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Preface

Computational Intelligence (CI) is usually defined as advanced computing methods and techniques that exhibit an ability to adapt to the current scenario and are based on the paradigms of artificial neural networks, fuzzy systems, and evolutionary algorithms, augmented with knowledge elements. CI systems are often designed to mimic one or more aspects of carbon-based biological intelligence. However, the observed explosive growth in the volume, velocity, and variety of the data created on a daily basis, defines new challenges and new possible trends in evolution CI models and methodologies.

On one hand, the issues so-called the "Big Data" and "Data Intensive" (DI) problems, require the continuous increase of the processing speeds of the data servers and whole network infrastructure and becomes difficult for the analysis and interpretation with on-hand data management applications. The Big Data era poses a critically difficult challenge and striking development opportunities to DI and High-Performance Computing: how to efficiently turn massively large data into valuable information and meaningful knowledge. CI techniques seem to be the effective means for supporting data intensive computing, management and by trading-off various preferences and goals of the system users, resource and service managers, system administrators and resource owners.

However, on the other hand, today's large-scale intelligent networks – such as future generation grids, peer-to-peer and ad-hoc networks, and clouds-enable the aggregation and sharing of geographically-distributed resources from different organization with distinct owners, administrators, and policies. With the advent of such systems, where efficient inter-domain operation is one of the most important features, it is arguably required to investigate novel methods and techniques to enable secure access to data and resources, efficient scheduling, self-adaptation, decentralization, and self-organization.

The concept of the support the data intensive systems and large-scale intelligent networks by computational intelligence methodologies and models brings together results from all those three areas, making a positive impact on the development of new efficient data and information systems. This issue encompasses twelve research papers reporting the recent results on models, solutions, and techniques from such a wide research area, ranging from conceptual and theoretical developments to advanced technologies and innovative applications and tools.

In the first five papers the authors present interesting examples of innovative applications of the well known theoretical models. Vitabile *et al.* used the 17-node Bayesian Networks for supporting the decision process in integrated coastal zone management. In this practical approach the authors analyzed coastal area in the Trapani Province (Sicily, Italy) and the Bayesian Network has been trained on the real data sets. The achieved results makes the presented technology useful for the local authorities in Italy, but also in the other regions with the similar infrastructure and environment, to support their decisions for improving the economic and social activities and revenues of target area. The critical issue in Artificial Neural Networks (ANNs) – one of the fundamental CI technologies – is the optimization learning process in order to improve the performance of the whole network used for data partitioning and classification, data acquisition or supporting the monitoring systems in the new era intelligent networks. Krok in the second paper presents an interesting idea of the improvement of such a ANN learning by Kalman filters, very well known methods used in pattern recognition. Along with the improvement of the training method, those filters allow to optimize the whole network architecture to adapt it to a particular scenario. The paper contains valuable theoretical analysis, which makes it very useful as a good complete background material for the further developments in the partitioning of the Big Data streams. In the next paper, Abaev *et al.* present a model of traffic optimization and control of Session Initiation Protocol (SIP) proxy servers, which is one of the major problems in large-scale networks. It is important not only from the research point of view, but also with several examples of successful practical applications. They collected data for network traffic circulating between two geographical regions, provided a simple statistical analysis and then modeled the traffic in SIP by using the Markov Modulated Poisson Process (MMPP) theory. The problem of the overloading of the servers remains challenging in today's networking, especially when it is related to the Big Data processing and management. This problem is addressed in the fourth paper in the context of the Voice over Internet Protocol systems (VoIPs). The authors present the prediction model for cloud-based VoIP systems for the incoming traffic (in charge of anticipating the number of the calls in the system) which can be used as an input for the dynamic load balancing in the cloud and multi-cloud environments. They combined in their model maximum likelihood and the weighted likelihood based prediction techniques. The provide experiments show the significant improvement of the prediction accuracy of the proposed methods compare with the ANN technologies. Finally, Iacono and Marrone in the fifth paper present a tool – Telemaco – for the optimization of graph layout. It can be effectively used for solving many problems modeled by using the graph theory, but also for the optimization of the structure of multi-population genetic algorithm or the architecture of the Wireless Sensor Networks or resource allocation schema in clusters, grids and clouds.

The problem of the effective resource allocation in large-scale heterogeneous systems remains still a challenging task especially in the light of the distributed datacenters and data storage nodes in the network and mobile devices, which can be defined as additional nodes, not just the clients for the data and service remote access. Smelcerz presents in her paper a short characteristics and comparative analysis of the main features of conventional and mobile cloud systems. This comparison shows from the other perspective the high complexity of the resource allocation and scheduling problems, which should be solved in the mobile environment, especially when energy consumption criterion is considered as the main objection. Burceanu *et al.* define a framework for automatic context-aware data provisioning in the mobile cloud systems, which – based on their result for Smart City project – seems to be a good candidate for supporting Big Data processing in mobile distributed dynamic environments, where the outsourcing of the mobile devices is a crucial issue.

The last part of the issue includes five papers with interesting practical applications of the intelligent methodologies in engineering, media and data analysis. First, Plichta *et al.*, Suciu *et al.* and Grzonka present various practical engineering approaches in speech recognition and mechanics. Lopez-Fuentes developed a model based on the resource reputation analysis and indexation for the improvement of the communication and collaboration in peer-to-peer networks. This problem is crucial in the management of big mixed media (audio and video) streams. In the last paper, Felkner presents the background of the set-theoretic and operational semantics of the Role-based Trust (RT) management language with negation, which is applied for the improvement of the users' authorization procedures and data protection in the large-scale networks.

I believe that all papers presented in this issue ought to serve as a reference for students, researchers, and industry practitioners interested or currently working in the evolving and interdisciplinary areas of data intensive computing, modeling and processing, and intelligent networking. I hope that the readers will find there new inspirations for their further research.

I am grateful to all the contributors of this issue. I thank the authors for their time and efforts in the presentation of their recent research results. I also would like to express my sincere thanks to the reviewers, who have helped us to ensure the quality of this publication. My special thanks go to the journal Editors, Managers and Publishers for their great support throughout the entire publication process.

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Paper

Assessing Coastal Sustainability: A Bayesian Approach for Modeling and Estimating a Global Index for Measuring Risk

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Abstract—Integrated Coastal Zone Management is an emerging research area. The aim is to provide a global view of different and heterogeneous aspects interacting in a geographical area. Decision Support Systems, integrating Computational Intelligence methods, can be successfully used to estimate useful anthropic and environmental indexes. Bayesian Networks have been widely used in the environmental science domain. In this paper a Bayesian model for estimating the Sustainable Coastal Index is presented. The designed Bayesian Network consists of 17 nodes, hierarchically organized in 4 layers. The first layer is initialized with the season and the physiographic region information. In the second layer, the first-order indexes, depending on raw data, of physiographic regions are computed. The third layer estimates the second-order indexes of the analyzed physiographic regions. In the fourth layer, the global Sustainable Coastal Index is inferred. Processed data refers to 13 physiographic regions in the Province of Trapani, western Sicily, Italy. Gathered data describes the environmental information, the agricultural, fisheries, and economical behaviors of the local population and land. The Bayesian Network was trained and tested using a real dataset acquired between 2000 and 2006. The developed system presents interesting results.

Keywords—Bayesian Networks, Decision Support Systems, Integrated Coastal Zone Management, Sustainable Coastal Index.

1. Introduction

The west coast of Sicily, Italy is an interesting local productive zone. This area is one of the main trade centers in the Mediterranean Sea, and it is a growing touristic area.

Nevertheless, the zone suffers of some problems, due its inhomogeneities. The local authorities are looking for a point of balance between various factors characterizing the area's economy, such as tourism, fishing, infrastructures, number of building, and the health of the environment, and the quality of life.

Data analysis techniques can help for modeling and reducing the above cited negative aspects through territory data processing. This makes possible to develop management tools for modeling territory's activities in order to combine and optimize socio-economic and environmental aspects.

These models have to be very sensitive to small changes of heterogeneous factors. Slight changes of environmental data such as temperature, water salinity, dissolved oxygen, the speed and direction of wind, but also the number of tourists or the percentage of buildings in the area can significantly change the sustainability of the entire region.

This work presents a Decision Support System (DSS), based on Computational Intelligent methods, for the coastal sustainability evaluation and forecasting. With more details, a Bayesian Network (BN) has been used to estimate a global index that can be used by local authorities to implement the appropriate decisions, improving the territorial economic and social activities.

The developed Bayesian Network consists of 17 nodes, hierarchically organized in 4 layers. The first layer is initialized with the season and the physiographic region information. In the second layer, the first-order indexes, depending on raw data, of physiographic regions are computed. The third layer estimates the second-order-indexes of the analyzed physiographic regions. In the fourth layer, the global Sustainable Coastal Index (SCI) is inferred.

The proposed model has been used for modeling and simulating the sustainable coastal index in the Province of Trapani area, western Sicily, Italy. The analyzed coastal area is 378.983 km^2 and both naturalistic and socio-economic points of views are very interesting. The Bayesian Network was trained and tested using a real dataset. Data series has been collected between 1 January 2002 and 31 July 2006 for a total of 1673 days and the measurements were daily gathered. The dataset is composed of heterogeneous data, since it contains anthropogenic factors, economic data, tourism data, infrastructure data and fishing data.

The developed model has been integrated as plug-in in a commercial Geographic Information System (GIS). The model can be used as DSS to estimate and evaluate the impact of anthropic, infrastructure, environmental changes over the coastal sustainability. The proposed system shows

interesting performances when employed as DSS for coastal zone management.
The remainder of the paper is organized as follows. Section 2 presents some relevant works about DSS and automatic models use. In Section 3, the proposed model is described. Section 4 shows some experimental results. Section 5 presents the integration of the proposed model in a commercial GIS. Finally, Section 6 contains some concluding remarks.

2. Related Works

Nowadays, automatic models and Decision Support Systems are used for analyzing and understating the future direction of interesting application fields, such as environmental sciences, agricultural sciences, transportation systems, economic sciences, and tourism [1]–[9]. On the other hand, Computational Intelligence is an enabling discipline for developing these approaches [10]–[12].
In [1], the authors present a work to assist local authorities and policy makers of interesting regions in order to implement a mobility scheme for managing the demand in a sustainable way.
The paper [2] describes a work developed as part of a DSS. Its aim is to support an operator in his/her tasks for analyzing soft data to monitor and anticipate a geopolitical crisis.
The work presented in [3] discusses an innovative approach for measuring tourism competitiveness using eight main indicators over 200 countries. Weights for each theme are derived using confirmatory factor analysis in order to compute an aggregate index. Cluster analysis is used to group destinations according to their performance level.
In [4], the authors develop an indexing model to evaluate sustainability performance of urban settings in order to assess environmental impacts of urban development and provide an indexing model to planning agencies. The model is a DSS to be used in curbing negative impacts of urban development.
The work presented in [5] examines factors influencing the sustainability of Integrated Coastal Management (ICM) projects in Philippines and Indonesia. Measures of project sustainability were developed. Primary data collected at the village level was analyzed to determine the effects of project activities and individual characteristics on ICM sustainability.
The authors of paper [6] underline the analysis and establishment of indicator-driven programs to assess change in coastal and watershed systems. The work reviews the need for and provide an assessment of important frameworks designed to foster the integration. It argues that the evolution of the Driver Pressure State Impact Response (DPSIR) framework provides an essential contribution.
In [7], the authors assert that the modeling of key environmental and socio-economic processes are a vital tools. They required to buttress coastal management institutions and practice. The proposed framework was based on a conceptual model that lays stress on functional value diversity and the links between ecosystem processes, functions and outputs of goods and services. In the work a three overlapping procedural stages for coastal resource assessment process are explained.
Article [8] discusses the potential contribution of indicators to assess the performance of the governance processes involved in integrated coastal management. They focused on the evaluation phase and on the need to complement process-oriented indicators with outcome-oriented indicators to improve adaptive management and accountability. The example of integrated management of marine protected areas is used to propose a menu of indicators of global applicability.
In [9], the authors present a sustainability indicator system based on three sub-systems: environment and resources, economic development, and society. It has been set up to evaluate the nature of development in the coastal zone in the administrative regions of the Municipality of Shanghai and Chong Ming Island.
In this work, a Decision Support System for local authorities is presented. The aim is to develop an useful model for Integrated Coastal Zone Management in order to take the appropriate decisions for improving the territorial economic and social activities and revenues. The DSS has been developed for a global index, namely Sustainable Coastal Index, evaluation and estimation.

3. A Four-Layer Bayesian Model

The most interesting feature of BNs, compared to decision trees or neural networks, is most certainly the possibility of taking into account prior information about a given problem, in terms of structural relationships among its features. This prior expertise, or domain knowledge, about the structure of a Bayesian Network has been used in this work.
A Bayesian Network is a particular Directed Acyclic Graph (DAG) for representing causes and effects scenarios [13]. BN variables are represented by nodes. The values assumed by nodes can be a state, or a set of probability values. Nodes are connected with edges, denoting causality relationships.

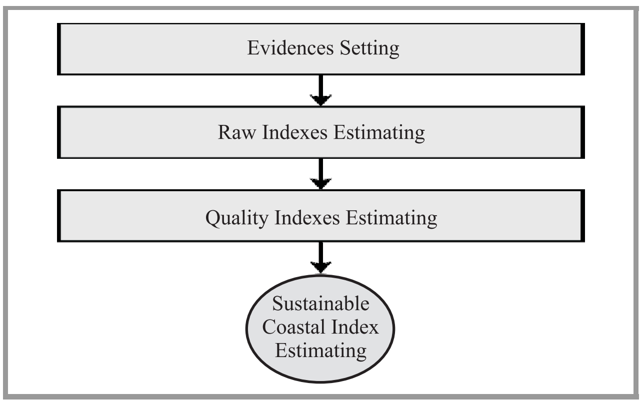

Fig. 1. The block diagram of the proposed model.

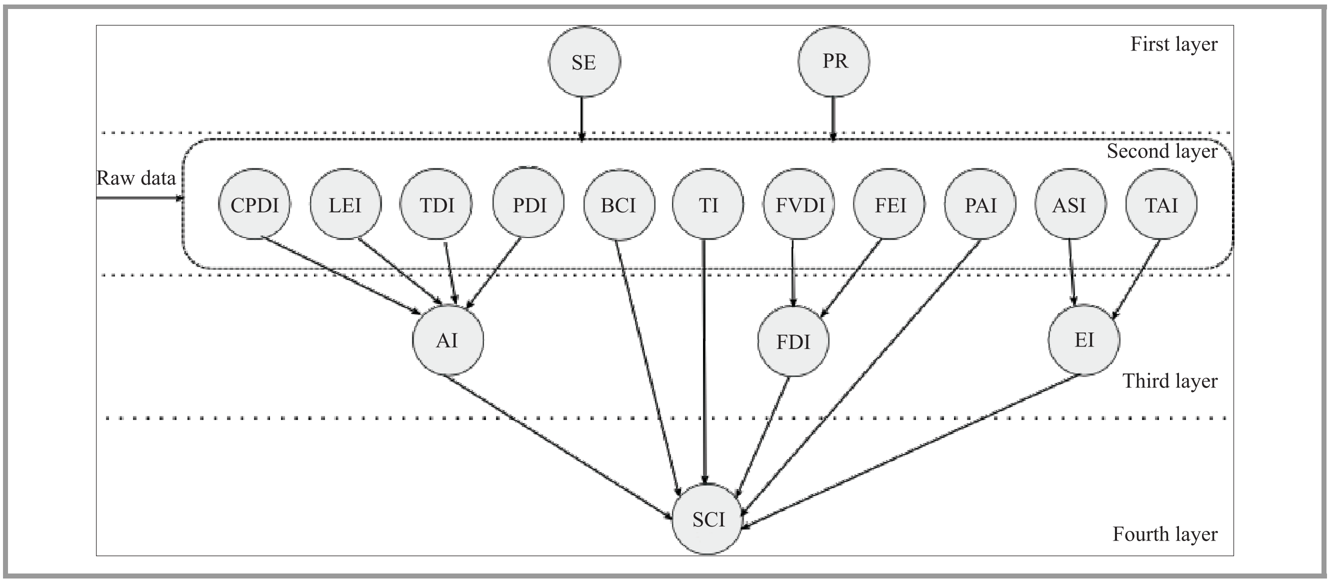

Fig. 2. The Bayesian Network structure is composed of 17 nodes in 4 layers. The first layer is initialized with the season and the PR information. In the second layer, the first-order indexes, depending on raw data, of PRs are computed. The third layer estimates the second-order indexes for the analyzed PRs. In the fourth layer, the global SCI is inferred.

The network models the probabilistic relationships between the variables of a system, accounting their historical information. They are used to model scenarios where some information are partially unavailable or with uncertainty. Like the uncertain human reasoning, they give a representation of knowledge producing probabilistic results, given a certain degree of truth on a topic.

A feature (node) is conditionally independent from its non-descendants given its parents: $X1$ is conditionally independent from $X2$ given $X3$ if $P(X1|X2,X3) = P(X1|X3)$ for all possible values of $X1$, $X2$, $X3$ [13].

A Conditional Probability Table (CPT) can be associated to each node. It contains the probabilities of the values of the node conditioned by the values of its parent nodes. For each parent and its possible states, a new entry in the CPT is created.

The variable probability in a state, considering the actual scenario, is known as belief. In particular, a-priori beliefs are a kind of belief calculated only on a prior information. They are defined considering the CPTs, whereas the evidence is the information of a current scenario.

A different approach, which is the one considered in this paper, is to directly include in the model the relationship between raw data, intermediate indexes, and the overall index. The four-layer Bayesian model, developed with the collaboration of domain experts, allows for aggregating and processing data with different abstraction levels.

With more details, the present work describes a Bayesian Network for estimating the Sustainable Coastal Index (SCI) of 13 physiographic regions (PRs).

Figure 1 shows the block diagram of the proposed model. The designed BN consists of 17 nodes, hierarchically organized in 4 layers. The first layer is initialized with the season and the physiographic region information. In the second layer, the first-order indexes, depending on raw data, of the 13 physiographic regions are computed. The third layer estimates the second-order indexes for the analyzed physiographic regions. In the fourth layer, the global SCI is inferred.

Figure 2 shows the structure of the BN used in this work. Each Bayesian Node expresses a quality index for a productive aspect of the processed physiographic unit. It is computed considering three factors: the physiographic region, the season and the related raw data.

Table 1
Grouping class for quality index values

Class	Range
Low	0 – 3.333
Medium	3.334 – 6.333
High	6.334 – 10

In the network training phase, the required probability distributions are derived from the domain experts through the use of a mathematical model. The mathematical model produces continuous values in the [0, 10] range, where 10 is the optimal sustainability value and 0 is the most critical value. A grouping operation has been performed for mapping the range in 3 main classes, as shown in Table 1. The above classes have been used for computing the CPT associated to each node.

3.1. First Layer – Evidence Setting

The first layer has two nodes for setting the season and the PR in order to start the network computation. The former node has 4 possible states, while the latter node has 13 possible states, related to the main physiographic regions in the Province of Trapani, west Sicily, Italy.

3.2. Second Layer – First Order Index Estimations

In the second layer, the probability of first-order indexes, depending on raw data, of physiographic regions is computed. Considering the evidence of the BN, the appropriate table of conditioned probability has been loaded. The second layer is composed of 11 nodes, described below by the following mathematical model [14]:

Coastal Population Density Index (CPDI): The CPDI deals with the population (tourists and residents) living in the coast over the surface of the coast. The CPDI is 0 if the ratio is less than or equal to 1, while it is equal to 10 if the ratio is greater than or equal to 1000. These two reference values have been interpolated with a logarithmic base 2 function:

$$CPDI = log_2\left(\frac{Kmq - surface - coast}{Total - population}\right). \tag{1}$$

Land Exploitation Index (LEI): The LEI deals with the built area over the whole surface of the physiographic region. The value is calculated as follows:

$$LEI = 10\cdot\left(1 - \frac{Surface - built}{Kmq - surface - PR}\right). \tag{2}$$

Fishing Exploitation Index (FEI): The FEI deals with the quantity of caught fish considering the surface of the physiographic region. The values are normalized between 0 and 10, computed from:

$$FEI = \frac{Kg - fish - caught}{Kmq - surface - PR}. \tag{3}$$

Fishing Vessel Density Index (FVDI): The FVDI deals with the number of fishing vessels over the surface of the physiographic region. The value is calculated as follows:

$$FVDI = \frac{Vessels}{Kmq - surface - PR}. \tag{4}$$

Agricultural Sustainability Index (ASI): The ASI deals with cultivated land over the whole surface of the physiographic region. The useful cultivated land deals with the farming types. In particular, 5 crops have been considered: olive groves, orchards, vineyards, citrus groves, and woods. Each crop has a different weight (*c-weight*) in the ASI computation.

$$ASI = \sum_{cultivation=1}^{5} \frac{c - weight\cdot Kmq - surface - cult.}{Kmq - surface - PR}. \tag{5}$$

Population Density Index (PDI): The PDI deals with the population density, considering the surface of the physiographic region:

$$PDI = \frac{Population}{Kmq - surface - PR}. \tag{6}$$

Tourist Density Index (TDI): The TDI deals with the number of tourists in a particular season and the whole population of the PR:

$$TDI = \left(1 - \frac{Tourists}{Population}\cdot\frac{1}{days - for - season}\right)\cdot 10. \tag{7}$$

Tourist Accommodation Index (TAI): The TAI deals with the arithmetic average of two indicators: Kiosks Sustainability Index (KSI) and Hotel Sustainability Index (HSI). KSI is 0 if the distance between two kiosks is equal to or less than 200 meters, while it is 10 if the distance exceeds 10,000 meters. KSI will be again 10 if no kiosks are in the area. HSI is 0 if the geographic area has 200 beds per square kilometer, while HSI is 10 if no beds are in the geographic area:

$$KSI = \frac{\log_4\left(\frac{Km-coast}{(n-Kiosks)} - 199\right) + \frac{Km-coast}{(685\cdot n-Kiosks)} - 0.25}{2}, \tag{8}$$

$$HSI = -\frac{Beds}{20\cdot Kmq} + 10. \tag{9}$$

Balneal Coast Index (BCI): The BCI deals with the balneal coasts, and it is calculated as follows:

$$BCI = \frac{LengthBalnealCoast}{LengthTotalCoast}\cdot 10. \tag{10}$$

Transportation Index (TI): The TI quantifies the transport infrastructures in the geographic area. It is computed considering the road surface, the stations and airports in the geographic area.

