ with a probability Q (That which is in increments between 0 and 6.66 %).

If in day $t-1$ the production cycles were higher than the available ones, the agent adjusts the probabilities values Q for each component according to the following rule:

$$Q(t) = Q(t-1) + \alpha + (M(t) - M(t-1)) \quad (5)$$

where α is a learning constant with a value of 0.05 and $M(t)$ is the margin of contribution of the product in the time t . In the case when the production cycles were lower than the available ones, the probability is adjusted with a multiplicative rule, to make faster the decrease and avoids High prices:

$$Q(t) = Q(t+1) - \alpha(2 + R) \quad (6)$$

Where R is the reason of used cycles between available cycles. That is if $R < 1$, the agent knows that he needs decrease the probability Q .

With this method not only the agent manages the sell price but also he gets indirectly regulates the use of the line production and avoids delays, with the diminution of the number of orders and the increase in their profit.

6 Design of Experiments

For the construction of our agent we used the start agent released by the actual champion team TacTex [10]. The agent dont have the heuristiques used by the team for win the TAC-SCM tournament, but his infract Although the agent doesnt have the heuristics used by TacTex for win the championship, his infrastructure and general logic has the highest performance among the teams whose source code is available. It is also very documented and as consequence very easy of understand.

With this work we modified only the part of the start agent corresponding to the offers that the agent sends to the clients. That is, the only module that we touched is the corresponding to the demand manager that in the code is known as SimpleDemandManager.

Like this we assure that the difference in the performance between the start agent and the modified start agent is due only to the learning policy of prices.

We took the ideas of Borghetti et al. [3] to be able to eliminate the biggest quantity in randomness of the server but without modify the randomness of the clients.

In this way it is possible to study our improvement whose learning has a lot to do with this behavior more concretely. To evaluate the agent improvement we implemented a scenario where the modified start agent competes against the start agent without any modification. The remaining four slots were used by three agents TacTex 2006 and a Dummy agent. TacTex agent, like it was already mentioned, is the world current champion. The idea is that these last agents would create a strong competitiveness in the game so that our policy could be better evaluated. After each modification we run experiments in order to test if there exist a improvement or not. The games had duration of 220 days. Nevertheless, to accelerate the simulation, the duration of the days were decreased of 15 seconds that it is the official time to only 5. It was validated that all the agents had the time of deciding. But we must clarify that the time varies depending of the computer.

We designed another scenario with low competitiveness. We make this to test if the learning method works in this kind of scenario. And like this assure that the agent can adapt to other kind of agents and not only to TacTex agents. In order to create a scenario with low competitiveness the only thing that que made was to replace the TacTex agents by the default Server agents called dummys. Here we decrease the day duration to only three seconds.

We ran 30 games in each kind of scenario. Te results will be present in the tables of the next section.

7 Results and Discussion

Before explain the results we must explain the two names that will be used for the agents in question. The first name is StartAgent that is the agent released by TacTex team without any modification. The second name is BorregosSCM, name of our team that will participate by first time in the TACSCM tournament this year. This agent has the implementation of the learning method that the StartAgent has not. We present firstly a resume of the 30 games that we ran for each scenario in two tables.

In 1 y 2 we can observe that the difference in the gain between BorregosSCM and StartAgent is much greater when the competitiveness is not so important. In the case of high competitiveness, the improvement in the results is of 6.27M, while in the other case is of 13.71M.

This difference can be attributed to the fact that TacTex agents have a good learning model and are highly competitive, forcing the prices to go down.

An important similarity between both situations is that while the amount of orders received by BorregosSCM is much smaller in comparison to StartAgent and that the ma-

Table 1. Results in a High Competitive Marked

Player	Orders	Revenue	Material	Penalty	Result
BorregosSCM	7 120	130 400 549	119 063 087	1 570 868	7 864 423
BorregosOriginal	8 986	126 466 287	120 567 458	2 430 397	1 596 188

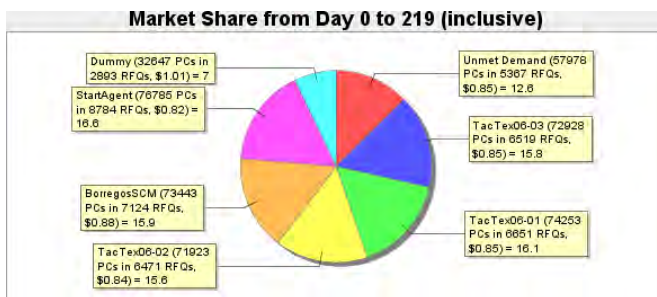
Table 2. Results in a Low Competitive Marked

Player	Orders	Revenue	Material	Penalty	Result
BorregosSCM	7 190	154 305 738	96 448 539	373 723	56 459 527
BorregosOriginal	8 005	140 650 626	95 257 010	1 434 942	42 748 170