Protected Area Index (PAI): The PAI is calculated considering either the surface of protected land areas and the surface of sea protected areas in the geographic area.

In the second layer, the BN computes a probability value for each state of the node. The probability values are used as inputs for the third layer of the BN.

3.3. Third Layer – Second Order Index Estimation

The third layer estimates the second-order indexes. It is composed of 3 nodes: Anthropic Index, Economic Index, and Fishing Density Index. The possible states of each node are: Low, Medium, and High. The means of each node is described below:

Table 2
Conditioned Probability Table of the SCI node

AI	BCI	FDI	PAI	EI	TI	Low [%]	Medium [%]	High [%]
Low	Low	Medium	Low	High	Low	39.202	22.619	38.179
					Medium	72.692	24.360	2.948
					High	23.910	23.408	2.948
			Medium	Low	Low	6.496	25.341	68.163
					Medium	65.101	18.379	16.520
					High	35.209	22.431	42.360
				Medium	Low	16.883	27.720	55.398
					Medium	9.319	55.531	35.150
					High	34.921	44.847	20.232

Anthropic Index (AI): The index is produced by the following second-layer-indexes: PDI, TDI, CPDI, and LEI.

Economic Index (EI): The index is produced by the following second-layer-indexes: ASI and TAI.

Fishing Density Index (FDI): The index is produced by the following second-layer-indexes: FEI and FVDI.

3.4. Fourth Layer – Sustainable Coastal Index Estimation

In the fourth layer, the global Sustainable Coastal Index is inferred. The BN computes the probability for the three possible states of SCI: Low, Medium, and High. If the predominant class is the High class, the related physiographic region has an high sustainability features, considering the socio-economic and environmental factors. In the same way, the Medium class and the Low class can be defined. SCI can be used by local authorities in order to take the appropriate decisions for improving the territorial economic and social activities.

4. Experimental Results

The Bayesian Network was trained and tested using a real dataset. The dataset refers to the Province of Trapani, western Sicily, Italy. The series was gathered between 1 January 2002 and 31 July 2006 [14].

The BN has been developed using the Norsys Netica v3.19 software [15]. The development framework allows for creating and simulating advanced bayesian belief network. Netica uses the *Message Passing* algorithm [16]–[19] for extracting probabilistic inference in a BN.

In the first phase, Netica compiles the BN into a *junction tree*. The *junction tree* is a set of Netica internal connections, and they are not visible to users. In the second phase, the Netica user inserts the evidences in the network and selects the output nodes computing the global beliefs.

The 1637-days-dataset has been split in two subsets for the training phase and the testing phase, respectively. Data series gathered from 2002 to 2004 has been used for training phase, while data series gathered from 2005 to 2006 has been used for the testing phase.

In the learning phase the Conditioned Probability Tables are carried out. As example, the CPT of the SCI node is shown in Table 2. After the training phase, the learned network has been tested using the appropriate dataset.

The proposed model performances have been evaluated comparing the network output against the findings of a group of domain experts. With more details, 120 simulation trials, with a flat distribution between the classes, have been carried out changing network inputs, i.e., time and/or physiographic unit. The confusion matrix describing the network accuracy is shown in Table 3.

Table 3
The Confusion Matrix describing the model accuracy

	Low	Medium	High
Low	98%	2%	–
Medium	1%	97%	2%
High	–	2%	98%

In what follows, two examples, showing system functionalities, are shown. In first example, the evidence of the nodes SE, PR, CPDI, DPI, TII, BI, DBI, FI, and PAI has been superimposed. In particular, SE is *summer* and PR is *Marsala*. Netica computes the indexes for the remaining node, and, at the end, the SCI node class is inferred. Figure 3 shows the Netica network structure: the gray nodes are superimposed, while the yellow nodes are inferred. In this case, the SCI predominant probability is the High class (P(High) = 63.3%, P(Medium) = 20.8%, and

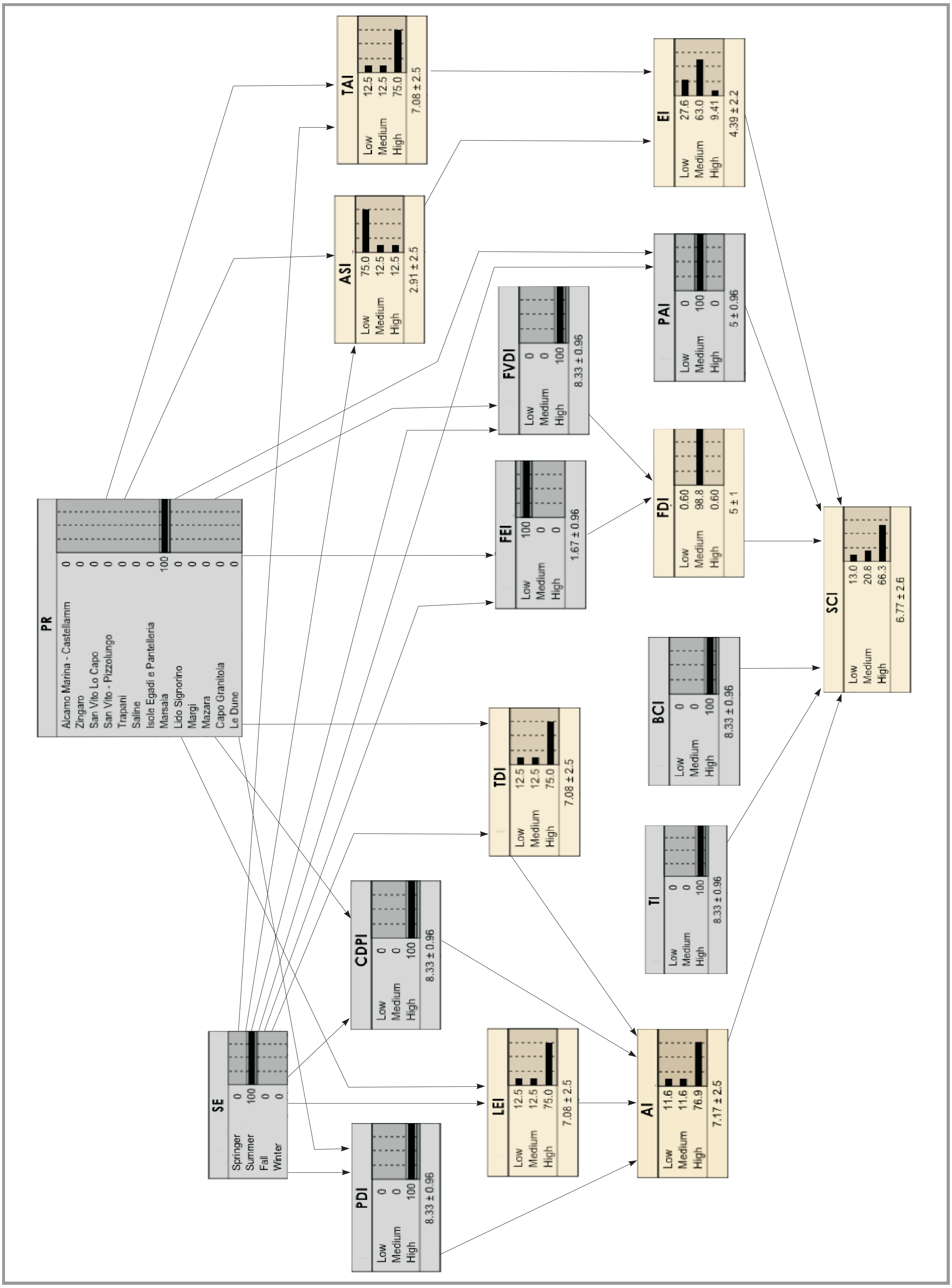

Fig. 3. The BN diagram extracted from the Netica software. The network is related to Marsala PR and the the summer season. The inferred SCI node shows the 63.3% of probability associated to the High value.

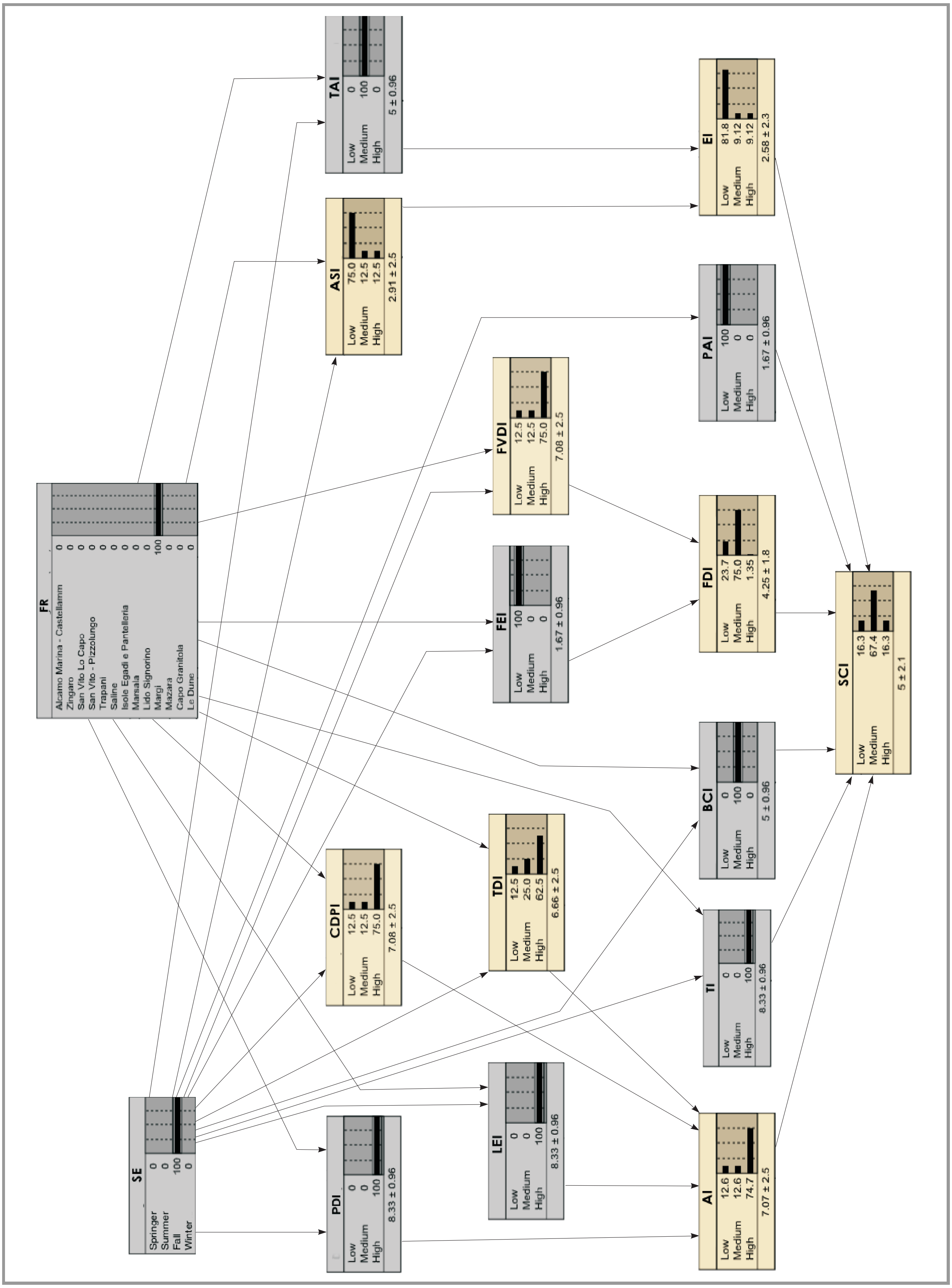

Fig. 4. The BN diagram extracted from Netica software. The network is related to Margi PR and the fall season. The inferred SCI node shows the 67.4% of probability associated to the Medium value.

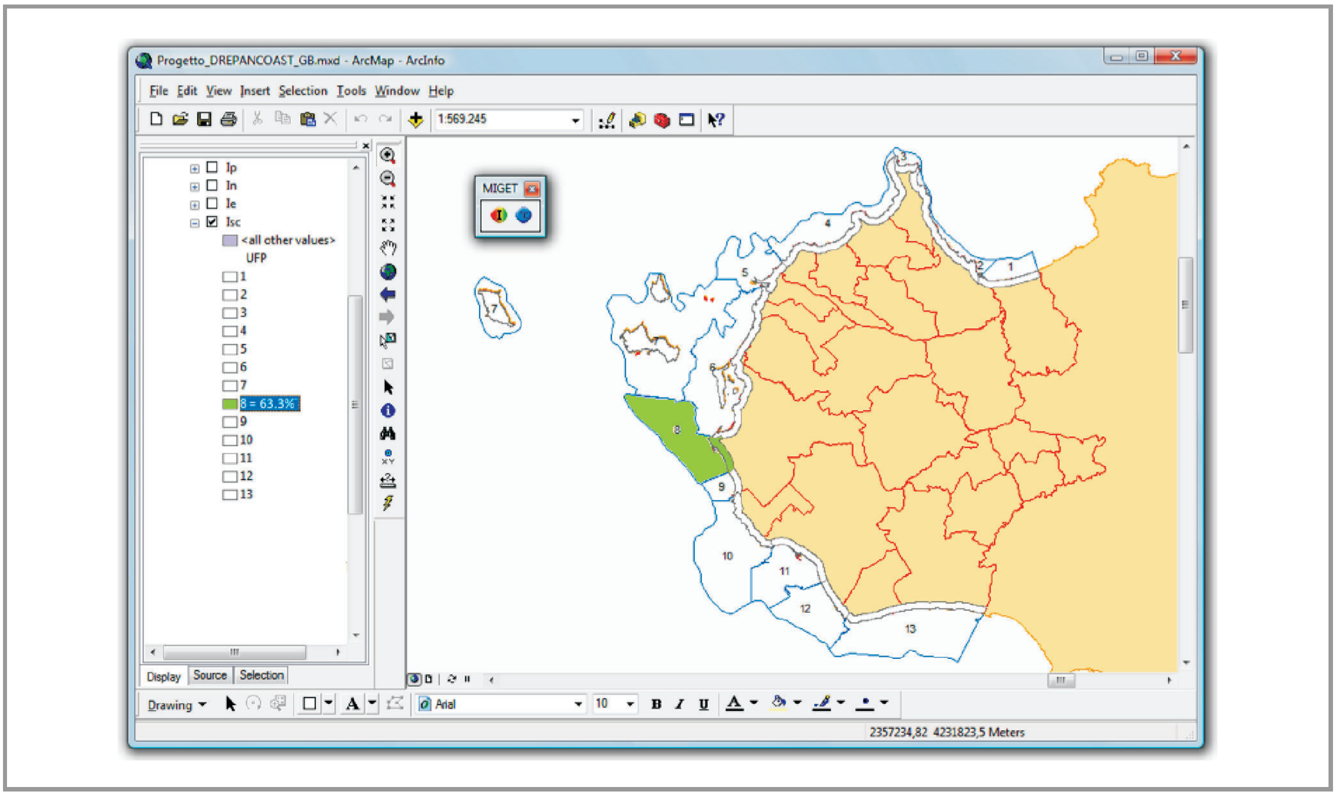

Fig. 5. The Bayesian model running in ERSI ArcGIS 9. The SCI has been computed for PR8 (Marsala).

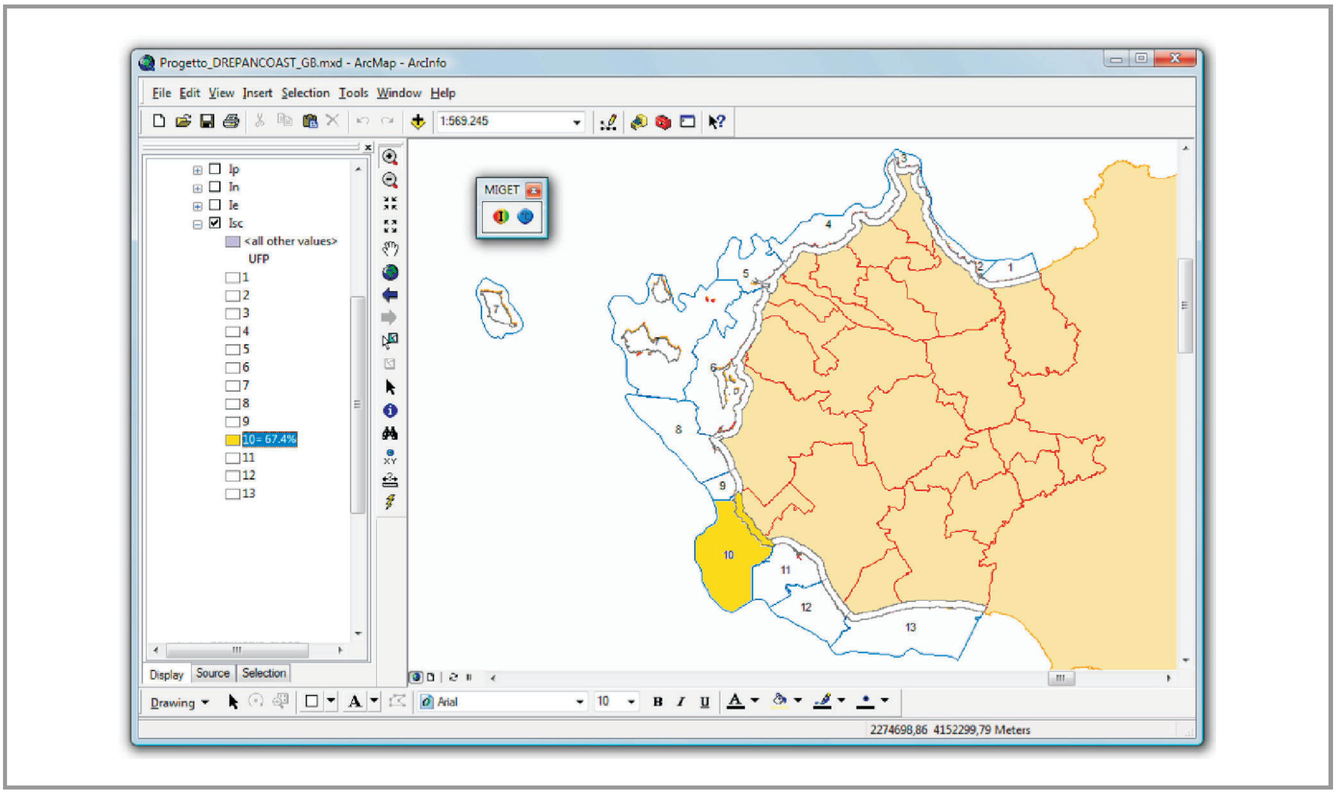

Fig. 6. The Bayesian model running in ERSI ArcGIS 9. The SCI has been computed for PR10 (Margi).

P(Low) = 13.0%). In the second example, the evidence of nodes SE, PR, DPI, ELI, TII, BI, FI, PAI, and ATI has been superimposed. In particular, SE is *fall* and PR is *Margi*. Figure 4 shows the Netica network structure. The gray nodes are superimposed, while the yellow nodes are inferred. In this case, the SCI predominant probability is the Medium class (P(High) = 16.4%, P(Medium) = 67.4%, P(Low) = 16.3%). Similar results can be obtained selecting different physiographic units and/or seasons.

5. Integrating GIS and Bayesian Network

The presented Bayesian model has been also integrated in the ESRI ArcGIS 9 Desktop [20]. ArchGis is a set of integrated software tools for the Geographic Information System management. It is based on a shared library of GIS components, namely ArcObjects.

ESRI provides a set of development tools, namely ArcGIS Developer Kit. This framework is useful for creating standalone applications, based on ESRI ArcGIS Engine, as well as for extending the capabilities of existing software, such as ArcMap. The programming environments used for developing the present work are the .Net SDK, Visual Basic, C++ and Java.

Netica v3.19 environment allows for exporting the Bayesian network using the appropriate APIs. The APIs allow for exporting the code of the developed BN in several programming languages, such as Java, C, C++, and C#.

Due to the previous features, an ad-hoc plug-in was developed for the SCI estimating.

The proposed ArcGIS plug-in consists of a toolbar linked to the Netica library. This toolbar is loaded into ArcMap software and integrated in the ArcGIS user interface, becoming part of the original software tool.

Two screenshots of the running system are shown in Figures 5 and 6. The plug-in associates the red color to the Low class, the yellow color to the Medium class and the green color to the High class.

Figure 5 shows the computed SCI for PR 8 (Marsala). The left bar shows the green SCI probability value (63.3%) of the highlighted physiographic region (in right side). Figure 6 shows the computed SCI for PR 10 (Margi). The left bar shows the yellow SCI probability value (67.4%) of the highlighted PR.

6. Conclusions

Integrated Coastal Zone Management (ICZM) is an emerging research area. Computational Intelligence methods and Geographic Information can be successful used to estimate and forecast the impact of environment and anthropic actions and strategies.

In this work a Decision Support System, based on a Bayesian Network, for Sustainable Coastal Index evaluation and forecasting has been presented. Local authorities can use the developed model in order to take the appropriate decisions for improving the economic and social activities and revenues of target area.

The Bayesian Network was trained and tested using a real dataset. The dataset used has been acquired between 2000 and 2006 in the Province of Trapani, western Sicily, Italy. Model results are very interesting and they have been validated by a group of domain experts.

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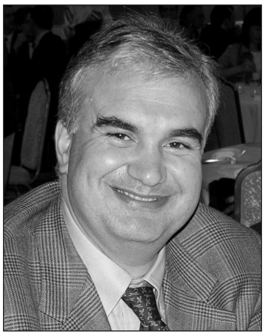

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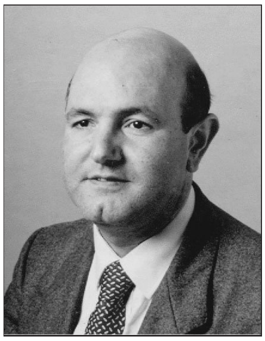

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Paper

The Development of Kalman Filter Learning Technique for Artificial Neural Networks

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Abstract—The paper presents an idea of using the Kalman Filtering (KF) for learning the Artificial Neural Networks (ANN). It is shown that KF can be fully competitive or more beneficial method with comparison standard Artificial Neural Networks learning techniques. The development of the method is presented respecting selective learning of chosen part of ANN. Another issue presented in this paper is the author's concept of automatic selection of architecture of ANN learned by means of KF based on removing unnecessary connection inside the network. The effectiveness of presented ideas is illustrated on the examples of time series modeling and prediction. Considered data came from the experiments and situ measurements in the field of structural mechanics and materials.

Keywords—Artificial Neural Networks, Kalman Filter.

1. Introduction

The growing popularity and increasing use of the Artificial Neural Networks in virtually all fields of engineering and technical sciences lead to their fast development. From among the learning algorithms gradient descent methods, conjugate gradient methods, the Levenberg-Marquardt algorithm (LM), and the resilient back propagation algorithm (Rprop) were developed predominantly. Their advantages and disadvantages are thoroughly understood [1]. In the paper the alternative learning algorithm was exploited. The nonlinear Kalman Filter can be considered as a simple dynamic Bayesian networks [2]. It calculates a minimum mean-square error estimator for the underlying process and it is adopted into estimating weights and biases of ANN. After development the KF learning method enabled to control the learning process strongly that resulted in very efficient simulations and predictions made by ANN and may be alternative for traditional ANN learning methods [2]–[5].

2. Multilayered Feed Forward Artificial Neural Networks Learned by Kalman Filtering

2.1. Artificial Neural Networks

Standard Multilayered Feed Forward ANN with the same activation function in the each node was considered. Assuming that N, M, K, is the number of inputs, outputs and hidden neurons respectively the answer of the network for the input vector

$$\mathbf{x} = [x_1, \ldots, x_N] \tag{1}$$

for the m-th ($m = 1, \ldots, M$) output is

$$y_m = F\left(\sum_{i=0}^{K} x_i^u w_{m,i}^2\right), \tag{2}$$

where

$$x_i^u = F\left(\sum_{j=0}^{N} x_j w_{i,j}^1\right). \tag{3}$$

Finally

$$y_m = F\left(\sum_{i=0}^{K} F(\sum_{j=0}^{N} x_j w_{i,j}^1) w_{m,i}^2\right), \tag{4}$$

where: F is the activation functions for first and second layer and $w_{i,m}^2$ is the connection between i-th ($i = 1, 2, \ldots, K$) hidden neuron with m-th output neuron ($m = 1, 2, \ldots, M$). Similarly for $w_{i,j}^1$, where $j = 1, 2, \ldots, N$. Vector $[x_1^u, \ldots, x_K^u]$ is created by signals coming from K hidden neurons. Lets assume that $w_{m,0}^1$ and $w_{i,0}^2$ are ANN biases. For real ϕ, the activation functions in the form

$$F(\phi) = \frac{1}{1 + e^{-a\phi}} \tag{5}$$

or

$$F(\phi) = tanh(\phi/2) = \frac{1 - e^{-a\phi}}{1 + e^{-a\phi}} \tag{6}$$

were considered.

The ANN described above is learned by means of KF using the teacher - standard procedure was used: the pth input vector $\mathbf{x}^p$ should generate the desirable target $\mathbf{t}^p$ were p from 1 to P are known learning patterns. The magnitude between target and the ANN output Eq. (4) gives the level of ANN changes that are necessary to obtain its goal. The performance of the network and its generalization abilities were tested on separate testing set of patterns like in every standard method of ANN learning and testing [2]. In the paper the change of ANN weight vector is calculated using estimation inside KF model.

2.2. Algorithm Node Decoupled Extended Kalman Filter

The Kalman Filter is an algorithm that uses a series of measurements observed over time, containing noise and produces estimates of unknown variables. It operates recursively on streams of noisy input data to produce a statistically optimal estimate of the underlying system state. The basic Kalman Filter is linear [6]. In the Extended Kalman Filter (EKF), the state transition and observation models may be differentiable functions. This procedure can be adopted into the process of ANN learning, assuming that the main function in the model represent ANN itself [2]. Among a number possibilities for decoupling the single ANN into several Kalman filtering models, it was proposed to use new KF model for each neuron. It resulted in smaller dimensions of matrix being proceed and additional advantages in the field of controlling learning process.

Let I be the number of all neurons in ANN ($I = M + K$) and let $\mathbf{w}^i$ be the connection weight associated with the ith neuron $i = 1, 2, \ldots, I$. For each neuron the Kalman Filter model is built including process Eq. (7) and measurement Eq. (8):

$$\mathbf{w}^i_{k+1} = \mathbf{w}^i_k + \omega^i_k\,, \tag{7}$$

$$\mathbf{t}_k = \mathbf{h}(\mathbf{w}_k, \mathbf{x}_k) + \nu_k\,, \tag{8}$$

where: k is discrete pseudo-time parameter (for ANN marking the number of learning pattern), $\mathbf{w}$ is the state vector of KF model (here corresponding to the set of synaptic weights and biases), $\mathbf{h}$ is non-linear function that is representing ANN and takes the Formula (4), **x/t** is input/output vectors that corresponds to input of ANN and known target, ω^i_k, ν_k are Gaussian process and measurement noises with mean and covariance matrices defined by:

$$\mathbf{E}(\nu_k) = \mathbf{E}(\omega^i_k) = 0\,, \tag{9}$$

$$\mathbf{E}\big((\omega^i_k)(\omega^i_l)^T\big) = (\mathbf{Q}^i_k)\delta_{kl}\,, \tag{10}$$

$$\mathbf{E}(\nu_k \nu_l^T) = \mathbf{R}_k \delta_{kl}\,, \tag{11}$$

where: E is the expected value of a random variable, δ_{kl} is equal to 1 if $l = k$ or 0 if not.

The change of ANN weights in i-th node ($\mathbf{w}^i$) during the presentation of k-th learning pattern takes than the following form, see Fig. 1:

$$\mathbf{K}^i_k = \mathbf{P}_k \mathbf{H}^i_k \left[\sum_{j=1}^{g} (\mathbf{H}^j_k)^T \mathbf{P}^j_k \mathbf{H}^j_k + \mathbf{R}_k \right]^{-1}, \tag{12}$$

$$\mathbf{w}^i_{k+1} = \mathbf{w}^i_k + \mathbf{K}^i_k \xi_k\,, \tag{13}$$

$$\mathbf{P}^i_{k+1} = \left(\mathbf{I} - \mathbf{K}^i_k (\mathbf{H}^i_k)^T \right) \mathbf{P}^i_k + \mathbf{Q}^i_k\,, \tag{14}$$

where: $\mathbf{K}^i_k$ is Kalman gain matrix, $\mathbf{P}^i_k$ is approximate error covariance matrix, $\xi_k = \mathbf{t}_k - \mathbf{y}_k$ is error vector, with the target vector $\mathbf{t}_k$ for the k-th presentation of a training pattern, $\mathbf{y}_k$ is output vector given by Eq. (4).

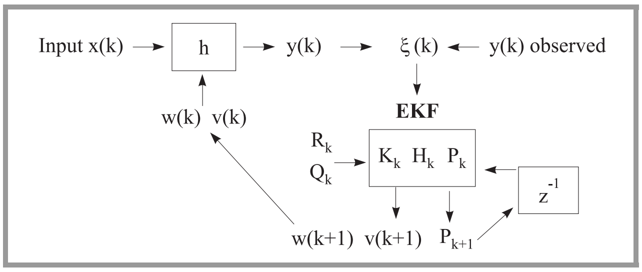

Fig. 1. Kalman Filtering learning algorithm.

$\mathbf{H}$ is the matrix of current linearization of Eq. (8):

$$\mathbf{H}^i_k = \frac{\partial \mathbf{h}}{\partial \mathbf{w}^i}\,. \tag{15}$$

The considered parameters for the Gaussian noise may be for example in the form:

$$\mathbf{Q}^i_k = a \cdot e^{\frac{s-1}{b}} \cdot I \tag{16}$$

$$\mathbf{R}_k = c \cdot e^{\frac{s-1}{d}} \cdot I\,, \tag{17}$$

where: a, b, c, d real, positive numbers, I is identity matrix which dimension depends on the dimension of the model that is the number of connection weights for the particular neuron g, s is the number of learning epoch [7].

In this approach the typical problem of multidimensional optimization of gradient descent type [8] is changed into prediction – correction problem. The prediction phase uses the state estimate from the previous time step to produce an estimate of the state at the current time step – Eq. (13). In the correction phase the quality of estimation is included, see Eq. (14).

3. Development of Kalman Filter Learning Technique

The choosing of the proper (according to considered numerical problem) architecture of ANN is very important. The ANN too small may be insufficient to model the data, too large ANN may be very time consuming and the over fitting may occur. It leads to the very poor results as far as testing set is considered [1]. The choosing of the proper architecture of ANN may be done automatically without involving a priori guess from the network constructor.

In standard ANN learning techniques the whole network is modified during learning process. The magnitude of networks weights correction is the issue that differs from each pattern to another. The KF model gives us also the equipment to measure the quality of estimation, made during learning for the each single weight. The areas (particular nodes or chosen parts of the ANN) can be selected were the quality of estimation is insufficient. Then the network (due to decoupling) may be than learned selectively. Only the areas that are under the average quality of estimation level are changed. In the paper this technique is presented as far

as the particular nodes are concerned, but the decoupling into single weights may be applied also.
The techniques presented in this paragraph were proposed by the author and their effectiveness was examined by a lot of tests presented at the end of the paper.

3.1. ANN Pruning

There are two main approaches to the automatically driven ANN architecture setting. In the paper the pruning method was chosen [1]. It assumes starting from large ANN and making it smaller during learning process by erasing the particular weights between neurons. After initial learning the weights are examined as far as their impact into the quality of learning is considered. Ineffective weights are erased. Then the smaller network is learned and the procedure is repeated until it is beneficial for the learning and testing process.
The initial test showed that methods dedicated for traditional ANN pruning cannot be applied for KF learning. Due to the fact that in KF model estimation is made – the statistically based methods of pruning were adopted [9], but the scheme had to be adjusted into KF technique.
Let

$$\mathbf{w} = [w^1, w^2, ..., w^I] \tag{18}$$

be the vector made of all ANN weights. The statistics Λ is calculated:

$$\Lambda(w^i) = \ln \frac{\left| \sum_{p=1}^{L} w^i - \eta \frac{\partial E(p)}{\partial w^i} \right|}{\eta \sqrt{\sum_{p=1}^{L} \left(\frac{\partial E(p)}{\partial w^i} - mean_{\{p=1,2,...,L\}} \frac{\partial E(p)}{\partial w^i} \right)^2}} \tag{19}$$

there the summation is made according to p from 1 to the L-th (last element of the learning set), η is a constant number (learning rate), $mean_{\{p=1,2,...,L\}}$ – is the arithmetical mean in the learning set, $E(p)$ is the learning error for input x^p, that is for p-th learning pattern:

$$E(p) = ||\mathbf{h}(\mathbf{t}_p) - \mathbf{y}_p||^2 \tag{20}$$

where t_p is the pth target whereas h represents the ANN structure, see Eq. (4), $||.||$ is the standard Euclidean norm.

The larger values of Λ statistics for the particular node means that these weights are significant for the learning process. There are the following parameters that were stated to be the most important during the process of pruning:

- the initial ANN is learned until S epoch is reached;
- from $S_{start} = S+1$ epoch network is changed;
- setting the parameters of white noise in the Eqs. (7)–(8) to enable smaller network learn as fast as at the beginning of the learning process; after cutting the parameters of white noise is shifted back to the value from $S_{reset} < S_{start}$ epoch;
- choosing of k the number of epochs between consecutive cuts;

  setting the Λ_{edge} limit value for the Λ; weights that are below Λ_{egde} are erased;
- after each cutting the testing error is examined. If it is occurred to be growing, the procedure is stopped and the network is shifted back to the last better architecture.

For the particular problem without the previous knowledge of ANN behavior the genetic algorithm may be responsible for the optimal parameter selection.

3.2. Selective Learning of ANN Based on Approximate Covariance Error Matrix

Using the KF learning method it is enable to build separate model for each node of the network. Due to the possible highly asymmetric values of the inputs of the ANN and different values of the initial weights – the values of the weights may differ much. Moreover in KF model their values are estimated with the known quality. By the examining of the diagonal values of the matrix (13), it is possible to select those weights for which the estimation quality is unsatisfactory.
The proposed algorithm is based on dynamic changes of the areas of ANN for the current learning.
Calculating matrix $\mathbf{P}^i_{k+1}$ that approximates

$$E(\mathbf{w}^i_{k+1} - \widehat{\mathbf{w}}^i_{k+1})(\mathbf{w}^i_{k+1} - \widehat{\mathbf{w}}^i_{k+1})^T, \tag{21}$$

where $\widehat{\mathbf{w}}^i_{k+1}$ is the estimator for $\mathbf{w}^i_{k+1}$ calculated after presenting k-th learning pattern, on the main diagonal of the matrix the level of estimation quality for the single weight can be obtained:

$$P^i_{k+1}(m) \approx E\left(w^i_{k+1}(m) - \widehat{w}^i_{k+1}(m)\right)\left(w^i_{k+1}(m) - \widehat{w}^i_{k+1}(m)\right)^T, \tag{22}$$

for m from 1 to the numer of weights in the considered neuron. These values represents the approximation of errors of estimation for the single m-th weight of the i-th neuron in the selected node.
After initial examination the quality of estimation represented by Eq. (22) was treated separately inside each layer. The following algorithm was adopted (presented here for first layer):

- initial learning for $s = 1, 2, \ldots, S_0$ epochs, and then inside layer,
- calculating the mean level of error of estimation inside the layer

  $$MEAN = mean_{m,i} P^i_{k+1}(m), \tag{23}$$

  for i taken from 1 to the range of first layer, m taken from 1 to the number of weights inside layer,

- calculating the mean level of error of estimation inside the i-th neuron

$$Mean(i) = mean_m P_{k+1}^i(m), \tag{24}$$

m taken from 1 to the number of weights inside layer

- selecting those neurons that were learned very good, by finding i such that

$$Mean(i) < \alpha MEAN, \tag{25}$$

where α is assumed percentage constant,

- for those neurons for S_{freeze} epochs the values remains constant,
- the rest of the neurons were learned for S_{learn} epochs,
- calculating $MEAN$, $Mean(i)$ and repeating the procedure until they differ much.

4. Numerical Testing of the Proposed Solutions

Many numerical tests proving the effectiveness of the basic KF method were conducted. Considered numerical problems came from experiments and situ measurements considering: mining tremors, cyclic loading of steel and concrete specimens, hysteresis loops for superconductor. The tests were designed to develop a methodology that could be helpful during modeling and predicting time series and other time dependent phenomenon. The comparison between KF solution and standard neural solution (Back Propagation learning technique, Rprop) was investigated. In the absence of neural solution the results were compared to the available classical methods of mechanics of materials. Considered numerical problems were stated to be very difficult to solve either by ANN or for classical methods [7].

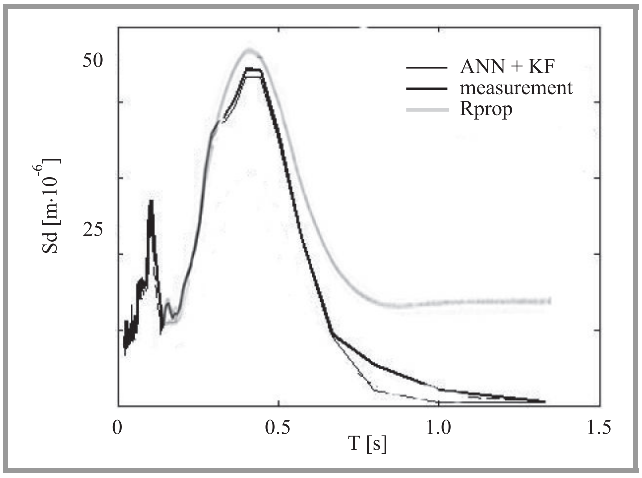

Fig. 2. Neural prediction of response spectra from mining tremors.

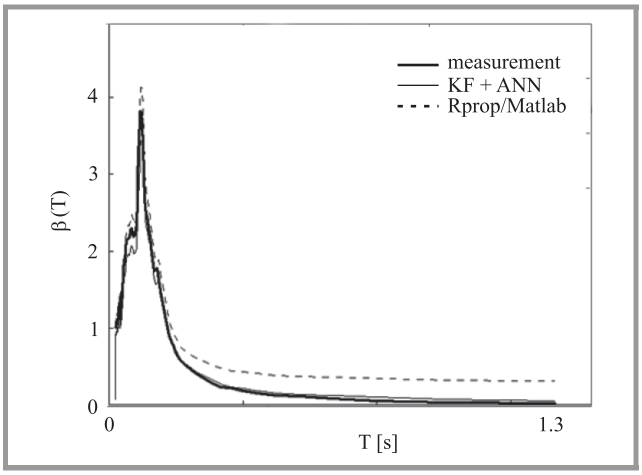

Fig. 3. Neural prediction of response spectra from mining tremors.

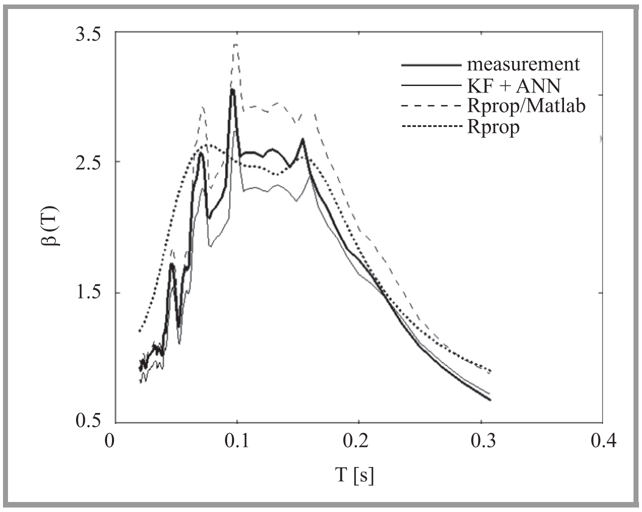

Fig. 4. Neural prediction of response spectra from mining tremors.

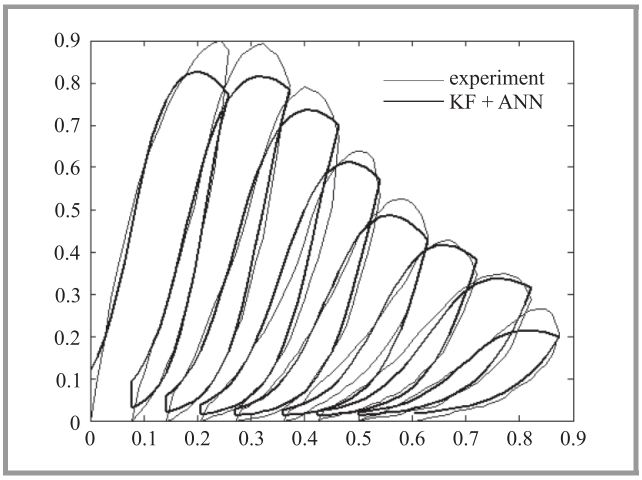

Fig. 5. The analysis of cyclic behavior of concrete specimens.

The chosen results are presented in graphical demonstrative form below, see Figs. 2–9 [10]–[12]. The chosen results for Selective learning of ANN based on Approximate

Covariance Error Matrix and pruning are presented below for the selected numerical problem, see Figs. 10–11 [7].

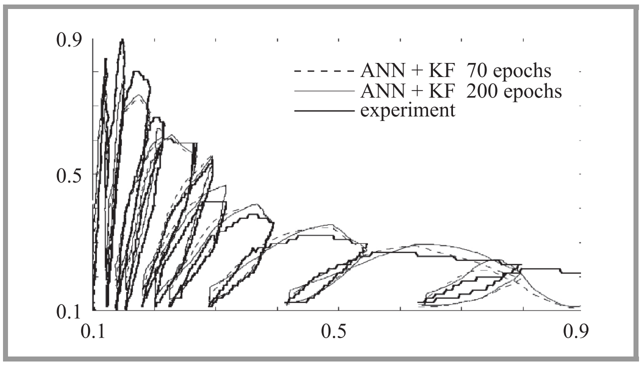

Fig. 6. The analysis of cyclic behavior of concrete specimens.