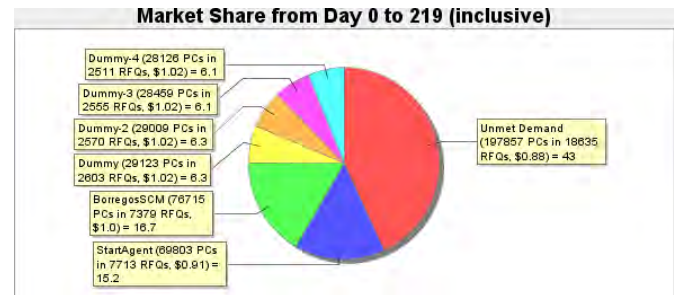
material acquired by the two agents is similar, BorregosSCM has better revenue than StartAgent. This is an indicator of higher prices per unit and therefore a better bidding strategy. It can be also appreciated that the penalties are much less important in BorregosSCM. We can conclude that a good bidding strategy does not only have a benefit in sales but, as the amount of orders received is smaller, it will be more feasible to fulfill all of them without delays. In other words the penalty received by BorregosSCM is smaller than the penalty received by StartAgent. This also suggests a possibility for improvement in the supply chain, in order to optimize inventories and storage expenses. Nevertheless a experience TAC-SCM player can see that this number is still very big.

For the rest of this section one game of each scenario was used. These two game represents strongly the behavior of the agents.

In the figures 1 and 2, the results of the market share of all products for each scenario are shown. It can be appreciated from these graphs that high competitiveness is inversely related to number of demands that were not taken care of. We can see in figure 1 for example that the unmet demand is smaller than Figure 2. This is because in figure 1 there are three TacTex agents that covers a big demand percentage contrary to the Dummies capacity in figure 2.

**Figure 1. High competitiveness**

To demonstrate the learning process, in the figures 3 and

**Figure 2. Low competitiveness**

4 is shown the behavior of the price strategy of the two agents in competition, both in high and low competitiveness. It can be observed in both images that the prices with the BorregosSCM learning agent are well above those found with StartAgent. Also, we can appreciate that BorregosSCM is able to adapt to the type of market in each situation. This is particularly useful in high competition environments like the one shown in 3, since in this type of scenario the competitive teams forces the prices downwards and with that the gain margins are reduced drastically. It is clear that the performance with the BorregosSCM learning model is higher compared to the original one.

We can see in both figures that BorregosSCM can follow the form of the distribution. We see also that figure 4 is more distributed than figure 3. This is because the behavior of the Dummies agents is governed by some randomness.

Finally we can see in figure 5 and 6 a plot where we compare our predicted price, called Price Strategy, and the Average Price for product 1 in both scenarios. The average price was acquired from the marked report. To make this, we changed the default configuration in the server from 20 days to 1 day. This price was only used to plot the Average Price curves. This is not used in the learning process.