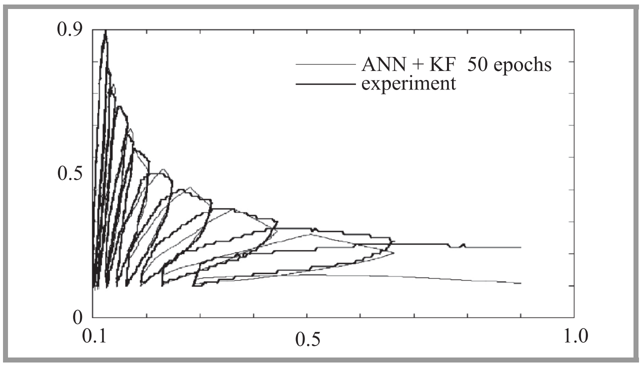

Fig. 7. The analysis of cyclic behavior of concrete specimens.

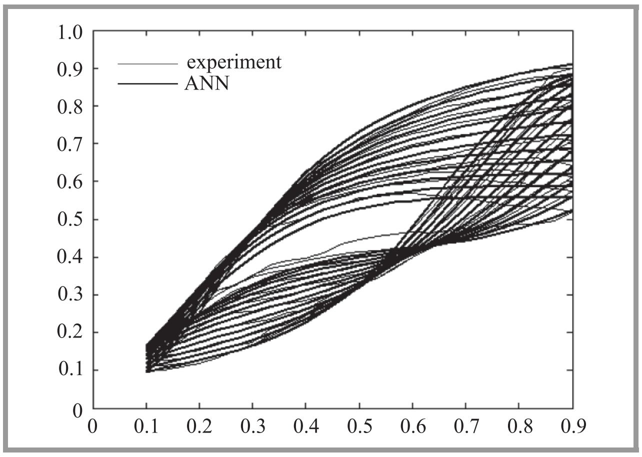

Fig. 8. Simulation and prediction of cyclic loading of steel specimens.

All the algorithms were implemented using Matlab. It is a high-level language and interactive environment for numerical computation, visualization, and programming. Is provides very efficient libraries for matrices operations. KF algorithm was vectored. Neural Network Toolbox including gradient descent methods, conjugate gradient methods, the Levenberg-Marquardt algorithm, and the resilient backpropagation algorithm was used as the reference for

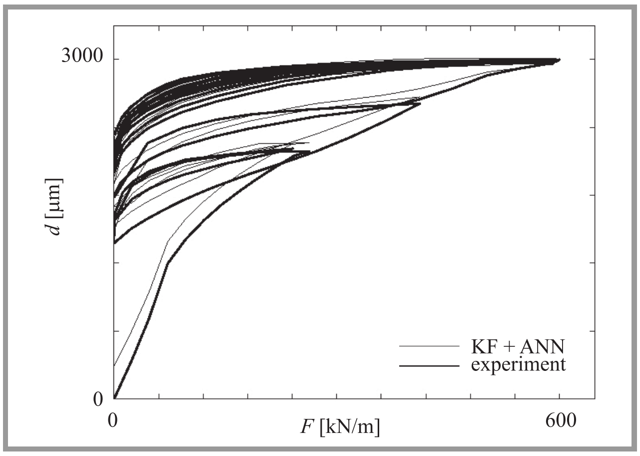

Fig. 9. Simulation of hysteresis loops for superconductor.

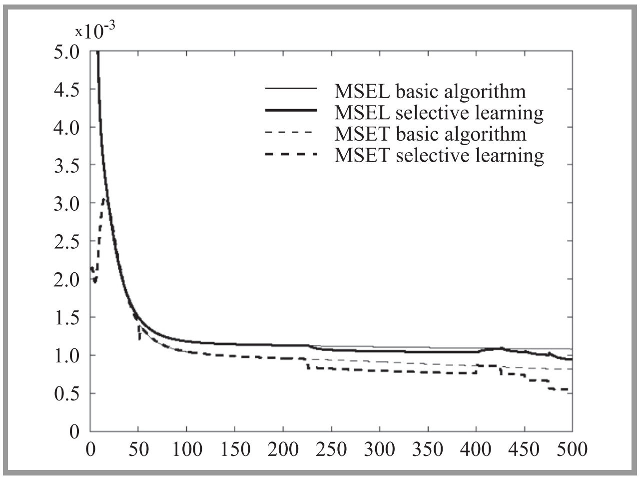

Fig. 10. The pruning of KF neural network for simulation of hysteresis loops for superconductor, MSEL=Mean Square Errors in the Learning Set, MSET=Mean Square Errors in the testing set.

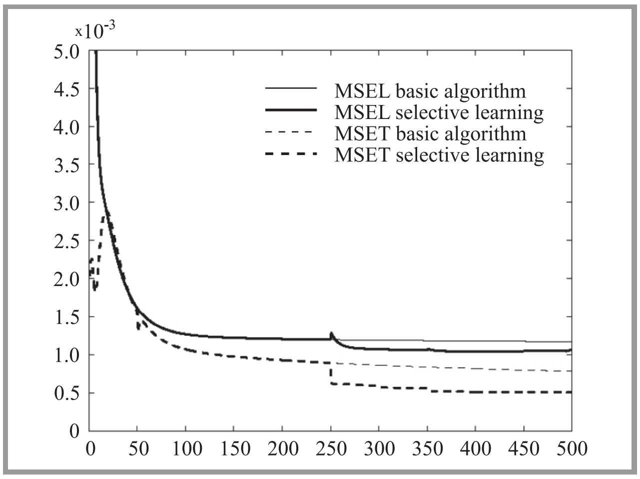

Fig. 11. The pruning of KF neural network for simulation of hysteresis loops for superconductor.

investigating the efficiency of KF learning method. The set of good practices were found for applying KF as a learning technique for ANN. It is presented in Table 1.

Table 1
Good practices for applying KF as a learning technique for ANN

Problem	KF+ANN good practice
Number of epochs	About 10^2
Input vector	Time window include
Input vector magnitude	[0.2, 0.8]
Number of layers	2
Activation function	Nonlinear (5) or (6)
Noise function for Eqs. (6)–(7)	(16)–(17)
Number of weights in ANN	K About 10
Starting ANN for pruning	Input-50-50-output
Parameter Λ_{edge} for pruning	0.75
Parameter α for Eq. (25)	1

5. Conclusions

Kalman filtering method was stated to be very beneficial during learning of ANN solving numerical problems of real data simulation and prediction. Data for time series with correlation are modeled very precisely. For those type of data the obtained results shown that this method of learning seemed to be matched to the nature of considered numerical problem. For this reason it is beneficial to apply KF instead of traditional ANN learning methods like resilient propagation or Levenberg-Marquardt algorithm.

The effective method for automatic pruning of ANN learned by KF was proposed. It enable to shorten the time of calculations by learning the network containing smaller number of parameters.

The efficient method for expanding the basic KF learning method was developed. The analysis of approximate error covariance matrix values shown that this is the successful way to control the quality of learning process. It may result in avoiding the retarding of the learning process manifested by negligible changes of error of the ANN during learning process.

Both proposed methods are using the specific properties of KF learning method and enable to obtain better numerical solution for the considered problem of time series modeling and predicting.

Acknowledgements

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Paper

Statistical Analysis and Modeling of SIP Traffic for Parameter Estimation of Server Hysteretic Overload Control

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Abstract—The problem of overload control in Session Initiation Protocol (SIP) signaling networks gives rise to many questions which attract researchers from theoretical and practical point of view. Any mechanism that is claimed to settle this problem down demands estimation of local (control) parameters on which its performance is greatly dependent. In hysteretic mechanism these parameters are those which define hysteretic loops. In order to find appropriate values for parameters one needs adequate model of SIP traffic flow circulating in the network under consideration. In this paper the attempt is made to address this issue. Analysis of SIP traffic collected from telecommunication operator's network is presented. Traffic profile is built. It is shown that fitting with Markov Modulated Poisson Process with more than 2 phases is accurate. Estimated values of its parameters are given.

Keywords—Markov Modulated Poisson Process, overload control, SIP server, statistical analysis.

1. Introduction

The problem of overload control in SIP signaling networks gives rise to many questions which attract researchers from theoretical and practical point of view. Any mechanism that is claimed to settle this problem down demands estimation of local (control) parameters on which its performance is greatly dependent. This research is motivated by the idea that hysteretic control which is successfully deployed and used in SS7 networks may also be applicable and beneficial for overload control in next generation networks. Note that in hysteretic mechanism parameters that are subject to estimation are those which define hysteric loops. It is known that overload conditions that arise occasionally in signaling networks lead to severe loss of service quality which eventually affects network operators and/or service providers. There are many research papers that deal with analysis of systems with different overload control mechanisms, including hysteretic policy – [1]–[9] just to mention a few. In this paper system under consideration is SIP proxy server. The are two main approaches to analyze behavior of control mechanism in SIP networks: mathematical modeling and simulation. One of the drawbacks in mathematical modeling is the assumption concerning input flow and its parameters. Clearly if one chooses a simple model (say Poisson) then there is a risk that the nature of the real flow that is fed into the SIP proxy server is neglected too much and consequently the values of control parameters that are obtained may be inadequate. If one chooses more complex model, which reflects nature of real flow, it may lead to hardly tractable (with the growth of initial parameters values, i.e., system capacity, thresholds etc.) mathematical model. In this paper effort is made to address the issue of choosing input flow model and its parameters for SIP traffic, captured on SIP proxy server operating in telecommunication operator's network. We collect traffic circulating between two geographical regions, analyze it with well-known statistical methods and then try to fit Markov Modulated Poisson Process (MMPP) in SIP traffic data using different algorithms proposed in the literature [10]–[13]. In [14] there was build mathematical model of SIP server with MMPP input flow and two-level hysteretic policy and proposed optimization problem for choosing policy parameters. Using estimated in this study values of MMPP based on real SIP data one can calculate values of policy parameters for model in [14] and use them in practical implementations.

The paper is organized as follows. In the Section 2 description of traffic collection procedure and some insight into the traffic nature is given. Then SIP-I traffic model is being built in Section 3. Section 4 is devoted to statistical analysis of SIP-I traffic. In Section 5 the authors present the results of MMPP fitting. Conclusion contains short overview of obtained results and gives a glimpse of further research.

2. Traffic Collection and Data Description

Let's consider the fragment of the transit network of telecommunication operator depicted in Fig. 1. In this paper SIP-I traffic circulating between two different regions is analyzed [15]. Traffic aggregation happens on regional access

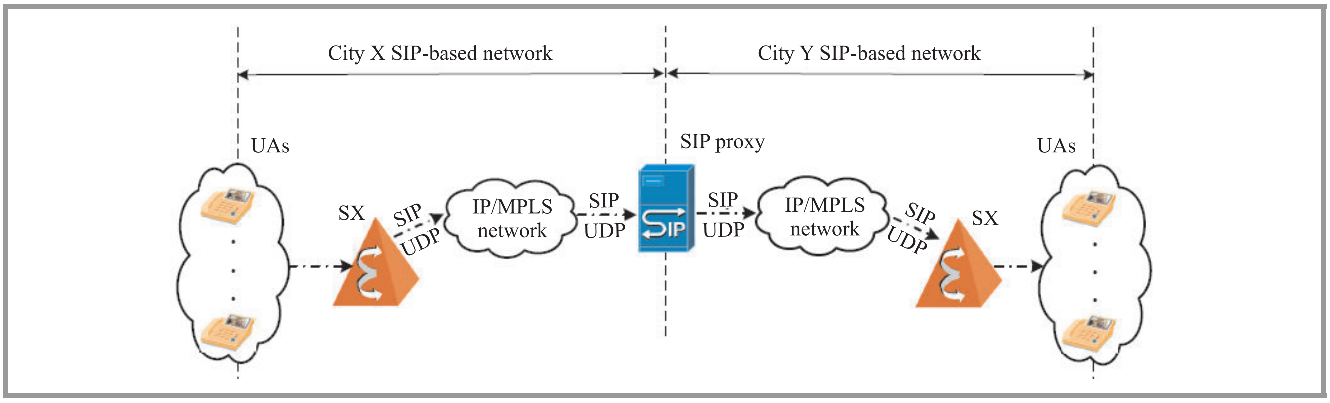

Fig. 1. SIP based transit network between two cities.

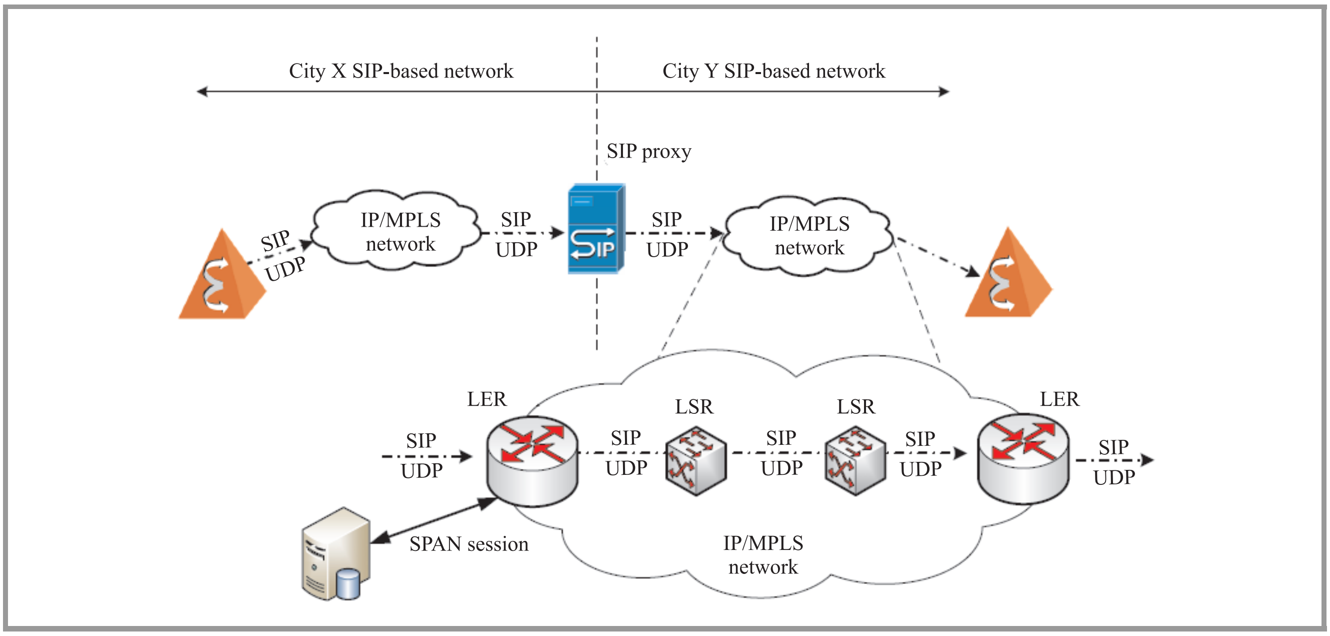

Fig. 2. Scheme of trace collection from transit segment.

network using Signaling System no. 7 (SS7). As the caller and call are located in different regions, the traffic goes through the two transit regional nodes – soft switches of the 5th class. Signalling exchange between them is organized by means of the SIP-I protocol. On the traffic route between soft switches SIP proxy server is set for the purpose of logical separation of regional networks. In the considered case this server performs the functions of the Interconnection Border Controller [16]. Aggregation networks are based on TDM technology with SS7 protocol stack. Besides traffic transit functions, soft switches may perform TMD-IP traffic conversion. Process of an exchange of signaling messages between the regions is completely described by the IETF RFC 3665 [17]. Number of messages that is necessary for call session initialization and termination in the majority of cases does not exceed 8.

Traffic data was captured on SIP proxy server's network interfaces. The organization of traffic measurement described in this paper is based on technical report [18] (Fig. 2). Passive recording system was chosen in order to guarantee the absence of significant influence of measurements on signaling flows. Analysis showed that for data recording it is possible to use standard network interface cards, installed on the data collection server [19]. During experiment L2/L3 switches were loaded up to 30–40% of its maximum throughput capacity. This guaranteed the absence of considerable influence of measurements on the values under test. Further analysis of the collected SIP-I traffic takes into account SIP protocol stack [15], [17].

Signaling information was collected from the network using mirroring technology with subsequent recording on L2/L3 network devices [20]. Traffic collection scheme is depicted in Fig. 2. All traffic circulation between two regions and entering one of L2/L3 switches was mirrored. Its recording was performed on the dedicated server for data recording and storage operation under Linux. Information was gathered with the standard utility (tcpdump) in the open libpcap format [21]. All network equipment which participated

in SIP-I traffic processing was synchronized using NTP protocol [22], [23].
The primary processing of libpcap files with SIP-I traces was carried out using standard functions of Wireshark software [24]. Automation of the processing was done using shell-script. As the result there were obtained data arrays, that contained all necessary information for further analysis of SIP traffic which was carried out in numerical computing environment Matlab [25].
All the SIP-I traces were captured during one week (7 consecutive days, starting from Saturday, 24 hours per day) by means of span session created on one L2/L3 switch. At this node the traffic is duplex. During one workday the maximum number of new call attempts per second (sum for two directions) does not exceed 95, on weekend – 70. Maximum number of concurrent active sessions during busy hour does not exceed 5000 and on weekend 7500. This information is summarized in Table 1.

Table 1
Basic characteristics of new call attempts flow

	Maximum new call attempts per second	Maximum number of concurrent active sessions (busy hour)
Workday	95	5000
Weekend	70	7500

The dynamics of new call arrivals (sum for two directions) during one week is depicted in Fig. 3. It can be seen that from 8 a.m. the load starts to increase and reaches its maximum level at 10 a.m. The sharp decline is observed at 8 p.m. and it continues up to minimum level at 11 p.m. The intensity of new call arrivals for workdays and weekend is somewhat similar. During workdays and weekend it alters in legible boundaries in the period between 10.30 a.m. – 4 p.m.

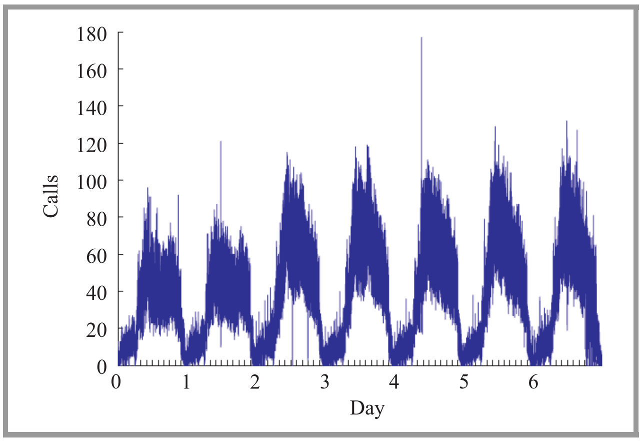

Fig. 3. Total number of new call attempts per second in both directions (one segment covers 2 hours).

The dynamics of new call arrivals (sum for two directions) during one workday (weekend) between 10.30 a.m. and 4 p.m. is depicted in Fig. 4, on weekend in Fig. 5. On

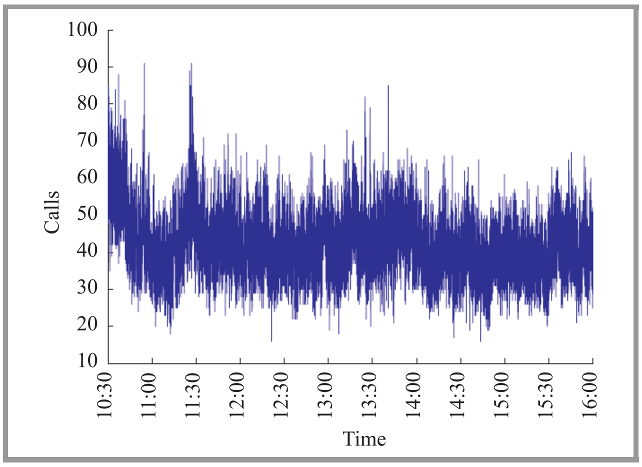

Fig. 4. Dynamic of new call attempts per second during one workday between 10.30 a.m. and 4 p.m.

the weekend one can see that there are not many intense changes. Herewith on a workday behavior is somewhat similar but one can see more abrupt jumps and absolute values of intensity are higher (Table 1).

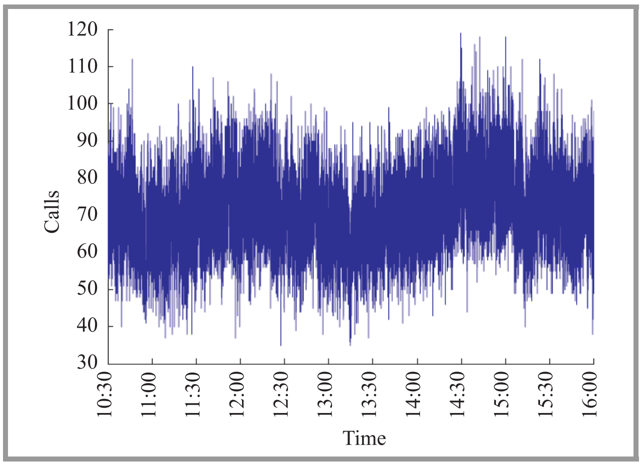

Fig. 5. Dynamic of new call attempts per second during one weekday between 10.30 a.m. and 4 p.m.

3. Building SIP-I Traffic Model

In the network under consideration the maximum size of SIP-I message is restricted by the maximum size of Ethernet frame (1542 bytes) [26]. During analysis the presence of fragmented packets in traffic traces was detected. Minor number of fragmented packets was found only for SIP-I messages that use INVITE, ACK, BYE, CANCEL methods. Among them the biggest number is for INVITE messages which is due to the big size of encapsulated IAM SS7 message. The presence of fragmented packets in the joint SIP-I flow may have impact on processors in network nodes but in the data set under study their number may considered to be insignificant in caparison to the its total volume.