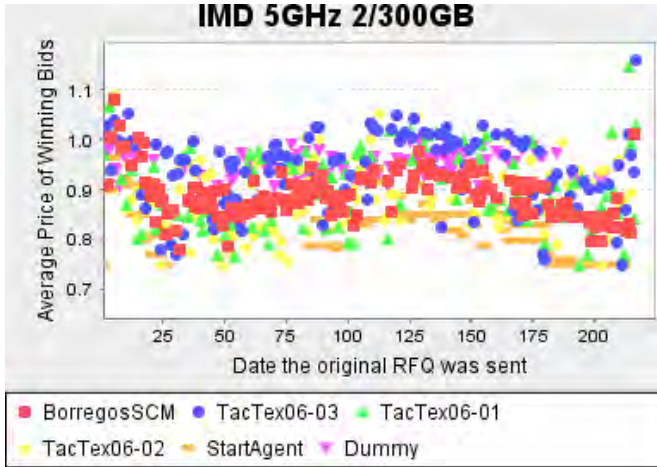


Figure 3. High competitiveness

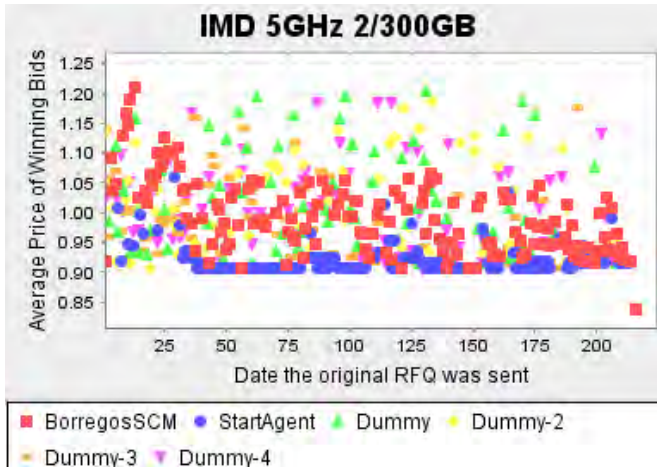


Figure 4. Low competitiveness

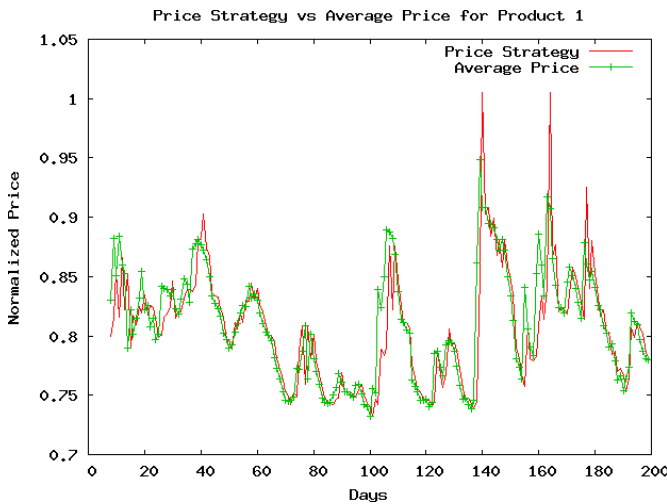


Figure 5. High competitiveness

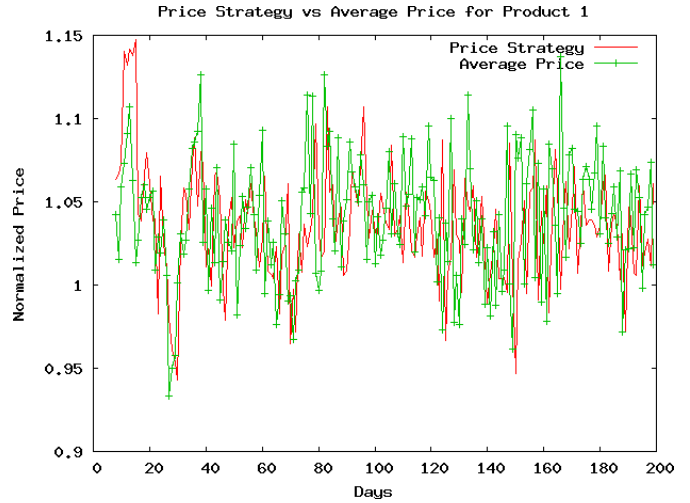


Figure 6. Low competitiveness

8 Conclusions and Future Work

The reader can see that in the figure 5 the Strategy Price is very similar to the Average Price. That is a prove that our strategy is very good. But in fact it has a little problem. If the reader see with detail, he will discover that the agent sells a little cheaper than the average price in almost all the game. And that means that another agent sells above the average price in order to have an equilibrium. This last agent is surely in this figure a TacTex agent.

In the figure 6 we can see that our price strategy is not very similar to the average price. This is because of the dummies behavior, that has certain degree of randomness.

The start idea was that the temporal differences learning method could give us a wide advantage in any kind of scenario. Nevertheless, as we observed in the results the learning method performs better in a low competitiveness scenario. In spite of the above-mentioned, the modified agent was able to win the StartAgent agent in two totally opposed kinds of scenarios, what speaks well of the established bidding strategy. In the carried out experiments, the improved agent obtains a yield of up to 6.27M in a highly competitive scenario and of 13.71M in a scenario of low competitiveness. In the reach of the present work a strategy of prices learning was designed, however, the application of this kind of learning can be applied in many of the problems than we must solve in order to have a successful agent for the supply chain management.

In fact our team, BorregosSCM, has just started the development of a TAC-SCM agent. We hope that with the time we will improve all the start agent modules in order to have a good participation in the TAC-SCM tournaments. We saw that the agent behavior has improved considerably only with the Demand Manager alteration. Therefore we

hope that with the improvement of other important modules as Supply Manager the agent can reach a very good behavior.

We have experimented only with two learning strategies. But in fact, as was documented in the Related Work section, there are many others strategies that surely will improve our agent if we mix them with our actual strategies. That is one of our future works.

But in fact the principal reason of choosing TAC-SCM as research scenario is that game theory can be applied here. We want to explore some ideas from game theory as Markov equilibrium or Stochastic Systems in order to improve our agent behavior. Principally we want attack with this strategies a better learning approach for BorregosSCM agent.

References

- [1] I. M. Author. Some related article I wrote. *Some Fine Journal*, 99(7):1–100, January 1999.
- [2] M. Benisch, A. Greenwald, I. Grypari, R. Lederman, V. Naroditskiy, and M. Tschantz. Botticelli: A supply chain management agent designed to optimize under uncertainty. *SIGecom Exchanges*, 4(3):29–37, 2004.
- [3] B. Borghetti, E. Sodomka, M. Gini, and J. Collins. A market-pressure-based performance evaluator for tac-scm. *AAMAS-06 Workshop on Trading Agent Design and Analysis (TADA/AMEC)*, 2006.
- [4] J. Collins, R. Arunachalam, N. Sadeh, J. Eriksson, N. Finne, and S. Jansonc. The supply chain management game for the 2007 trading agent competition. December 2006.
- [5] E. Dahlgren and P. R. Wurman. Packatac: A conservative trading agent. *SIGecom Exchanges*, 4(3):38–45, 2004.
- [6] J. Eriksson, N. Fine, and S. Janson. Evolution of a supply chain management game for the trading agent competition. *AI Communications*, 2006.
- [7] A. N. Expert. *A Book He Wrote*. His Publisher, Erewhon, NC, 1999.
- [8] I. Kontogounis, K. Chatzidimitriou, A. Symeonidis, and P. Mitkas. A robust agent design for dynamic scm environments. 4th Hellenic Joint Conference on Artificial Intelligence (SETN06), 2006.
- [9] D. Pardoe and P. Stone. Tactex-03: A supply chain management agent. *SIGecom Exchanges*, 4(3):19–28, 2004.
- [10] D. Pardoe, P. Stone, and M. VanMiddlesworth. Tactex-05: An adaptive agent for tac scm. *AAMAS-06 Workshop on Trading Agent Design and Analysis (TADA/AMEC)*, 2006.
- [11] R. S. Sutton. Learning to predict by the methods of temporal differences. *Machine Learning*, 3:9–44, 1988.
- [12] G. Tesauro. Practical issues in temporal difference learning. In J. E. Moody, S. J. Hanson, and R. P. Lippmann, editors, *Advances in Neural Information Processing Systems*, volume 4, pages 259–266. Morgan Kaufmann Publishers, Inc., 1992.
- [13] D. Zhang, K. Zhao, C.-M. Liang, G. B. Huq, and T.-H. Huang. Strategic trading agents via market modelling. *SIGecom Exchanges*, 4(3):46–55, 2004.

Use of Intelligent Emotional Agents in the Animation of Autonomous Virtual Creatures

Héctor Rafael Orozco Aguirre,
Félix Francisco Ramos Corchado,
Luis Alfonso Razo Ruvalcaba,
Jaime Alberto Zaragoza Rios
Centro de Investigación y
de Estudios Avanzados del I.P.N.
Unidad Guadalajara
Av. Científica 1145, Col. El Bajío
45010 Zapopan, Jal., México
{horozco, framos}@gdl.cinvestav.mx

Daniel Thalmann
École Polytechnique Fédérale de Lausanne
EPFL IC ISIM VRLAB
Station 14 CH-1015
Lausanne, Switzerland
daniel.thalmann@epfl.ch

Abstract

Recently several psychologists have begun the exploration and study of emotions to explain better their functioning. The obtained results give evidence that emotions have an important impact on thinking, judgment, reasoning, memory and decision making of human being. In this work we describe a new faculty called Artificial Emotional Intelligence (AEI), and we propose a model based on Emotional Intelligence (EI) to develop a new approach to the problem of mood and emotion control in the animation of avatars or virtual creatures (VC). We use intelligent emotional agents to control the behavior of the avatars in 3D virtual environments. Our proposal about AEI relates cognitive processes of avatar involving emotional information. These processes include mainly: appraising and expressing emotions and regulating emotion in the self and others, and using emotions in adaptive ways.

1. Introduction

All interactive applications, such as computer games, video games and collaborative virtual environments, are in need of virtual entities that exhibit a complex and believable behavior. But actually, the behavior of the VC in current applications and systems is still very artificial.

Inspired by the psychological models of emotion, many researchers have recognized in Artificial Intelligence (AI) the importance and utility for improving complex, dynamic and interactive virtual environments with the help of com-

putational models of emotions. However, most of designed computational models of emotion only represent specific situations and respond in predetermined way to them.

Emotions are an important aspect in the functioning of the human mind. A lot of studies about emotions exist and there are several models of emotion proposed in the literature, unfortunately we cannot know for sure which models are correct, because the role that emotions play in our actions, behavior and thinking has been misapprehended and misinterpreted.

Damasio [4] demonstrated in neurological studies that people who lack the capacity of emotional response can take incorrect decisions and execute mistaken actions that can limit their performance in society. On the other hand, based on the fact that emotions are an important part of the human intelligence, Goleman [9] coined the concept of EI.

In this work, we contribute with our research to propose an alternative solution for the problem of behavior control of an avatar, using a representation based on a computer model combining the use of EI. We are inspired by the presented ideas by Goleman [9] and works published by Mayer, Salovey and Caruso about EI [13, 14].

We apply the principles of EI (*self-consciousness, mood and emotion management, self-motivation, empathy and management*) to provide the avatars the ability to understand, distinguish and manage its emotions and those of other avatars. Thus, the AEI of an avatar includes: *self-consciousness, goal understanding, intentions, reactions and behaviors, and consciousness of other avatars and their feelings*.

Our main objective is not modeling the complexity of human being's behavior, but simulating autonomous emo-

tional agents with a personality and dynamic emotional behavior. That is to say, the decision making and action selection of the autonomous intelligent agents is regulated and controlled not only by external stimuli, but also by their personality, emotions and moods. Finally, we use the Emotional Competence Framework (ECF) to apply the following personal and social competencies of an avatar:

1. Personal Competence:

- (a) self-consciousness (awareness of emotions, accurate self-assessment and self-confidence),
- (b) self-regulation (self-control, self-consciousness and adaptability), and,
- (c) self-motivation (achievement tendency and commitment to the task).

2. Social Competence:

- (a) social awareness (empathy, service orientation, awareness of other avatars and political awareness), and,
- (b) social skills (communication, conflict management and cooperation with other entities).