Table 2
Statistical properties of SIP-I data flow

		ALL msg.	INVITE msg.	non-INVITE msg.
Mean	Workday	0.002022	0.028092	0.002359
	Weekend	0.003202	0.0231635	0.003716
Standard deviation	Workday	0.003001	0.0352475	0.003269
	Weekend	0.004909	0.028447	0.005280
Coefficient of variation	Workday	1.48386	2.01778	1.38626
	Weekend	1.5332	1.2283	1.4212
Skewness	Workday	3.16522	3.80822	2.76976
	Weekend	2.97935	2.3496	2.6567
Kurtosis	Workday	21.7262	18.497	17.0056
	Weekend	16.4755	12.255	13.968

Thorough analysis of collected traffic allowed us to estimate the frequency of each SIP-I message (Table 4). During traffic analysis one was interested in building traffic profile by identifying all possible session establishment scenarios. In order to do this all messages in the traces we grouped by Call-ID attribute. The depth of search was restricted by the maximum call length for the considered network which was 30 minutes. Clearly messages with identical Call-ID belong to the same call flow and form certain scenario. Going through the traces one can detect all unique scenarios and estimate their frequency observing that if two call flows consist of the same number and sequence of messages then they belong to one scenario. As the result traffic profile was obtained (Table 5). Very rare and very long scenarios (of 0.1% frequency and less) were left out of scope. Analyzing scenarios from Table 5 one can arrive at the following conclusions, based on assertions coming from RFC 3261 [15]:

- connection is established successfully with reply and correct session termination happens in 20.26% of all call cases (scenarios no. 3, 10, 13, 18, 23, 28, 30, 32, 37);
- attempting to call a busy callee and correct session termination happens in 10,12% of all call cases (scenarios no. 4, 12, 15, 20, 35, 40, 45);
- correct session termination without session establishment due to incorrect dialed number or unavailability of mobile subscriber happens in 25.92% of all call cases (scenarios no. 11, 14, 19, 26, 27, 29, 31, 34, 39, 44);
- connection termination on caller's side before session establishment, connection termination on callee's side, unavailability of signalling network (due to insufficient number of free time-slots etc.) happens in 36.3% of all call cases (scenarios no. 1, 2, 5–9, 16, 17, 21, 22, 24, 25);
- connection termination due to errors during negotiation of session establishment parameters, unexpected behavior of software etc. happens in 7.4% of all call cases (scenarios no. 33, 36, 38, 41–43, 46–48).

Notice that high number of call cases without reply is due to the fact that most of calls are long-distance/international, i.e., callers and callees are in different time zones.

4. Statistical Analysis of SIP-I Data Flow

Based on collected data during period 10.30 a.m. – 4 p.m. for each day of the week (weekend and 5 workdays) two types of samples for further analysis were created. One type of sample was composed only of SIP-I messages interarrival times. Three samples of this type were defined. The first sample (further referred to as ALL msg.) contained timestamps of consecutive message arrivals to SIP proxy server irrespective of their source (from A or from B). The second and third samples (further referred to as INVITE msg. and non-INVITE msg.) contained timestamps of consecutive INVITE (non-INVITE msg.) arrivals to SIP proxy server irrespective of their source. The second type of sample was formed based on the first type. Data in each of the three samples (ALL msg., INVITE msg., non-INVITE msg.) was aggregated over different timescales (i.e., representing number of SIP-I messages in respective time bin). Five time bins was used: 10 ms, 100 ms, 1 s, 10 s, 100 s. Note that in samples of this type SIP-I messages that arrive at SIP proxy server are counted irrespective of their source.

In Table 2 one can find basic statistical properties of the SIP-I data flow based on ALL msg., INVITE msg. and non-INVITE msg. interarrival samples for a workday and a weekend.

For workday (weekend) the value of each parameter is arithmetic average of the values, calculated for each workday (weekend) individually.

It can be seen that standard deviation is far above 1 in all three cases indicating that flows are not Poisson. The data is skewed to the right with sharp peak and long tail as indicated by the values of skewness and kurtosis respectively. In addition, the data is not normally distributed as indicated by the p-value of the Jarque-Bera statistic. There was validated the independence of messages interarrival times and number of arrived messages in disjoint time intervals. It was done using various tests: autocorrelation function (ACF), Box-Ljung statistic and visual inspection of consecutive arrivals (scatter plots). They all rejected independence assumptions.

In Fig. 6 the autocorrelation function for 1000 lags for ALL msg. sample (approx. 10 million consecutive interarrival times) is shown. At small as well as at higher time lags interarrivals are correlated. It is decaying slowly thus suggesting the presence of long-range dependency. In Fig. 7 the autocorrelation function for 1000 lags for ALL msg. sample aggregated over 1 s. time bin is presented. Again significant correlation is observed. Increasing the number of lags or the aggregation level sample (size of time bin) does not change significantly the behavior of ACF.

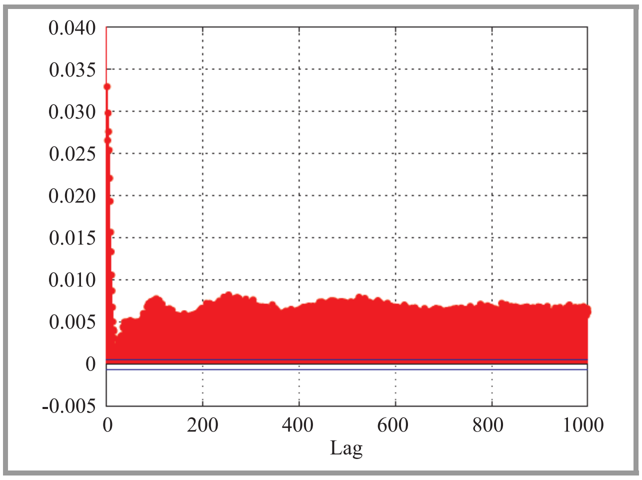

Fig. 6. Autocorrelation function for ALL msg. sample containing consecutive interarrival times and 95% confidence intervals.

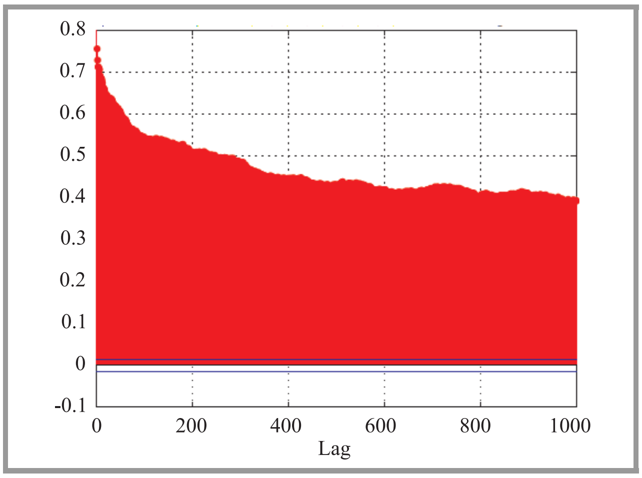

Fig. 7. Autocorrelation function for ALL msg. sample aggregated over 1 s time bin and 95% confidence intervals.

The Box-Ljung statistic shows that interarrival times and number of messages per respective time bin cannot be considered as independent and identically distributed with 95% confidence.

In order to reveal the presence or absence of dependencies in the dataset interarrival times in INVITE msg. sample, ALL msg. sample and its aggregation over different time bins were visually examined using scatter plots (Figs. 8–10). In Fig. 8 the X axis shows the time of i-th packet arrival, and the Y axis shows the time of (i+1)-th packet arrival. The plot is not symmetric and thus dependencies exist. For higher values (above 0.05) weak negative relationship is suggested. For aggregated sample (in Fig. 10) there seems to by weak positive (linear) relationship.

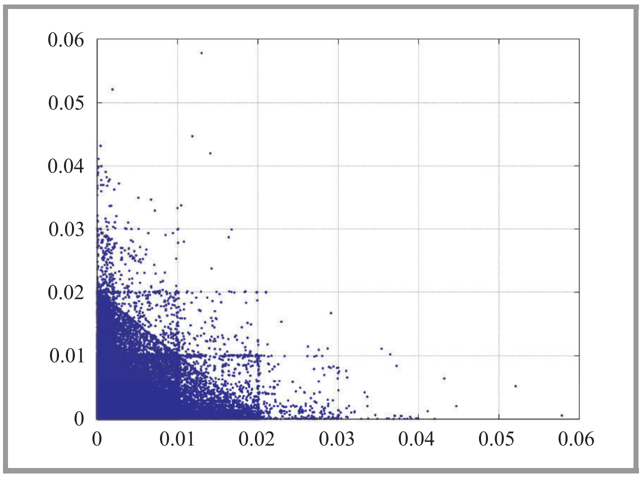

Fig. 8. Scatter plot for ALL msg. interarrival sample.

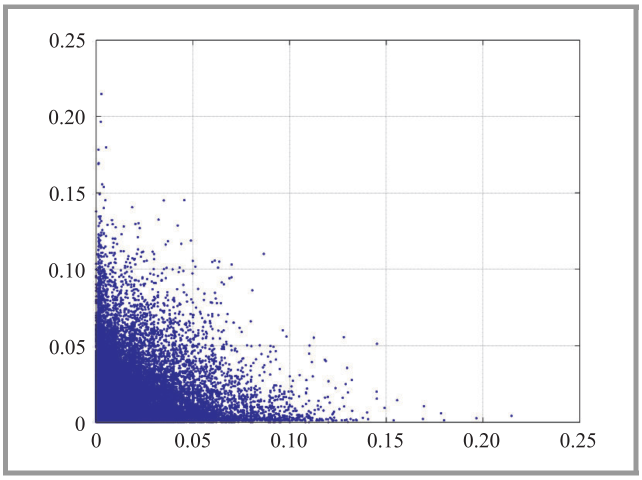

Fig. 9. Scatter plot for INVITE msg. interarrival sample.

Note that the presented results concern data flow consisting of consecutive message arrivals to SIP proxy server irrespective of their source (from A or from B). But one may be interested in nature of INVITE or non-INVITE flows only. Thus it is important to notice that test similar to presented above indicate that in this study one cannot closely approximate flow of SIP-I messages that use INVITE and non-INVITE methods with Poisson distribution.

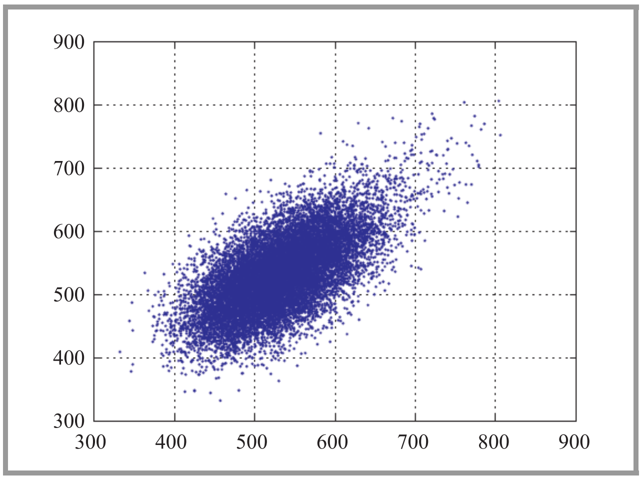

Fig. 10. Scatter plot for ALL msg. sample aggregated over 1 s time bin.

Notice that INVITE msg. sample includes not only INVITE messages that initiate new sessions but any re-INVITE messages that may appear during SIP dialogs. Statistical test show that if one sifts INVITE msg. sample, leaving only INVITE messages that initiate new sessions, such flow is also not Poisson but can be modeled by simple on-off source. Corresponding estimated values of fitted MMPP are given at the end of Section 5.

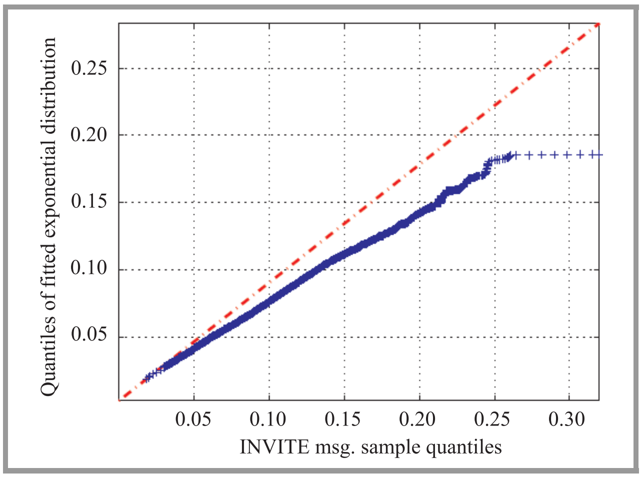

Fig. 11. Q-Q Plot for INVITE msg. sample and simulated exponentially distributed data.

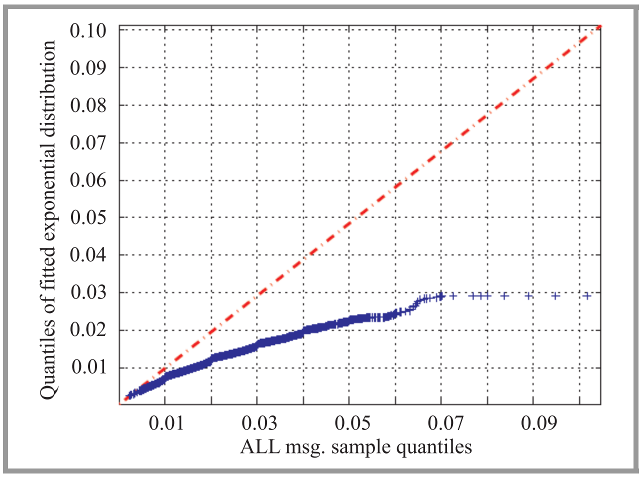

Fig. 12. Q-Q Plot for ALL msg. sample and simulated exponentially distributed data.

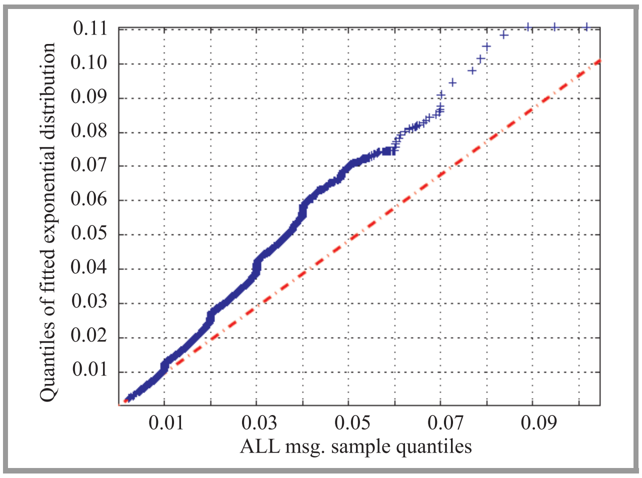

Fig. 13. Q-Q Plot for ALL msg. sample and simulated Weibull distributed data.

For each of the interarrival samples (ALL msg., INVITE msg. and non-INVITE msg.) its components were tested for exponential, Weibull and Pareto distributions using the Kolmogorov-Smirnov, Anderson-Darling and Chi-Squared tests. All tests reject null hypothesis with significance level of 5%. In Figs. 11–13 Quantile-Quantile (Q-Q) plots of different collected samples and data simulated from different distributions is shown. Necessary parameters for simulated distributions were estimated from samples.

Table 3
Estimated values of Hurst parameter from ALL msg. sample

Time bin	Absval	Aggvar	Diffvar	R/S
10 ms	0.8556	0.8583	0.5411	0.9001
100 ms	0.9435	0.9488	0.7056	0.9754
1 s	0.9578	0.9648	0.9272	0.9751
10 s	0.9485	0.9563	1.1138	0.9000
100 s	0.9414	0.9384	0.8509	0.6932

Finally for each of the aggregated samples Hurst parameter was calculated using the following methods: absolute moment, aggregate variance, difference variance, and R/S [27]. The estimated values for ALL msg. sample are given in Table 3.

Table 3 shows that basically Hurst exponent value does not vary significantly for bigger time scales (around 0.9)[1].

[1] Yet there exist several outliers depending on the estimation method.

This suggests that the arriving SIP-I messages to SIP proxy server (irrespective of source) is a persistent process with long-term memory.

5. Fitting SIP-I Trace to MMPP

In Fig. 14 ALL msg. sample aggregated over 1 s time bin is depicted. The plots for 10 s and 100 s time bins are similar. The arrival rates appear to vary among several distinct values, which is consistent with the path behavior of MMPP.

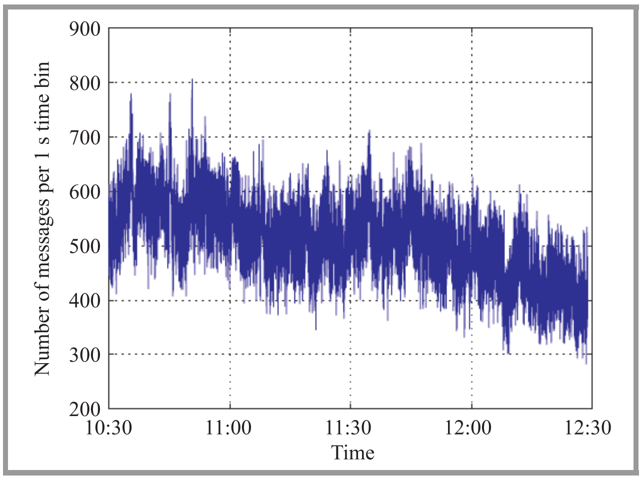

Fig. 14. ALL msg. sample for one workday aggregated over 1 s time bin.

Algorithm Lambda was used to fit ALL msg. sample aggregated over different time scales to a MMPP [28]. It was found that in this study MMPP model was able to adequately model collected data. For aggregation over 10 ms, 100 ms, 1 s, 10 s, 100 s time bins the number phases in fitted MMPP varies from 3 to 31. Here we restrict ourselves to the Q-Q plot as the measure of the goodness of fit of the model. In Fig. 15 the quantiles of the ALL msg. sample aggregated over 1 s time bin and of a simulation of the fitted process are shown. If both sets of data are drawn from the same distribution the plot is expected to be linear. It can bee seen, that the fit appears to be fairly good.

Phase transitions of fitted MMPP from Fig. 15 are governed by the infinitesimal matrix (generator) Q_6 of the form

$$Q_6 = \begin{pmatrix} -0.5275 & 0.4835 & 0.0440 & 0 & 0 & 0 \\ 0.0218 & -0.5679 & 0.5218 & 0.0253 & 0 & 0 \\ 0.004 & 0.1165 & -0.3634 & 0.2399 & 0.0066 & 0 \\ 0 & 0.0095 & 0.3075 & -0.4294 & 0.1099 & 0.0025 \\ 0 & 0 & 0.0274 & 0.4327 & -0.4874 & 0.0274 \\ 0 & 0 & 0 & 0.1739 & 0.7826 & -0.9565 \end{pmatrix}.$$

The vector of estimated arrival rates equals $\vec{\lambda}_6 =$(751.1845, 645.5535, 547.9225, 458.2916, 376.6606, 303.0296).

There was made an attempt to fit collected data to MMPP with two phases using interarrival statistic instead of frequency one. The goal was to check whether the simplest

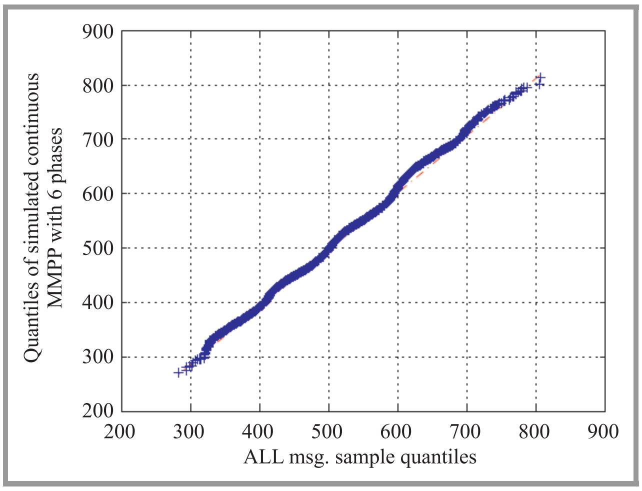

Fig. 15. Q-Q plot of the ALL msg. sample aggregated over 1 s time bin and simulated data, using fitted (with Lambda algorithm) MMPP with 6 phases.

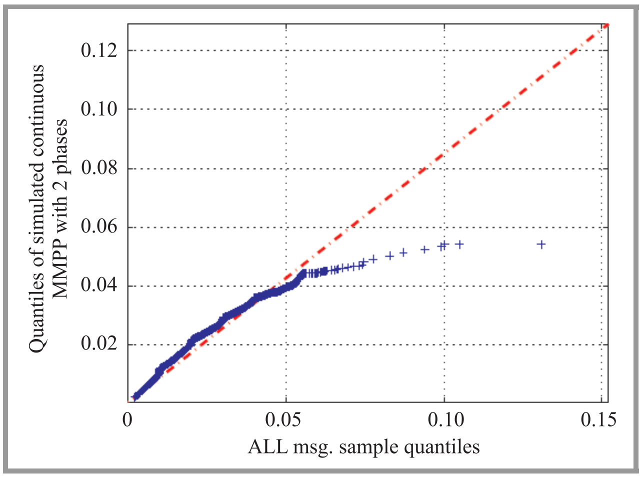

Fig. 16. Q-Q plot of the ALL msg. sample and simulated data, using fitted continuous two-phase MMPP.

MMPP model may be fairly accurate as well. In order to do this parameters of two-phase MMPP flow based on ALL msg. sample were estimated using KPC Toolbox [29]. Then synthetic data was generated using built in KPC Toolbox MAP random sample generator. Quantiles of the ALL msg. sample and of simulated fitted two-phase MMPP process are shown in Fig. 16.