This work is organized as follows. The next section presents a detailed revision of the state of the art about emotional agents, models of emotions and personality, and models of EI. The third section is dedicated to the proposal of this work. In this section we propose the use of EI in the animation of avatars, and we describe the functioning of the Amygdale in the behavioral animation of VC. In addition, we will give a brief description of a new ontology for Affective Personified Emotional (APE) Agents. In the last section we will present the conclusions obtained from this work.

2. Related Work

Through the time, several models have been proposed in a broad range of scientific areas to describe the functioning of the human mind. Emotions in special have received increasing attention and interest in several fields related to AI and Computer Sciences (CS), mainly in Human-Computer Interaction (HCI) and Human-Robot Interaction (HRI), where emotional receptivity (perceiving and interpreting of facial expressions) and emotional expressivity (expressing emotions) play an important role. In this section we will summarize the most important related topics and give our opinion about them.

2.1. Emotional Agents and the Role of Emotions

Several emotional architectures of agents have been proposed. Researchers have been particularly interested in de-

signing models that improve the believability of the agents in artificial situations, such as simulations in virtual environments. Some works incorporate emotions into rational agents, taking into account both primary emotions and secondary emotions for modeling decision making processes [10]. However, we believe that it is necessary to add human characteristics, such as personality and moods, in order to design more conceivable and believable agents.

Multi-Agent Systems (MAS) cover problems related to the autonomy, the cooperation and coordination between agents as well as the interaction of believable agents in virtual environments. In MAS, individual agents are assumed to be autonomous. That is to say, they should have the capability to deliberate and decide which actions to take or which tasks to perform in order to reach their goals.

In [18] is described an architecture based on Emotional-Belief-Desire-Intention (EBDI) agents, using emotional updating functions into four components: Emotion, Belief, Desire and Intention. This architecture demonstrated that EBDI agents have better performance than rational agents, because they have more flexibility and ability to be adaptable and survive in dynamic environments.

Based on the studies realized by researchers in different areas, such as psychology, neuroscience and philosophy, the main roles of the emotions in the agents are: *action selection, motivation, adaptation, social regulation, goal management, attention focus, strategic processing and self-model*.

2.2. Emotion and Personality Models Applied to Agents

The OCC model [17] is considered as a standard model for emotion synthesis and as the best model of categorization of emotions (22 emotion categories). The OCC model explains the human emotions and tries to predict under which situations, which emotions can be experimented. Emotions are divided into the following groups: reactions (positive or negative) to events, actions and objects. Thus, the consequences of an event can please or displease the agent (*pleased/displeased*), the agent can accept or reject actions (*approve/disapprove*), and the characteristics of an object can attract or not the attention of agent (*like/dislike*).

Nevertheless, The OCC model was developed to understand the emotions instead of simulating them. In 2003, Ortony [16] proposed a simplified version of OCC model using six positive categories (*joy, hope, relief, pride, gratitude and love*) and five negative categories (*distress, fear, disappointment, anger and hate*) to describe the emotions of believable agents.

Roseman, Jose and Spindel [19] proposed a different model of emotions based on events. They classified the events in two groups: *motive-consistent events* (events

consistent with the goals) and *motive-inconsistent events* (events that threaten the goals). This model determines the occurrence of an event based on the certainty that an event would really occur. However, this model does not provide a complete understanding and explanation of the emotional processes, since it does not describe a method for categorizing perceived events. For instance, some events can be perceived simultaneously as motive-consistent and motive-inconsistent, which will generate conflicting emotions or contradictory emotions.

Another approach was presented in Frijda's theory of emotion [8]. The central idea of this theory is the term *concern*. A concern represents the predisposition of a system to prefer certain states of the environment. Thus, the intensity of the emotions is determined essentially by the influences generated by relevant concerns.

Searching for a better model of emotion, the model FLAME (Fuzzy Logic Adaptive Model of Emotions) was created to produce emotions and simulate the emotional intelligence process [7]. This model uses fuzzy rules to explore the capability of fuzzy logic for modeling emotional processes, and capturing the fuzzy and complex nature of emotions. Taking advantage over other conventional models, this model creates a better simulation and representation of reality. Nonetheless, it is very important to consider the personality for determining the consistency of the emotional reactions over time.

The OCEAN model or Five Factor Model is a purely descriptive model of personality [15], it groups personality traits of human being in five dimensions: *Openness*, *Conscientiousness*, *Extraversion*, *Agreeableness* and *Neuroticism*. Although this model is widely accepted, it has many criticisms, because it does not indicate how exactly the personality is affected by obtained stimuli and experienced situations.

Masuch, Hartman and Schuster [12] presented a different model represented by seven personality dimensions: *Suspicion*, *Curiosity*, *Sociability*, *Aggression*, *Helpfulness*, *Vividness* and *Conscientiousness*. These seven personality dimensions are closely related to specific aspects of more general dimensions of the OCEAN model.

Due to the direct correspondence between emotions and facial expressions, many researchers prefer to employ the Ekman's six basic emotions (*joy*, *fear*, *sadness*, *dislike*, *anger* and *surprise*) for facial expression classification [6] and the OCEAN model, or the OCC model in combination with the OCEAN model.