Here phase transitions are governed by the infinitesimal matrix (generator) Q_2 of the form

$$Q_2 = \begin{pmatrix} -276.3723 & 276.3723 \\ 300.6946 & -300.6949 \end{pmatrix},$$

and vector of estimated arrival rates equals $\vec{\lambda}_2$=(63.7806, 978.6974).

One may observe too many outliers at high quantiles and fitting is not so accurate as with MMPP with more than 2 phases. Thus MMPP with two phases cannot adequately capture the behavior of real flow of SIP messages in the considered scenario (Fig. 1) and its use may underestimate impact of input flow on SIP proxy server performance.

It is worth mentioning that two-phase MMPP model (on-off source) turned out to be better than Poission for modeling of INVITE flow, consisting only of messages that initiate sessions. In Figs. 17–18 one can see Q-Q plots of the sifted INVITE msg. sample and simulated data, using fitted exponential distribution and continuous two-phase MMPP correspondingly. The estimated infinitesimal generator Q_{INVITE} of corresponding two-phase MMPP has the form

$$Q_{INVITE} = \begin{pmatrix} -182.9091 & 182.9091 \\ 129.2100 & -129.2100 \end{pmatrix},$$

and vector of estimated arrival rates equals $\vec{\lambda}_{INVITE}$=(165.4, 0).

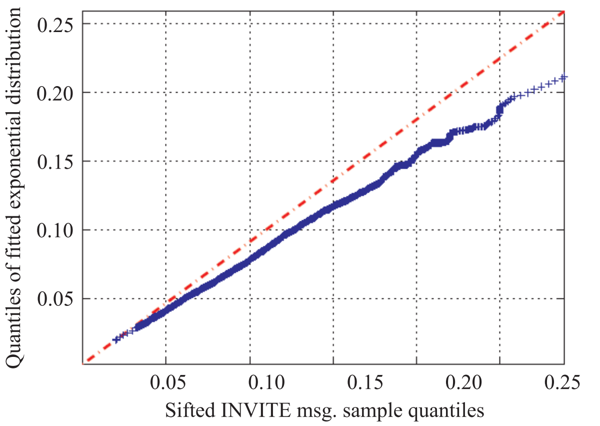

Fig. 17. Q-Q plot of the sifted INVITE msg. sample and simulated exponentially distributed data.

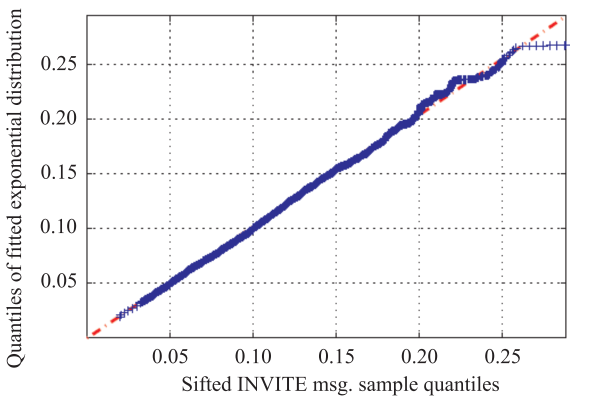

Fig. 18. Q-Q plot of the sifted INVITE msg. sample and simulated data, using fitted continuous two-phase MMPP.

Following [28] we tried to test the ability of the fitted on-off source to capture the features of flow of INVITE messages that initiate new sessions. In order to do this we simulated infinite buffer queue with constant service time using discrete-event simulation approach and fed this queue with sifted INVITE msg. sample, which contained interarrival times between consecutive arrivals of INVITE messages that initiate new sessions. The traffic intensity was varied by changing the service time, i.e., as mean time between consecutive arrivals in the considered sample equals ≈ 0.0146 sec., service time was varied between 0.009 and 0.0131. From simulation model we computed the mean number of customers in the system. We also computed mean number of customers in $MMPP|D|1$ using analytical expressions with infinitesimal generator and arrival rates given by Q_{INVITE} and $\vec{\lambda}_{INVITE}$. The results are shown in Fig. 19.

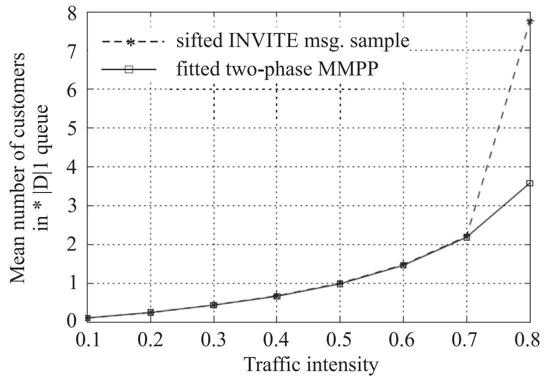

Fig. 19. Comparison of mean number of customers in $\cdot|D|1$ queue with sifted INVITE msg. sample as input flow and fitted continuous two-phase MMPP as input flow.

Mean delays computed from simulation and analytic models almost coincide for the traffic intensity less than 0.7 and in this region MMPP captures the relevant features of the sifted INVITE msg. sample. But at traffic intensities above 0.75 mean delay from simulation model is significantly larger than the one obtained from analytic expressions, which is consistent with observations in other studies, e.g. [28], [30].

6. Conclusion

In this paper the analysis of interregional SIP-I signaling traffic which circulates in telecommunication operator's network is presented. Traffic profile is built and typical call scenarios are identified. Statistical tests show that considered joint SIP-I traffic flow (and its INVITE component) is not Poisson and exhibits long range dependence. Traffic may be accurately fitted with MMPP with 3–31 phases. Based on obtained results further research will be devoted to estimating control parameters in mathematical models that represent SIP servers with embedded hysteretic load control mechanism and real network simulation.

Acknowledgements

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APPENDIX

Table 4

Size and frequencies of joint SIP-I messages flow

Message type	Total	Mean size (bytes)	Percent
ACK	2232516	476	14.03
BYE	5636610	606	3.54
INVITE	22341251	1314	14.04
CANCEL	7624459	451	4.79
INFO	212417	597	0.13
UPDATE	1	550	0
100	22063983	382	13.87
180	9810796	1329	6.16
183	33221739	937	20.88
200	19329687	713	12.15
400	18865	495	0.01
403	67135	593	0.04
404	544597	592	0.34
408	27792	277	0.01
410	16040	588	0
415	21	614	0
480	5090740	606	3.20
481	4046	422	0
483	944	944	0
484	23771	598	0.01
486	1965894	592	1.23
487	7666305	518	4.82
488	13153	520	0
500	343735	604	0.21
501	833	833	0
502	435713	595	0.27
503	244849	552	0.15
504	9124	634	0
604	455	578	0

Table 5

Typical session establishment scenarios and their frequencies

No.	Sequence of messages	Percent
1	INVITE, 100, CANCEL, 200, 487, ACK	1.14
2	INVITE, 100, 180, CANCEL, 200, 487, ACK	2.28
3	INVITE, 100, 180, 200, ACK, BYE, 200	0.97
4	INVITE, 100, 180, 486, ACK	0.42
5	INVITE, 100, 183, CANCEL, 200, 487, ACK	4.62
6	INVITE, 100, 183, 180, CANCEL, 200, 487, ACK	14.84
7	INVITE, 100, 183, 180, 183, CANCEL, 200, 487, ACK	0.85
8	INVITE, 100, 183, 180, 183, 183, CANCEL, 200, 487, ACK	0.32
9	INVITE, 100, 183, 180, 183, 183, 183, CANCEL, 200, 487, ACK	0.26
10	INVITE, 100, 183, 180, 183, 200, ACK, BYE, 200	0.11
11	INVITE, 100, 183, 180, 183, 480, ACK	0.33
12	INVITE, 100, 183, 180, 183, 486, ACK	0.16
13	INVITE, 100, 183, 180, 200, ACK, BYE, 200	9.72
14	INVITE, 100, 183, 180, 480, ACK	0.99
15	INVITE, 100, 183, 180, 486, ACK	3.37
16	INVITE, 100, 183, 183, CANCEL,200, 487, ACK	3.14
17	INVITE, 100, 183, 183, 180, CANCEL, 200, 487, ACK	4.96
18	INVITE, 100, 183, 183, 180, 200, ACK, BYE, 200	1.95
19	INVITE, 100, 183, 183, 180, 480, ACK	0.20
20	INVITE, 100, 183, 183, 180, 486, ACK	1.02
21	INVITE, 100, 183, 183, 183, CANCEL, 200, 487, ACK	2.36
22	INVITE, 100, 183, 183, 183, 180, CANCEL, 200, 487, ACK	0.11
23	INVITE, 100, 183, 183, 183, 180, 200, ACK, BYE, 200	0.42
24	INVITE, 100, 183, 183, 183, 183, CANCEL, 200, 487, ACK	1.15
25	INVITE, 100, 183, 183, 183, 183, 183, CANCEL, 200, 487, ACK	0.27
26	INVITE, 100, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 183, 480, ACK	0.19
27	INVITE, 100, 183, 183, 183, 183, 183, 480, ACK	0.33
28	INVITE, 100, 183, 183, 183, 183, 200, ACK, BYE, 200	0.70
29	INVITE, 100, 183, 183, 183, 183,480, ACK	1.61
30	INVITE, 100, 183, 183, 183, 200, ACK, BYE, 200	0.09
31	INVITE, 100, 183, 183, 183, 480, ACK	5.09
32	INVITE, 100, 183, 183, 200, ACK, BYE, 200	2.57
33	INVITE, 100, 183, 183, 404, ACK	0.22
34	INVITE, 100, 183, 183, 480, ACK	6.86
35	INVITE, 100, 183, 183, 486, ACK	0.69
36	INVITE, 100, 183, 183, 502, ACK	0.47
37	INVITE, 100, 183, 200, ACK, BYE, 200	3.73
38	INVITE, 100, 183, 404, ACK	0.65
39	INVITE, 100, 183, 480, ACK	6.72
40	INVITE, 100, 183, 486, ACK	1.73
41	INVITE, 100, 183, 500, ACK	0.83
42	INVITE, 100, 183, 502, ACK	1.51
43	INVITE, 100, 404, ACK	2.04
44	INVITE, 100, 480, ACK	3.60
45	INVITE, 100, 486, ACK	2.73
46	INVITE, 100, 500, ACK	0.53
47	INVITE, 100, 502, ACK	0.10
48	INVITE, 100, 503, ACK	1.05

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Paper

Predictive Modeling in a VoIP System

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Abstract—An important problem one needs to deal with in a Voice over IP system is server overload. One way for preventing such problems is to rely on prediction techniques for the incoming traffic, namely as to proactively scale the available resources. Anticipating the computational load induced on processors by incoming requests can be used to optimize load distribution and resource allocation. In this study, the authors look at how the user profiles, peak hours or call patterns are shaped for a real system and, in a second step, at constructing a model that is capable of predicting trends.

Keywords—particle algorithms, prediction, user-profiles, VoIP.

1. Introduction

In the past few years, many researchers focused on deploying and improving the Voice over IP (VoIP) technology. VoIP platforms are subject to extremely fast context changes due to the dynamic pricing and automatic negotiation, availability and competition for resources in shared environments. One thus needs to provide powerful prediction models that are able to automatically evolve and adapt in order to consistently deal with the varying nature of a VoIP system. Furthermore, as such systems generally do not allow having a centralized management, mainly due to scale concerns, one would also like to have a modular approach for, e.g., resource allocation or load balancing [1].

This study is built on a collaboration between the University of Luxembourg and MixVoIP, a company that hosts and delivers commercial VoIP services, with an important market share in the Luxembourg area and, at this time, a significant number of subscribed clients [2]. The clients of MixVoIP are small businesses, libraries, small companies. VoIP is any type of technology that can transmit, in real-time, converted voice signals into digital data packets over an IP network. A VoIP service, e.g., as implemented and provided by a public or private company, allows placing calls over the Internet via an ordinary phone or computer. A VoIP phone only needs to connect to a home computer network using a special adapter. One of the main advantages of using VoIP in enterprises for example is the reduced cost of sharing a certain number of external phone lines, furthermore avoiding the allocation of a line per user. The most well known telephone system capable of switching calls and of providing a powerful control over call activity (through the use of channel event logging) is Asterisk [3]. It is a framework for building multi-protocol, real-time communications solutions. Asterisk is in charge of establishing and managing the connection between two end devices by sending the voice portion of the call and everything that is not voice, also known as overhead. The protocols in charge of controlling multimedia communication sessions, respectively of delivering information and transferring data are Session Initiation Protocol (SIP), and Real Time Protocol (RTP). SIP is a signaling communications protocol, widely used for controlling multimedia communication sessions such as voice and video calls over Internet Protocol (IP) networks. SIP is one of the most known protocols in charge of signaling, establishing presence, locating users, setting, modifying or tearing down sessions between end-devices. After the connection is established, the media transportation is done via RTP. Codecs are used for converting the voice portion of a call in audio packets and the conversation is transmitted over RTP streams. The calls are stored by the VoIP providers in a Call Detail Record (CDR) database for billing.

An example of a telephone system solution for VoIP is given in Fig. 1. The users connect via Internet to a server that runs either on a physical machine or in a data center or in the cloud. All users must be registered to a SIP Registrar Server in order to indicate the current IP address and the URLs for which they would like to receive calls. The security layer can include SSH, FTP and access control or sessions for password protected phone login. The voice nodes control call features such as voice mail, call transfer, conference functions. The solutions are mainly deployed on Linux distribution. After authorizing the user to place calls, two links are made. First, the user calls Asterisk that, in turn, connects to the database server in order to check the credentials of the user and if the call can be made, e.g., credit on a pre-pay account. Second, Asterisk tries to reach the other end-device. After the call is finished, when the user hangs-up, call detailes are stored in the CDR (e.g. client id, destination, prefix, call duration, billing). The situation presented above stands for outbound calls placed by the users of the VoIP service provider. Depending on the codecs of each device on the call path, the decoding and re-coding operations increases the CPU usage. The VoIP provider is in charge of forwarding the call to different telecom operators. For incoming calls, the car-

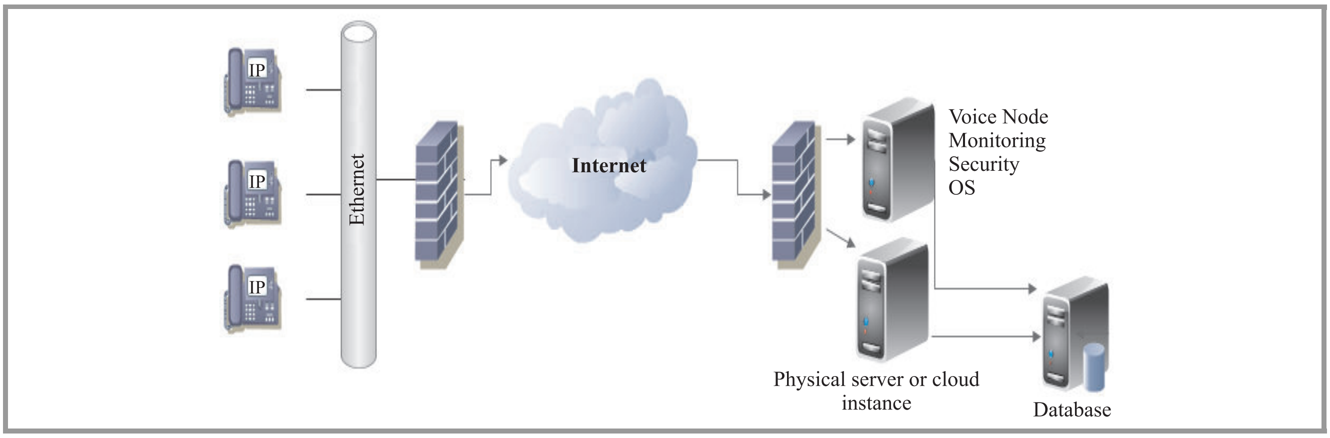

Fig. 1. Basic example of VoIP telephone system.

rier connects to the VoIP provider which, in turn, looks up for the number and returns the number and location where the user can be reached.

Together with MixVoIP the authors work on a next generation cloud-based product range, a model that adapts to and copes with the highly dynamic evolution of requests, load or other stochastic factors. At the same time, in order to deal with a model based on an extremely large number of parameters, all in a time-dependent framework we look at dynamic factors and stochastic processes while taking into account repeating activity patterns, failures, service level agreements. Thus, one needs to know how to define time dependency in order to bridge, in a coherent manner, online (real life) processes, the concepts employed in modeling those aspects, and the (classically) static representations used in optimization. As an example, a specific aspect one can exploit is that activity patterns inside such systems tend, over specific periods of time, to express regular or localized characteristics, e.g., the evolution of energy price when relying on renewable sources, cyclic load over a 24 hours span.

The final goal of the project developed by the University of Luxembourg and MixVoIP company is to implement such an approach within MixVoIP's environment, i.e., capable of providing a prediction system for cloud-based platform. The project can be split into two main research areas: prediction and load balancing. Currently, we focus on building the prediction model (in charge of anticipating the number of the calls in the system) and be used as an input for the dynamic load balancing. The transition to a direct implementation for the cloud environment, to develop intelligent load balancing mechanisms that optimally spread the traffic inside the cloud, is therefore needed. To this end, with the support of MixVoIP, the paradigms designed inside the project will also be put into practice, hence resulting into a fully functional predictive optimization system for real cloud based VoIP platforms. The mechanism should be robust, flexible and scalable. Moreover, the load balancing mechanisms should address several requirements like increased scalability, high performance, high availability or failure recovery.

The need for resource capabilities arises as the cost-saving benefit of dynamic scaling is brought by the cloud phenomenon and given that cloud computing uses virtual resources with a non eligible setup time. Prediction is therefore necessary. Statistical models that describe resource requirements in cloud computing already make the object of pay-as-you-go, e.g., providers of environments that scale transparently in order to maximize performance while minimizing the cost of resources being used [4]. By predicting the load, allocating tasks to processors and dynamically turning off reserve computational resources, the high power required by cloud computing system can be reduced drastically [5].

Cloud computing can be seen as an overlay where seamless virtualization is implemented while dealing with privacy constraints. It relies on sharing computing resources rather than on having local servers or personal devices to handle applications. Cloud computing is used to increase capacity or add capabilities on the fly without investing in new infrastructure, training new personnel, or licensing new software. As a specific aspect, one may consider a computational demand and offer scenarios, where individuals or enterprizes negotiate and pay for access to resources through virtualization solutions (administered by a different entity that acts as provider) [6]. Cloud computing encompasses any subscription-based or pay-per-use service that, in real time over the Internet, extends IT's existing capabilities [7]. Demanding parties may however have diverging requirements or preferences, specified by contractual terms, e.g., stipulating data security, privacy or the service quality level [8]. Moreover, dynamic and risk-aware pricing policies may apply where predictive models are used either in place or through intermediary brokers to assess the financial and computational impact of decisions taken at different time moments. Legal enforcements may also restrict access to resources or data flow, e.g., data crossing borders or transfers to different resource providers. As common examples, one can refer to Amazon Web Services [9] or Google Apps Cloud Services [10].

The impact of task scheduling and resource allocation in dynamic heterogenous grid environments, given independent jobs has been studied in [11]. The authors developed a hierarchic genetic schedule algorithm, capable of delivering high quality solutions in reasonable time; makespan and flowtime are the two objectives considered for the optimization problem (minimization). The algorithm improves the task execution time across the domain boundaries and the results are compared with mono-population and hybrid genetic-based schedulers. The allocation of tasks to processors in new distributed computing platform is challenging when interconnecting a large number of processing elements and when handling data uncertaintie. In [12] a scheduler in charge of stabilizing this process is presented. The authors discuss an algorithm where the application graph is decomposed into convex sets of vertices and it reduces the effects of disturbances in input data at runtime.

With respect to the previous considerations, the paper presents an algorithm capable of predicting the number of calls placed in a VoIP system during a given time frame. The data collected from MixVoIP is analyzed and user profiles are outlined. It has been observed that there are patterns in the number of calls placed during the different hours of a day, peak hours or in abnormal situations. During public holidays the number of calls heads to less than 10 per hour and, during a normal day, there are peak hours where the servers are overloaded or hours when the traffic is very low. While MixVoIP has chosen to have over capacity in order to avoid dropping the calls when there is a big traffic in the system, there are reasons to improve this approach and to implement a new model capable of adapting to the number of calls while optimally allocating resources. Statistics are used as an input for the prediction model which, based on a chosen time frame, can estimate the number of calls placed during the next time frame.

The rest of the paper is organized in the following manner. Section 2 discusses relevant background related to prediction models, and motivates the importance of predicting the load. In Section 3 the prediction model and detailed information is introduced. Section 4 presents experiments, results. Finally Section 5 concludes and gives remarks about future work.

2. Related Work

A series of different studies looked in the past few years on prediction models and VoIP. However, most existing approaches are based on predicting the speech quality of VoIP. The impact of packet loss and delay jitter on speech quality in VoIP has been studied for example by Lijing Ding in [13]. He proposes a formula used in Mean Opinion Score (MOS) prediction and network planning. A parametric network-planning model for quality prediction in VoIP networks, while conducting a research on the quality degradation characteristic of VoIP, was presented by Alexander Raake in [14]. The work addresses to the different technical characteristics of the VoIP networks linked with the features perceived by user. He gives a detailed description of VoIP quality and discusses how wideband speech transmission capability can improve telephone speech quality.

In [15] the authors present a solution for non-intrusively live-traffic monitoring and quality measuring. Their solution allows adapting to new network conditions by extending the E-Model proposed by the International Telecommunication Union-Telecommunication Standardisation Sector (ITU-T) to a less time-consuming and expensive model. Another model for objective, non-intrusive, prediction of voice quality for IP networks and to illustrate their application to voice quality monitoring and playout buffer control in VoIP networks is presented in [16]. They develop perceptually accurate models for nonintrusive prediction of voice quality capable of avoiding time consuming subjective tests and conversational prediction voice non-intrusive models quality for different codecs.

The problem of traffic anomaly detection in IP networks has been studied in [17]. The author presents an easy closed form for prediction of the mean bit-rate of one conversation generated by SID-capable speech codecs as a function of the codec and the number of frames per packet used. Reference [18] explores the cumulative traffic over relatively long intervals to detect anomalies in voice over IP traffic, to identify abnormal behavior when different thresholds are exceeded.

While extensive studies exist in each of the mentioned areas, only few sources consider such a holistic approach and analysis. Also, no conclusive agreement in the optimization domain exists on how to deal with highly dynamic time-dependent systems, causality and impact in a predictive framework, information coherence or descriptive power of the models when facing a fast changing environment where different scenarios are possible. Last, to the best of our knowledge, the use of such highly integrative techniques in a real world setup has not been addressed to this extent ever before and would therefore represent a premiere for the cloud based VoIP commercial domain.

3. Algorithm

The final goal of the project developed by the University of Luxembourg and MixVoIP is to implement an approach that, first, improves the VoIP service and that, second, scales to a cloud-based-solution. In this section an algorithm based on interactive particle algorithms [19] is presented, adapted, e.g., for VoIP. It allows predicting the traffic in servers, the number of calls, based on previous observations. The model is namely based on interacting particle algorithms used for parameter estimation, given a Gaussian model [20]. Pierre Del Moral and Arnaud Doucet define interactive particle methods as extension of Monte Carlo methods, allowing to sample from complex high dimensional probability distributions. The algorithm can estimate normalizing constants, while approximating the target probability distributions by a large cloud of random samples termed particles. Each particle can evolve randomly in the space and, based on its potential, can sur-

Algorithm 1 Estimation of parameters pseudo-code

Generation of Particles, $P(\mu,\Sigma)$
Fix some population size, N
Draw $X \sim \mathscr{N}_d(\mu,\Sigma)$,
for $i = 1 \to N$ **do**
{ For each particle }
Generate μ from $\mathscr{N}(0,1)$ e.g. Box Müller
Determine the (lower) triangular matrix A via a Cholesky decomposition of Γ as AA^T
Calculate $X \leftarrow \mu + A\mathscr{N}(0,I_d)$, d iid variable from $\mathscr{N}(0,1)$ e.g. Box Müller
Calculate $\Sigma \leftarrow \sum_{i=1}^{n} X_i X i^T$
Likelihood $L = (2\pi)^{-N\times d/2} \times |\Sigma|^{-N/2} \times \exp^{\sum_{i=1}^{N} \frac{-(x-\mu)^T \times \Sigma^{-1} \times (x-\mu)}{2}}$
end for

Perturbation of Particles
for $k = 1 \to steps$ **do**
Perturb the encoded vector $W \equiv \{X_i \sim \mathscr{N}_d(0,\Sigma)\}$
Draw samples $\{Y_i \sim \mathscr{N}_d(0,\Sigma)\}$, $1 \le i \le n$
Construct a new vector $\{X_i \leftarrow \sqrt{a} \times X_i + \sqrt{1-a} \times Y_i\}$, $1 \le i \le n$
Perturb μ, $\mu \leftarrow \mu + val$, val generated from $\mathscr{N}(0,1)$
Calculate new sigma, $\Sigma \leftarrow \sum_{i=1}^{n} X_i X i^T$
Calculate $Lnew$, likelihood with new Σ and μ
if $Lnew > L$ **then**
Set the new values of Σ,μ in the particle
end if
end for

Algorithm 2 Prediction Step

Choose the prediction method
Select the particle with maximum likelihood and extract μ and Σ that best describe the observed data
Let the distribution be $\mathscr{N}_d(\mu,\Sigma)$
For $Z \sim \mathscr{N}_d(\mu,\Sigma)$ we consider the following partition:

$$\mu = \begin{bmatrix} \mu_d^{\alpha} \\ \mu_d^{\beta} \end{bmatrix},$$

$$\Sigma_d = \begin{bmatrix} \Sigma_d^{\alpha_1} & \Sigma_d^{\alpha_2} \\ \Sigma_d^{\beta_1} & \Sigma_d^{\beta_2} \end{bmatrix}$$

$Z^{\alpha}|Z^{\beta} = z \sim \mathscr{N}(\mu^c,\Sigma^c)$
$\mu^c = \mu_d^{\alpha} + \Sigma_d^{\alpha_2} \times (\Sigma_d^{\beta_2})^{-1} \times (z - \mu_d^{\beta})$
$\Sigma^c = \Sigma_d^{\alpha_1} - \Sigma_d^{\alpha_2} \times (\Sigma_d^{\beta_2})^{-1} \times \Sigma_d^{\beta_1}$

Prediction based on the likelihood of each particle, weighted likelihood and the training test
Let $Ltotal = \sum_{i=1}^{N} L$
for $i = 1 \to size(samples)$ **do**
$Z^{\beta} = \frac{\sum\limits_{i=1}^{N} Z^{\alpha} \times L_i}{Ltotal}$
end for

vive or not. If a particle does not survive it will not be brought back. Many applications took benefit of this intuitive genetic mutation-selection type mechanism in areas as nonlinear filtering, Bayesian statistics, rare event simulations or genetic algorithms. The sampling algorithm applies mutation transitions and includes an acceptance – rejection selection type transition phase. To this end, the approach is closely related to evolutionary-life algorithm, though while providing theoretical error bounds and performance analysis results. For a full description, please refer to the work of Pierre del Moral [19]. The algorithm starts with N particles, denoted by ξ_0^i, $1 \le i \le N$, that evolve according to a transition $\xi_0^i \to \xi_1^i$, given a fix set A. If $\xi_1^i \in A$, it will be added to the new population of N individuals $((\hat{\xi}_1^i)_{1\le i\le N})$, else being replaced by an individual randomly from A. The sequence of genetic type populations is defined by $\xi_n := (\xi_n^i)_{1\le i\le N} \xrightarrow{selection} \hat{\xi}_n := (\hat{\xi}_n^i)_{1\le i\le N} \xrightarrow{mutation} \xi_{n+1}$. $\hat{\xi}_{n-1}^i \to \xi_n^i$, can be seen as a parent. An overview of convergence results including variance and mean error estimates, fluctuations and concentration properties is also given in [19].

For the estimation of parameters, a population of particles is generated as in the following. Each particle encodes a mean vector μ of size d, a matrix, $W \equiv \{X_i \sim \mathscr{N}_d(0,\Sigma)\}$ sampled from a Wishart distribution $W(\Sigma,d,n)$. It is used to model random covariance matrices and describes the probability density function of random nonnegative-definite $d \times d$ matrices [21]. The parameter n refers to the degrees of freedom while Γ in Algorithm 1 and denotes a scale matrix. After generating the initial population of particles, a perturbation step is repeated for a given number of times with a pre- specified constant value, a. The perturbation step consists in modifying the parameters of each particle subject to distribution invariance constraints. We generate a new value val from $\mathscr{N}_d(0,\Sigma)$ and $\mu \leftarrow \mu + val$ is calculated, and draw new samples $\{Y_i \sim \mathscr{N}_d(0,\Sigma)\}$, $1 \le i \le n$. The encoded vector is recalculated based on vector Y and value a, as described in the algorithm. After building the new Σ, the likelihood of the perturbed particle is determined. If the value of the likelihood is improved, then the perturbed particle is accepted and the old one is deleted from the population. This step is repeated for a number of times, in order to determine the values of the parameters that maximize the likelihood. The perturbation can be seen as a transition or mutation of each particle.

After the last perturbation step we get the final population. There are two prediction methods presented in Algorithm 2. First, the particle with the maximum likelihood is chosen, and then the parameters with the first component (Z^{α}, input for testing) are obtained, the second component (Z^{β}) is extracted. Second, we take in consideration the likelihood of all the particles. The component Z^{β} is calculated via a weighted sum of the likelihoods multiplied by the first component and divided by the sum of likelihoods. The result consists in an estimation of the traffic during the second time frame, for every working day of the chosen year.

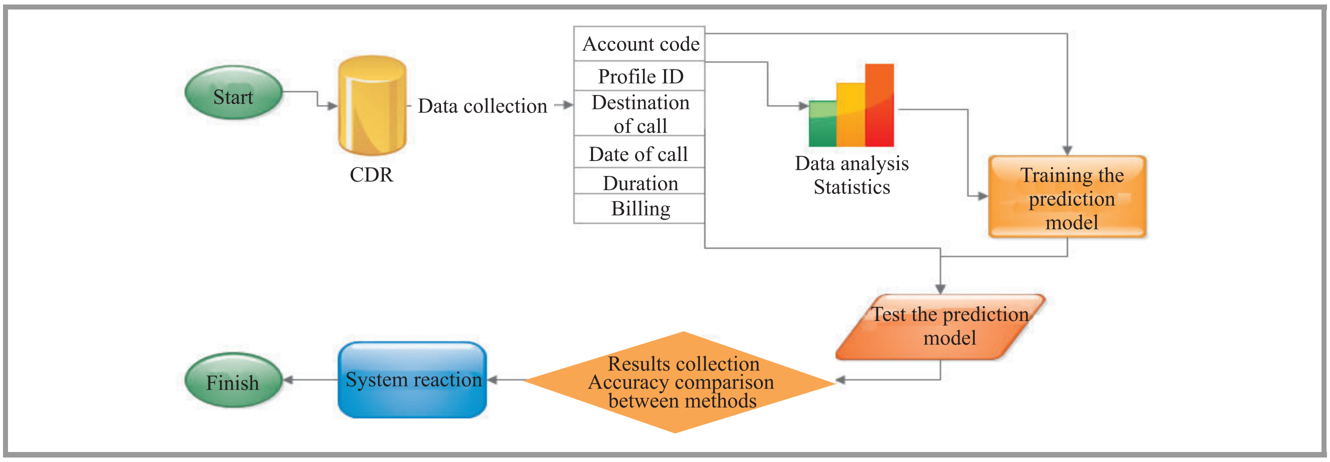

Fig. 2. Data collection, data analysis, prediction, decision.

4. Experiments and Results

The prediction model has been trained and tested on real data from MixVoIP. In Fig. 2 is given a description of the path leading from data collection to data analysis, prediction and then modification of the system. The first step collects and analyzes the information from the detailed records of placed calls, e.g., the account code and profile id. One profile id can have multiple account codes. We can categorize the users based on the number of calls placed and treat the main customers with a higher priority. The destination of the call is also recorded and, after analyzing the data we are able to see that, based on the location of the specific services, the calls are more likely to be placed in that area. The duration of the calls is an useful information along with the date of the call.

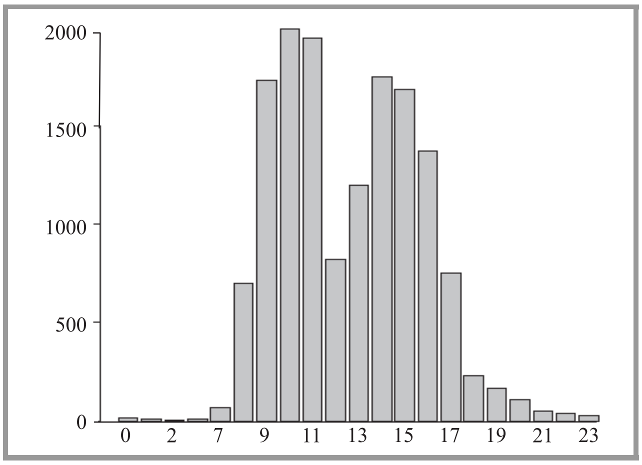

Fig. 3. The total number of calls placed in the third week of April 2012.

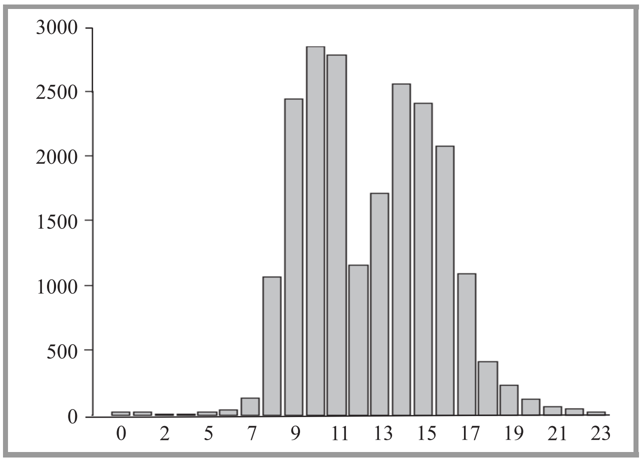

Fig. 4. The total number of calls placed in the second week of November 2012.

Figures 3 and 4 are two examples of the total calls placed during two different weeks from 2012. Throughout a day, the highest traffic is during working hours (8–11 and 13–18). Due the profile of the clients at MixVoIP, we can see in Figs. 5 and 6 that, over a week, the traffic is high from Monday till Friday, during working hours. Abnormal situations can be revealed if the load of the server becomes high outside the detected normal intervals and days as we can see in Fig. 7. This information is used as input to predict the load of the servers during working days and peak hours. As training set, the calls placed in 2012 from hour 10 to hour 11 is chosen. We compared the accuracy of the presented prediction methods with a feed-forward neural network approach [22].

Neural networks are widely used for prediction problems, classification or control, in areas as diverse as finance, medicine, engineering, geology or physics. An artificial neural network is a computational model inspired from the structure and function of biological neural networks. It is considered to be a strong nonlinear statistical data modeling tool where the complex relationships between inputs and outputs are modeled or patterns are found. Pattern classification, function approximation, object recognition, data decomposition are only a few examples of problems where artificial neural networks were applied. An artificial neuron receives a number of inputs corresponding to synaptic stimulus strength in a manner that mimes a biological neuron, are fed via weighted connections, and has a threshold value. Each neuron can receive $x_1, x_2, \ldots, x_m$ inputs

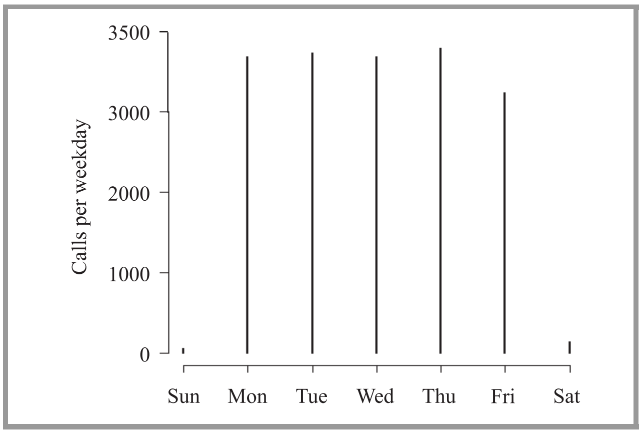

Fig. 5. Distribution of calls per weekday from the first week of February 2012.

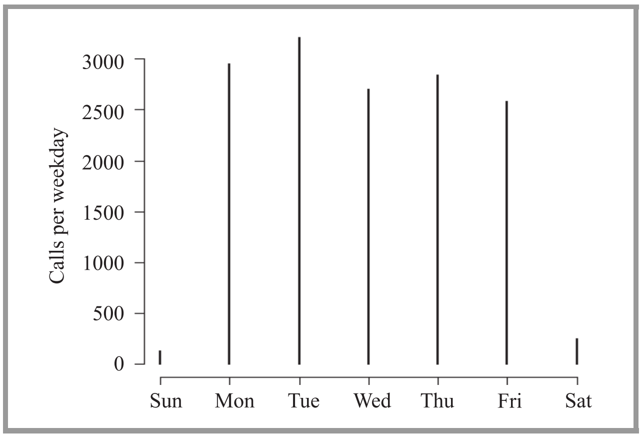

Fig. 6. Distribution of calls per weekday from the first week of August 2012.

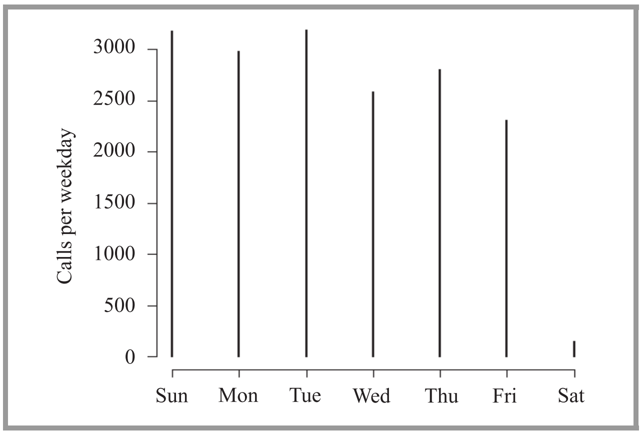

Fig. 7. Abnormal situation, possible attack in the first week of September 2011.

that have $w_1, w_2, \ldots, w_m$ weights. Using the weighted sum of the inputs and the threshold, the activation of a neuron is composed. To produce the output of the neuron, the activation signal is passed through, the neuron acting like the biological neuron. A feed-forward [23] neural network is a collection of neurons that connect in a network that can be represented by a direct graph with nodes (neurons) and edges (the links between them). The feed-forward neural network receives as input values that are associated with the input nodes while the output nodes are associated to output variables. Hidden layers may also appear in the structure of the network [24]. Various types of data can be predicted using neural networks, e.g., future value or trends of a variable (value increase or decrease).

In our model, each day is defined by two parameters: number of calls at hour 10, respectively 11, $D(X,Y)$. Depending on the number of segments chosen, e.g. $nrSeg = 300$, we can calculate, for each time frame, the size of the intervals as being the maximum number of calls for hour 10, respective by 11, divided by the number of segments $(size_1, size_2)$ (Algorithm 3). Each day will be classified as belonging to an interval for X, respectively Y. For, e.g., during a public holiday X and Y of the respective day will equal zero. We count how many days belong to each pair (X,Y) and the classes will be used as an input for training the model (Fig. 8). After training, the calls from hour 10 have been used to test the prediction model, to anticipate the calls at hour 11.

Algorithm 3 Extraction of samples pseudo-code

Store values from database in file, read file, set values X and Y for each day

$size_1 = \frac{max(X)}{nrSeg}$, $size_2 = \frac{max(Y)}{nrSeg}$

Count the days belonging to each interval where $(i-1) \times size_1 \leq X \leq i \times size_1$ and $(j-1) \times size_2 \leq Y \leq j \times size_2$, $i, j \leq nrSeg$

return samples

We generate the particles according to Algorithm 1. The number of components we use (to train the algorithm) is $d = 2$ which defines the dimension of the space. The number of degrees of freedom chosen is $n = 5$. The $N = 1000$ particles will encode the vector μ and the matrix Σ. Based on the parameters of the particles and the samples used

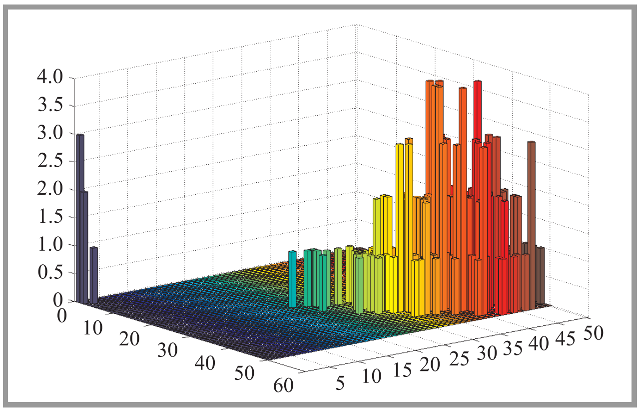

Fig. 8. Before prediction – sample of calls placed in 2012 from 10:00-10:59 AM and 11:00-11:59 AM.

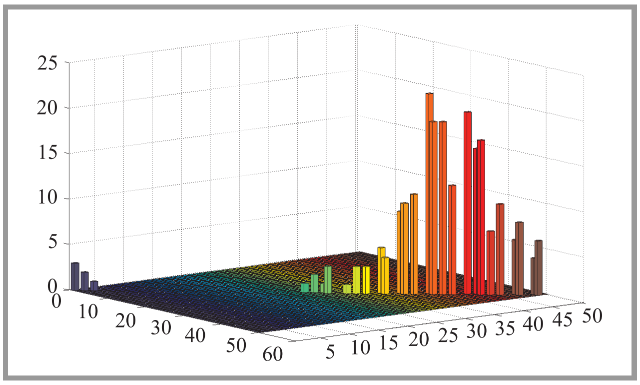

Fig. 9. After prediction – second component calculated based on the weighted likelihood of the particles.

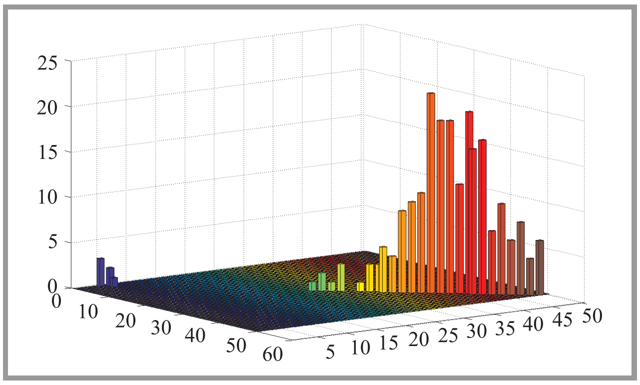

Fig. 10. After prediction – second component calculated based on the particle with maximum likelihood.

as an input, the likelihood of each particle is calculated. This will be considered as the initial population. After the generation, for each particle the values of the parameters μ and Σ will be perturbed according the algorithm. If the likelihood is improved, the new values will replace the old ones and move to the next step. The perturbation is applied for a number of steps, $steps = 100$. The final population of particles will be used for prediction that is done either by considering the particle with maximum likelihood (Fig. 9) or using the weighted likelihood of the of particles (Fig. 10). For comparison, we trained and tested a neural network, using the same training set and testing set

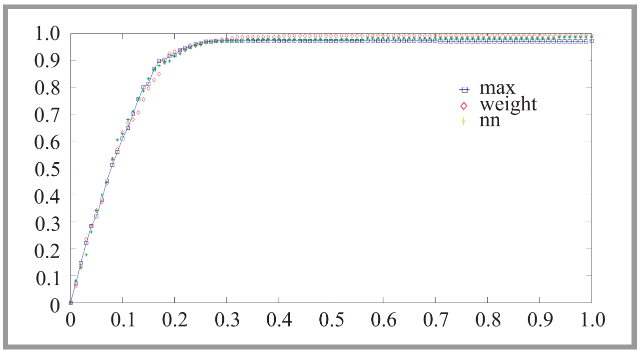

Fig. 11. Number of predictions below a given threshold. Mean absolute percentage deviation.