2.3. Competing Models of Emotional Intelligence

There are two main kinds of models of EI: *mental ability models* and *mixed models*. Mental ability models make

predictions about the internal structure of the intelligence and also its implications in the daily life. That is to say, a mental ability model focuses on emotions and their interactions. Mixed models are different, because these concern mental abilities and characteristics, such as motivation, states of consciousness and social activity. Thus, the mental ability models operate in a region defined by emotion and cognition, whereas mixed models label a multitude of components as EI.

In 1995, Goleman [9] coined the concept of EI and defined it as the set of abilities that includes *self-control*, *zeal*, *persistence* and *self-motivation*. Goleman proposed a mixed model with the following areas of skills: *Knowledge about emotions*, *management of emotions*, *self-motivation*, *recognizing emotions in others* and *handling relationships*.

Mayer and Salovey [14] presented a mental ability model that includes the following areas of skills: *perception and expression of emotion*, *assimilating emotion in thought*, *understanding and analyzing emotion*, and *reflective regulation of emotion*. They defined EI as *the ability to perceive and express emotion, assimilate emotion in thought, understand and reason with emotion, and regulate emotion in the self and others*.

Bar-On [3] defined EI as *an array of noncognitive capabilities, competences, and skills that influence ones ability to succeed in coping with environmental demands and pressures*, and presented another mixed model, whose major areas of skills are: *intrapersonal skills*, *interpersonal skills*, *adaptability scales*, *stress-management scales* and *general mood*.

We conclude that mental ability models can be described as standard models of EI. Thus, the EI consists of four specific tasks: *emotional perception*, *assimilation*, *understanding*, and *management*.

3. Animation of Virtual Creatures Using EI

Nowadays an important trend in the development of dynamic virtual environments is to integrate proper characteristics of human being, such as personality, moods and emotions into avatars. In this work the avatars are controlled by software agents using a model of emotion based on EI, with the aim of providing a behavior model to make them more believable and conceivable. This model enable the avatars to distinguish and manage emotions in the same way that human beings do. Thus, the emotions play crucial role in the cognitive processes of an avatar, and are a very important element to model mental activities such as: *perception*, *learning*, *decision process*, *memory*, *behavior* and other cognitive functions.

We define the AEI of an avatar as the set of cognitive and non-cognitive abilities, capabilities, intrapersonal and interpersonal skills of avatar. That is to say, the AEI relates

mental processes involving emotional information. Mental processes include mainly: appraising and expressing emotions and regulating emotion in the self and others, and using emotions in adaptive ways. Therefore, based on the previous assumptions, we present a new emotional model, which aims to explain the behavior of an avatar based on the following three sets: *Cognition* (learning, dynamic planning and thought, and information processing), *Motivation* (basic instincts and impulses, and goal-oriented activities), and *Emotional Valence* (personality, moods and emotions).

Cognitive processes of avatar defined in this work are: understanding, drawing conclusions, decision making, dynamic planning and learning, among others. Thence, the avatar has the capability to abstract and reason. This capability includes the concepts of beliefs, knowledge, desires, preferences and intentions of avatar. Therefore, an avatar can act and interact in its virtual environment according to its emotions, mood and personality as an autonomous intelligent entity.

3.1. Function of the Amygdale in the Behavioral Animation of VC

According to existing theories, in the human brain there is a small structure called Amygdale [1]. This cerebral structure performs functions biologically associated with emotions. It is supposed that the Amygdale works in the following way: the emotional brain scans everything happening to us from moment to moment. Its aim is to know if something that happened in the past and influenced our emotions (for instance, made us sad or angry) is similar to what is happening now. If so, the Amygdale calls alarm and declares emergency, so in a split second we are ready to act [2]. The Amygdale is not an emotional centre as it was traditionally considered, but it acts as regulating filter, catching stimuli to create a warning state. The Amygdale, also called emotional brain, can mobilize in less time than the thinking brain, which first needs to figure out what is going on.

We simulate the Amygdale into the avatars to control the behavior of them. Thus, we regulate the emotional and mood states of the avatars. Our method consists in showing the avatars some actions, which probably would exert the best possible influence over their emotions. This emotional influence will provide the values which are necessary to cover the best possible regulation of the internal emotional state of the avatars.

The module where the above mentioned method is included is the Amygdale Module (see figure 1). The Amygdale Module simulates biological reactions. Within the agents the mentioned module performs the same function as the real Amygdale performs in human beings. This module is used to choose the adequate action to be taken at the

precise moment and is required to solve the problem in the best possible way, that is, to regulate the internal emotional status optimally.

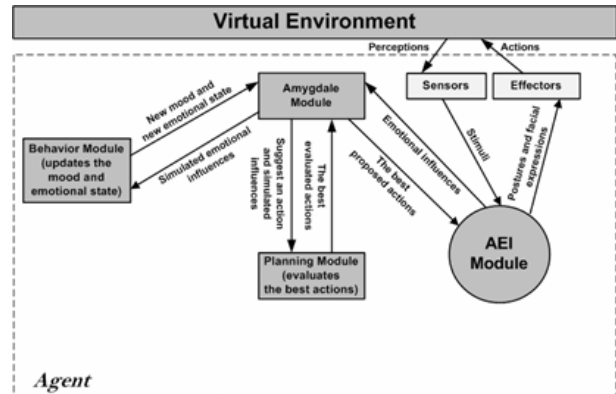


Figure 1. AEI Module and internal functionality of the Amygdale Module and Behavior Module.

3.2. An Ontology for APE Agents

In this work we propose an ontology for APE Agents in order to define the internal structure of an avatar (skeleton), its behavior (personality, emotions and moods) and its skills. Figure 2 shows the relationships between the main classes of the proposed ontology. These classes provide the semantic definition of an avatar. An avatar is defined using a morphology description (qualitative description) that defines its skeleton and an anthropometry description (quantitative description) that offers information about its age, gender, weight and height. In addition, as a part of its behavior, an avatar has a personality, emotions and moods. Thus, an avatar can develop its AEI to understand and to express its emotions and moods.

The internal skeleton of an avatar is formed by several parts, bones and joints in specific (skeleton parts). Each joint has a name and can have joints parents and/or joints children. There are motion constraints defined for each joint and a set of simple motions that define an alphabet of movements (micro-animation) that will be used to generate complex motions by means of combination between them (macro-animation).

The personality of an avatar is defined with the help of personality traits. Personality traits can determine the behavior of an avatar under the influence of certain events and stimuli. The emotional state is a set of emotions with an emotional history. A set of facial expressions and a set of postures associated with them correspond to every emotion of avatar. The facial expressions are produced with the help

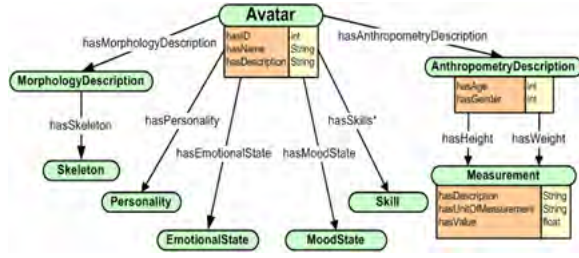


Figure 2. Semantic definition of an avatar.

of facial markers (similar to joints). Body markers provide each posture with its own set of animation sequences (see figure 3).

Emotions can be positive or negative, active or passive, but they always are of short duration. If an avatar is sad, it is able to adopt a facial expression and posture that show its emotional state. In the same way, the mood state of an avatar is a set of moods with a mood history. Moods can also be positive or negative and active or passive, but also neutral. Finally, each skill that an avatar learns is defined using a set of animation sequences. These sequences imply skeleton parts of avatar in motion and a set of corresponding facial expressions associated with them.

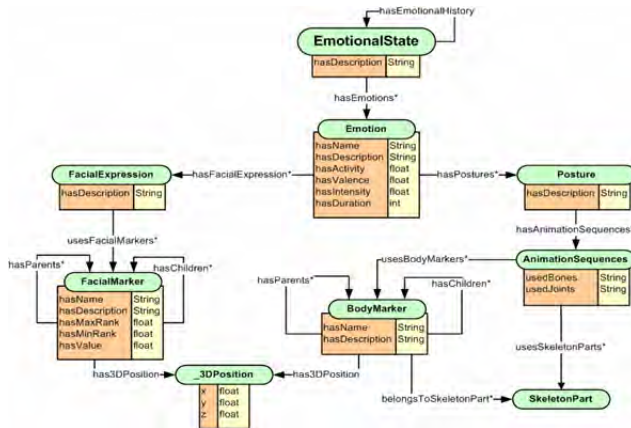


Figure 3. Emotional state of an avatar.

4. Cases Studies

The following cases studies were implemented over the GeDA-3D Agent Architecture [20]. We used the PEM model [5] for the first case study and the PEL model [11] for the last case study in combination with the ECF for updating the emotional and mood states of the avatars.

4.1. Persecution

In this implementation we used two kinds of APE Agents, one has an offensive behavior and the other a defensive behavior. The defensive agent tries to avoid the offensive agent's persecution, evading it when it is near and fleeing when it is closer. Three different emotions have been defined for these agents: anger, joy and fear. Avatars modify their facial expressions and postures, showing the predominant emotion in their emotional state. They change their facial expressions, when some interaction modifies their emotional and mood states in the virtual environment.



Figure 4. An offensive avatar that persecutes a defensive avatar.

4.2. Pin8 Family Interaction

Figure 5 shows different situations where the emotional skills can be used and manipulated for animating avatars (we have called them pin8 family). For example, if a dog attacks and persecutes a virtual human, the victim can run really scared away.

5. Conclusions

Though there are many research works, a consensual definition or dominant theory to explain the nature of emotions still does not exist, this partly due to the fact that emotion is a complex phenomenon. In order to provide the avatars with EI it is necessary to give them the capacity to understand their emotions. In this work we have presented a methodology to determine how the agents must express their emotions.

We have explained how to provide the avatars with knowledge necessary to express emotions. Firstly, we proposed an ontology to define emotions of avatars and their associated facial expressions. Later, we analyzed how avatars

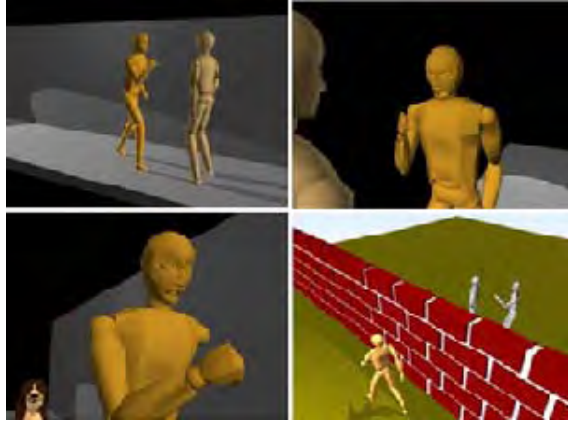


Figure 5. Pin8 family and their interactions in a virtual environment.

must express their emotions according to their emotional states. Finally, we presented a model of AEI in order to formalize the representation of the emotions of the avatars.

The AEI helps the avatars to take most accurate decisions. These decisions keep the agents' emotional status balanced and keep them alive in their virtual environment. Sensorial entrances and influences the avatars receive from their environment alter their behavior and motion and also affect their emotions and moods. The AEI is a topic that is slightly explored. However, in this work we presented a first approach that allows us to gain more knowledge about AEI and develop it in the context of emotional agents.

All the proposed models in this work were implemented over the GeDA-3D Agent Architecture. The GeDA-3D Agent Architecture owns the features defined in MAS theory, but it offers some extra features specific to the problem. This architecture contributes for example, to the topics of goal-specification, skill based behaviors, collective knowledge bases, posture descriptors and facial animation descriptors. The main idea of our work is the application of AEI to the behavior of avatars. Thus, the AEI makes the avatars act in a most credible manner.

References

- [1] R. Adolphs, D. Tranel, and A. R. Damasio. The human amygdale in social judgement. *Nature*, 393(6684):470–474, 1998.
- [2] A. K. Anderson and E. A. Phelps. Is the human amygdale critical for the subjective experience of emotion? evidence of intact dispositional affect in patients with amygdale lesions. *Journal of Cognitive Neuroscience*, 14(5):709–720, 2002.
- [3] R. Bar-On. *The Emotional Quotient Inventory (EQ-i): Technical Manual*. Multi-Health Systems, Toronto, 1997.
- [4] A. R. Damasio. *Descartes' Error: Emotion, Reason and the Human Brain*. Avon Books, New York, 1994.
- [5] A. Egges, S. Kshirsagar, and N. Magnenat-Thalmann. Generic personality and emotion simulation for conversational agents. *Computer Animation and Virtual Worlds*, 15(1):1–13, 2004.
- [6] P. Ekman. *Moods, Emotions, and Traits, the Nature of Emotion: Fundamental Questions*. Oxford University Press, New York, 1994.
- [7] M. S. El-Nasr, J. Yen, and T. R. Ioerger. Flame: Fuzzy logic adaptive model of emotions. *Autonomous Agents and Multi-Agent Systems*, 3(3):219–257, 2000.
- [8] N. Fridja. *The Emotions: Studies in Emotion and Social Interaction*. Cambridge Universty Press, New York, 1987.
- [9] D. Goleman. *Emotional Intelligence*. Bantam Books, New York, 1995.
- [10] H. Jiang and J. M. Vidal. From rational to emotional agents. 2006.
- [11] T. Li, Y. Qiu, P. Yue, and G. Zhong. Exploiting model of personality and emotion of learning companion agent. *International Conference on Computer Systems and Applications*, pages 860–865, 2007.
- [12] M. Masuch, K. Hartman, and G. Schuster. Emotional agents for interactive environments. pages 96–102, 2006.
- [13] J. Mayer. *Emotion, Intelligence, Emotional Intelligence*. Lawrence Erlbaum and Associates, Mahwah, New Jersey, 2000.
- [14] J. D. Mayer, P. Salovey, and D. R. Caruso. *Models of emotional intelligence*. Cambridge University Press, Cambridge, UK, 2000.
- [15] R. McCrae and O. John. An introduction to the five-factor model and its application. *Journal of Personality*, 60(2):175–215, 1992.
- [16] A. Ortony. On making believable emotional agents believable. *Emotions in Humans and Artifacts*, pages 189–212, 2003.
- [17] A. Ortony, G. L. Clore, and A. Collins. *The Cognitive Structure of Emotions*. Cambridge University Press, New York, 1988.
- [18] D. Pereira, E. Oliveira, N. Moreira, and L. Sarmento. Towards an architecture for emotional bdi agents. In *Proceedings of the 12th Portuguese Conference on Artificial Intelligence*, pages 40–47, 2005.
- [19] I. J. Roseman, P. E. Jose, and M. S. Spindel. Appraisals of emotion-eliciting events: Testing a theory of discrete emotions. *Journal of Personality and Social Psychology*, 59(2):899–915, 1990.
- [20] F. Zúñiga, F. F. Ramos, and I. Piza. Geda-3d agent architecture. In *Proceedings of the 11th International Conference on Parallel and Distributed Systems*, pages 201–205, Fukuoka, Japan, 2005.

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