(the first component) as previously discussed. The neural network has $nr = 10$ neurons, one input, one output and two layers. We obtain the output of the neural network and we compare the prediction given by each method.

In Fig. 11, we can see a comparison of the mean absolute percentage error that was calculated using the expected outcome and the output given by the three different methods. For the $threshold \leq 0.3$, the neural network method has better results while for higher values, the best results are given by the average maximum likelihood. This is what interests us: the predictor that works better for higher errors. The main reasons is that, for example, minus or plus 10 calls will not make a huge difference. Miss-estimating 150 calls can have a negative impact on the system that can end up dropping calls because of lack of resources. For each method, the Mean Absolute Percentage Error (MAPE) is calculated. MAPE is used in statistics to measure the accuracy of a method for constructing fitted time series values. MAPE is defined by the formula: $M = \frac{1}{N}\sum_{i=1}^{N}\frac{P(i)-Z(i)}{P(i)}$, with N the size of the set, $P(i)$ the actual value and $Z(i)$ the forecast value. The smaller the value of MAPE is, the better is the model. In Table 1, based on the calculated error we can see that the weighted likelihood is the best solutions and we are considering it.

Table 1
Mean Absolute Percentage Error

MAPE	Value
Maximum likelihood	0.1359
Weighted likelihood	0.0998
Neural network	0.1159

5. Conclusion and Future Work

In this paper, a prediction model we have built for a real Voice over IP system is presented. The authors have namely discussed a method capable to predict the load of the traffic during a time frame. It was also shown that there are patterns regarding the behavior of the users when placing calls. During public holidays, Sundays and Saturdays, the number of calls is very low while during the weekdays, peaks are likely to appear. This information is helpful when taking decision regarding resource allocations. The prediction model proposed in this work combines different methodologies and will be used as an input for a future load balancing model. We also explored two different methods for predicting the load of the servers and compared the results. The first one takes in consideration the entity with the maximum likelihood while the second one is a weighted likelihood based prediction. The proposed methods were compared with the prediction given by a neural network that was trained and tested with the same data.

Since the VoIP implementations and the technology in itself demand a sustained computational and bandwidth support, the decrease of operation costs at infrastructure level is expected together with the improvement of the VoIP quality. As future work, we will compare our model with Support Vector Machines (SVM) [25], [26], an approach frequently used in machine learning community for classification problems and regression analysis. SVMs are for example used for learning and recognizing patterns for a given input. Moreover, we are investigating the use of a higher number of time frames for the input and testing. We believe that adding this feature could help to improve our results. The behavior of the proposed algorithms in cases like uniformly distributed load, requests focused on one specific set of resources or fast changing profiles that switch between low and high activity periods, will also be analyzed. The idea of building a model per user is also taken into account. Depending on the period of the year, the clients are more or less active. For example, knowing that a school will be closed for summer holidays gives us the idea that less resources will be needed for that specific client. Finding patterns for the normal behavior of the system and defining a confidence interval (in which the traffic can fluctuate normally) is considered. Two different situations will be distinguished: peaks in normal situations (adapt the system to scale and load balance the resources) and peaks in abnormal situations (detection and reaction against attacks). For the current experiments, the samples have been filtered and the abnormal traffic was removed from the training set.

The existing prediction model will also be extended and, based on the outcome, a highly efficient load-balancing algorithm will be developed, allowing to deal with constraints and performance measures not addressed to such an extent and in such an integrative manner before in the literature.

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Fonds National de la Recherche Luxembourg

National Research Fund, Luxembourg, http://www.fnr.lu.

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Paper

Telemaco: A Language Oriented Tool for Graph-based Models Layout Optimization

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Abstract—Progress of ICT is shifting the paradigm of systems organization towards a distributed approach, in which physical deployment of components influences the evaluation of systems properties. This contribution can be considered as a problem of graph layout optimization, well-known in literature where several approaches have been exploited in different application fields with different solving techniques. Then again, complex systems can be only studied by means of different formalisms which codification is the aim of language engineering. Telemaco is a tool that supports a novel approach for the application of graph layout optimizations to heterogeneous models, based on the OsMoSys framework and on the language engineering principles. It can cope with different graph-based formalisms by exploiting either their core graph nature or their different specialized features by means of language hierarchies. In this paper Telemaco is introduced together with its foundations and an example of application to Wireless Sensor Networks (WSN) deployment.

Keywords—graph optimization, modeling languages, wireless sensors networks, WSN deployment.

1. Introduction

Graph layout manipulation is a powerful tool that finds application in many different fields: from computer networks to mechanical modeling, from resources allocation to discrete events systems models, a graph structure appears to be inherent in the inner nature of problems. Optimizing graphs is thus a general solving approach that can exploit either common aspects or specialized issues of models. According to these issues, it is necessary to find an unified way to deal with such different models expressed in different sub-languages conform to graph based ones – language engineering is a discipline that best fits these needs.

In this paper the authors introduce Telemaco, an extensible tool for the optimization of graphs layout under customizable metrics. Telemaco is designed to transparently optimize graph-based models written according to user-defined modeling formalisms by exploiting the advantages of model engineering techniques. The description of models is based on a description framework in which each model is expressed in form of a given formalism. This framework allows formalisms to be designed as extensions of simpler formalisms, actually inheriting all the characteristics of ancestor formalisms, with which they stay fully compatible, applying a sort of inheritance concept with consequent advantages. While this approach allows a systematic development of models, it allows a generalization of some mechanisms, extending the reusability of tools to different kind of models without the need for a software rewriting. As well as a good designed languages hierarchy seamlessly enables Telemaco to correctly operate on new formalisms, its properly defined architecture allows it to be easily extended in order to embed new features in addition to or for a better specialization of the native ones.

Telemaco is part of the OsMoSys framework, both a methodology and a support environment for multiformalism models evaluation and analysis. OsMoSys offers a comprehensive and coherent support for model development and study through its family of languages for the definition of object oriented models and formalisms. In OsMoSys a model is composed by instances of formalism elements, which in turn are described by metaformalisms and that can be inherited from each other both at the formalism level and at the formalism element level, thus allowing of derived formalisms used to describe models that can exploit the advantages of base formalisms. This inheritance process allows Telemaco to define graph optimization primitives at any level of the formalisms hierarchy that can be automatically applied to any model described by any inherited formalism.

Telemaco can be used both in the general context of the OsMoSys Multisolution Framework (the support environment for the OsMoSys Multiformalism Methodology) or as a standalone tool. In this paper it is used standalone in order to focus on its characteristics and its architecture.

Together with modeling oriented languages, Telemaco also benefits of the query oriented languages offered by the OsMoSys framework. Telemaco implements specialized queries oriented to different graph metrics, according to the supported optimization methods and obtained as extension of the OsMoSys query languages.

In order to allow general and performing optimizations, the core metrics are implemented by genetic algorithms. This enables Telemaco to include several different optimality criteria, including heuristics.

Currently Telemaco is limited to layout optimization and only implements general graph optimization techniques. In the future, it will straightforwardly extended as soon as application to real world case studies will be analyzed in

the next steps of this research activity. The application of genetic algorithms in Telemaco is not meant to be a comprehensive view of use of such technique but only wants to find a simple method to cope with different optimization heuristics, as different real world applications ask for.
The original contribution of Telemaco is in the ability of such tool, and of the underlying modeling approach, to cope with different aspects of system modeling and optimization. Since Telemaco essentially manipulates XML based models, it can easily be interfaced with the output of existing third-party tools and, since of its architecture, it can be easily extended to deal with a larger layout oriented set of problems.
After this introduction, a Section 2 gives a general introductions to relevant graph algorithms and genetic algorithms. Subsequently, model engineering is introduced in Section 3 with reference to the OsMoSys approach. Then the architecture of Telemaco is presented in Section 4, followed in Section 5 by a Wireless Sensor Networks based example and conclusions in Section 6.

2. Related Works

2.1. Graphs Optimization

Since graph optimization is a widely analyzed topic in literature, in this section the focus is limited to layout optimization. The problem has been solved with several different approaches and by different perspectives. Exact techniques are generally based on mechanical analogies while also many heuristic techniques proved to be effective.
From the first group Eades introduced the idea of considering springs in place of arcs to allow the optimization by using a mechanical potential function [1], further refined by Kamada and Kawai [2]. Particle physics inspired Fruchterman and Reingold [3], while Kumar and Fowler proposed a tridimensional version of the elastic method [4].
In the second group, Davidson and Harel proposed a heuristic function weighting vertex distribution, arc length and crossing and closeness to borders of the interest area [5]. Kirkpatrick, Gellat and Vecchi exploited simulated annealing [6], while Coleman and Parker [7] combined the advantages of [5] with the speed of [3]. Eloranta and Makinen introduce the use of genetic algorithms [8] and Branke and Bucher use a parallel algorithm based on the elastic approach in association with different criteria [9].

2.2. Tools for Graphs Optimization

Many tools available on the Internet exploit graphs optimization techniques in order to visualize information. A rough classification fitting the purposes of this paper refers to the implemented approach: physics based optimization or graphical optimization. In the first group, that uses algorithms simulating nodes as objects with masses and/or electric charges and consequently arrange the graph according to resulting forces, GraphOpt allows a layered structure to cope with very large graphs, CCVisu allows different energy models to tune the representation including a clustering function based on a LinLog approach, while GRINEdit allows plug-ins. In the second group, that rather implements graph structuring according to a selectable geometry (e.g., tree, circular, symmetric, hierarchical, orthogonal), GoVisual, based on the OGDF framework, allows animating layouts, GDToolkit allows optimization of parameters like length of the edges, the number of crosses and bends along them or the total area of the drawing on different graph types by exploiting object oriented features in the code and applying customizable layout constraints, while Guess applies graph manipulations and optimizations to database exploration and offers a sort of query language expressed in Gython. A list of references for these and other tools is available at http://www.dmoz.org.

2.3. Genetic Algorithms

Genetic algorithms are a heuristic method for search and optimization, inspired to the general principles of natural selection in biological evolution. The core concept is that an optimization process is designed as the creation of generations of candidate solutions, on which a fitness function is evaluated in order to detect the best candidates that are then combined by exchanging some of their characteristics to obtain the next generation. Genetic algorithms have been proved to best fit general situations where other methods can not use specialized knowledge about the heuristic function to optimize. An introduction to the topic can be found in [10]–[12].

3. A Language Oriented Approach

Model-driven Engineering (or Model Engineering tout court, ME) is a well known approach to the design and development of complex systems [13], that can be easily seen as a generalization of the widespread software engineering approach of the Object Managements Group Model Driven Architecture [14]. ME allows to separate conceptual aspects of design from implementation aspects, by exploiting the massive use of models and transformations between models [15]–[22]. These formal models capture different aspects of the design, including the model of the system architecture. Automatic transformations between different models, defined on the base of the underlying formalisms, allow designers to separately focus on parts of the problem and automatically adapt the results to the implementation. The goal of such approach is to obtain correctness-by-construction rather than construction-by-correction that is typical of several traditional and empirical system and software engineering approaches. Homogeneously defined models and metamodels form a Technological Space (TS) [23].
In order to support this philosophy of design, the infrastructure for models and manipulation is founded onto a coherent definition for description languages. A proper organization for languages consists of different levels of

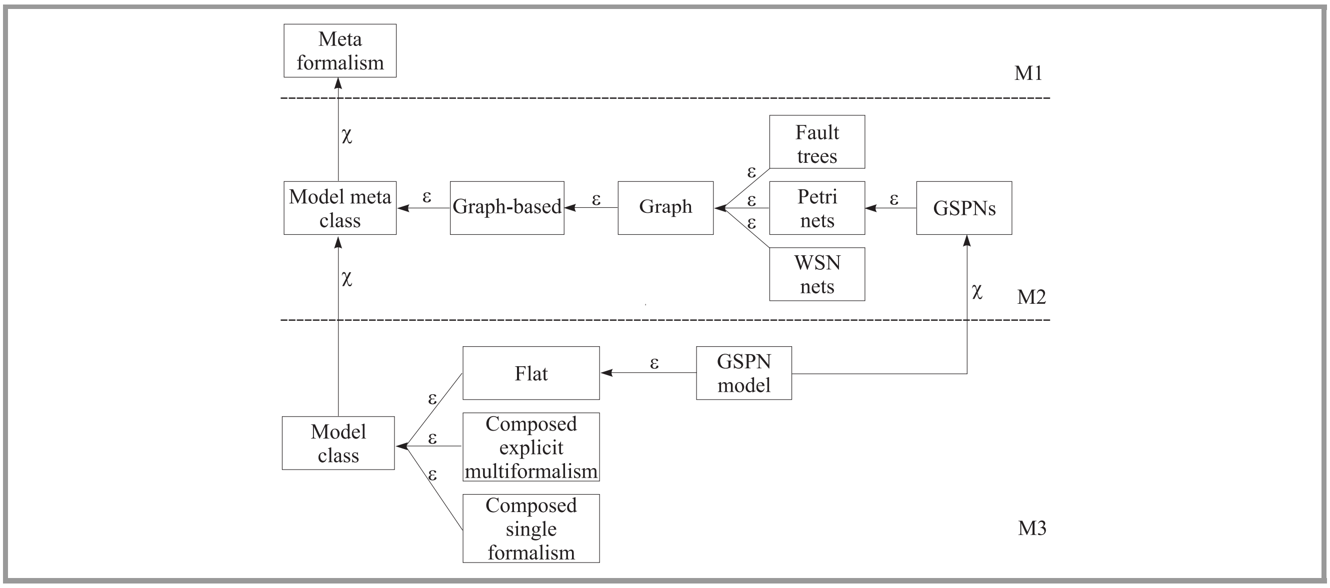

Fig. 1. Model Driven Engineering languages and models.

descriptions, that is the availability of a layer of languages aimed to describe models and a layer of languages aimed to describe such languages as is depicted in Fig. 1. A shared terminology defines as metamodels the languages aimed to models and as metametamodels the languages aimed to languages. The process of creating a models is thus considered as the instantiation of a number of elements on these layers, that generally define a tree structure through this stack. The motivation for this multilayered descriptions is twofold. The possibility of extending the number of available metamodels, by implementing new metamodels through existing metametamodels, and the possibility of establishing relations between models coherent with different metamodels by using existing relations between metametamodels. It is worth noting that the general representation of such models is usually a graph or can be mapped to a graph.

Other approaches exist that are founded on similar premises, but starting from a different point of view, such as the OsMoSys project, on which this paper focuses, and the SIMTHESys project [24]–[26]. The OsMoSys Multisolution Framework (OMF) [27] implements the OsMoSys Modeling Methodology (OMM) [28] that aims to multiformalism modeling, a modeling approach that allows different parts of a model (submodels) to be modelled with different metamodels (formalisms) in order to couple the description of each submodel with the best suited formalism. The OMM aims to build multiformalism models in order to evaluate some of their characteristics (i.e., performances, timeliness, dependability, availability) and supports the modeling process with proper semantic relations between submodels with different nature.

The OMF offers a number of model stacks (language families) to define not only the description of complex models, but also other relevant aspects as the queries with which the user can define the target of an analysis on the model or the kind of results a certain formalism can produce. Anyway, in this paper we will refer to the language family devoted to describe models. Models (model classes), metamodels (formalisms) and metametamodels (metaformalisms) are organized in order to allow the definition of model classes as compositions of submodels designed in different formalisms that can be related with each other by means of their description by a common metaformalism. New formalisms can be written from scratch by implementing their description in the most suitable metaformalism or can be obtained by extending existing formalisms with new elements or refining some elements, thus inheriting all the characteristics of the father formalism. This inheritance mechanism at the formalism level allows the modeler to exploit on new formalisms advantages designed for existing formalisms and automatically enables interactions between submodels not explicitly designed to interact with each others. For a deeper insight into inheritance in OsMoSys the reader can refer to [28].

The graph-based model description language family of OsMoSys can be considered in the ME perspective as a TS. This allows a further formalization of the process of generating any kind of graph-based (derived) model as in Fig. 1. In the figure, ε defines a is-a relation and χ defines a conform-to relation. On the model layer (M3) an example GSPN model, conform to the GSPNs formalism, is derived by a more abstract flat model (without submodels). On the formalism layer (M2), the GSPNs formalism is showed to be derived from the Petri Nets formalism, rather than from Fault Trees, another formal language, or WSN Nets, a description for Wireless Sensor Networks models. That in turn is derived from a simple Graph formalism, and next is derived by an abstract Graph-based formalism, that is a model metaclass (synonymous for formalism in OsMoSys). A model class conforms to a model metaclass, that in turn is conform to a metaformalism (M1).

Telemaco is designed to transform models in the OsMoSys graph-based TS and to be integrated into the OMF architecture as an OsMoSys adapter/solver couple [27]. Telemaco transforms a model written in a certain formalism into another model of the same formalism but with a different layout, according to a proper query formulated in the OsMoSys model query language. The tool is currently designed to operate on the graph-based formalism, and since of the OsMoSys languages inheritance properties, is capable of operating on all models conform to a formalism derived from it.
Since all languages of the OMF are implemented in XML, Telemaco can easily operate on the output of third-party tools, such as J-Sim, a a component-based, compositional simulation environment that has already demonstrated its effectiveness in modeling Wireless Sensor Networks [29].

4. Architecture of Telemaco

4.1. Layout Optimization

The problem of graph layout optimization, as seen, has been widely examined in literature. Two contributes affect the process: a functional aspect, connected to the nature of data represented by the graph, and an aesthetic aspect, common to all graphs. In order to detect unifying features in the process of graph optimization some layout and graph inputs and some heuristics are described. Inputs are:

- the dimensions of the visualization area; according to this parameter it is possible to scale properly all the computations;
- the initial positions for nodes;
- the arcs between nodes.

Possible metrics are:

- distance between nodes,
- uniformity: balances the density of the nodes in the available area,
- arcs intersection: avoids intersection of arcs if possible,
- average (or maximum) arcs length,
- symmetry.

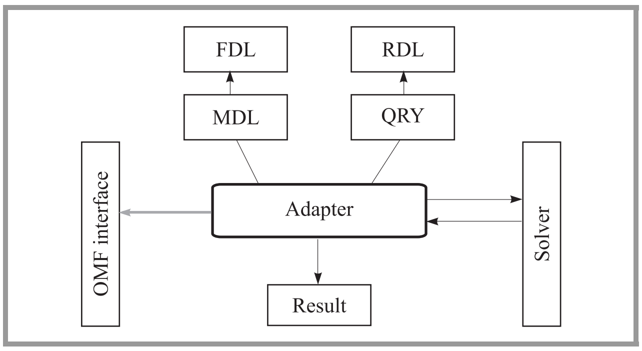

Fig. 2. An input/output view of Telemaco

The general structure of Telemaco is shown in Fig. 2. The solver (Telemaco.core) implements the optimization strategies while the adapter (Telemaco.ext) is in charge of accessing data from the model and the formalism (MDL, FDL) and from the query (QRY, RDL), and of producing results.

4.2. Solution Engine

The solution engine of Telemaco is based on genetic algorithms rather than physically derived algorithms because of the better potential of the first solution in terms of flexibility and extensibility. A genetic algorithm operates repeating a cycle of three phases until the desired number of generations has been reached, starting from an initial population automatically generated according to the initial parameters. The three phases are:

- **Selection**. During which the fitness function is evaluated on every element of the population to select which ones will contribute to the new generation. In order not to take always the locally optimal solutions, besides taking the best ones Telemaco implements two other strategies from the literature, namely the roulette [8] and the lottery strategies, the best strategies in a random subset;
- **Crossover**. During which it can happen that parts of the binary description of the coordinates of the nodes of two selected elements are swapped. Telemaco supports single and double swappings;
- **Mutation**. During which it can happen that some of the information about the coordinates of the nodes in a model are randomly changed. Telemaco supports four different mutation techniques.

Fitness evaluation is evaluated as the weighed sum of single fitness metrics that consider arcs intersections, node distances, density, angles between arcs, arcs lengths and arcs uniformity.

4.3. Internal Architecture

According to the OMF, Telemaco is composed by two components: an adapter and a solver (Fig. 3).

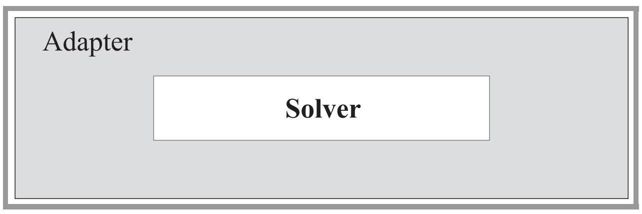

Fig. 3. Telemaco structure according to OsMoSys

In addition, Telemaco also offers a GUI (Telemaco.GUI) for the visualization of results when used standalone. The architecture of the tool is described in the UML class

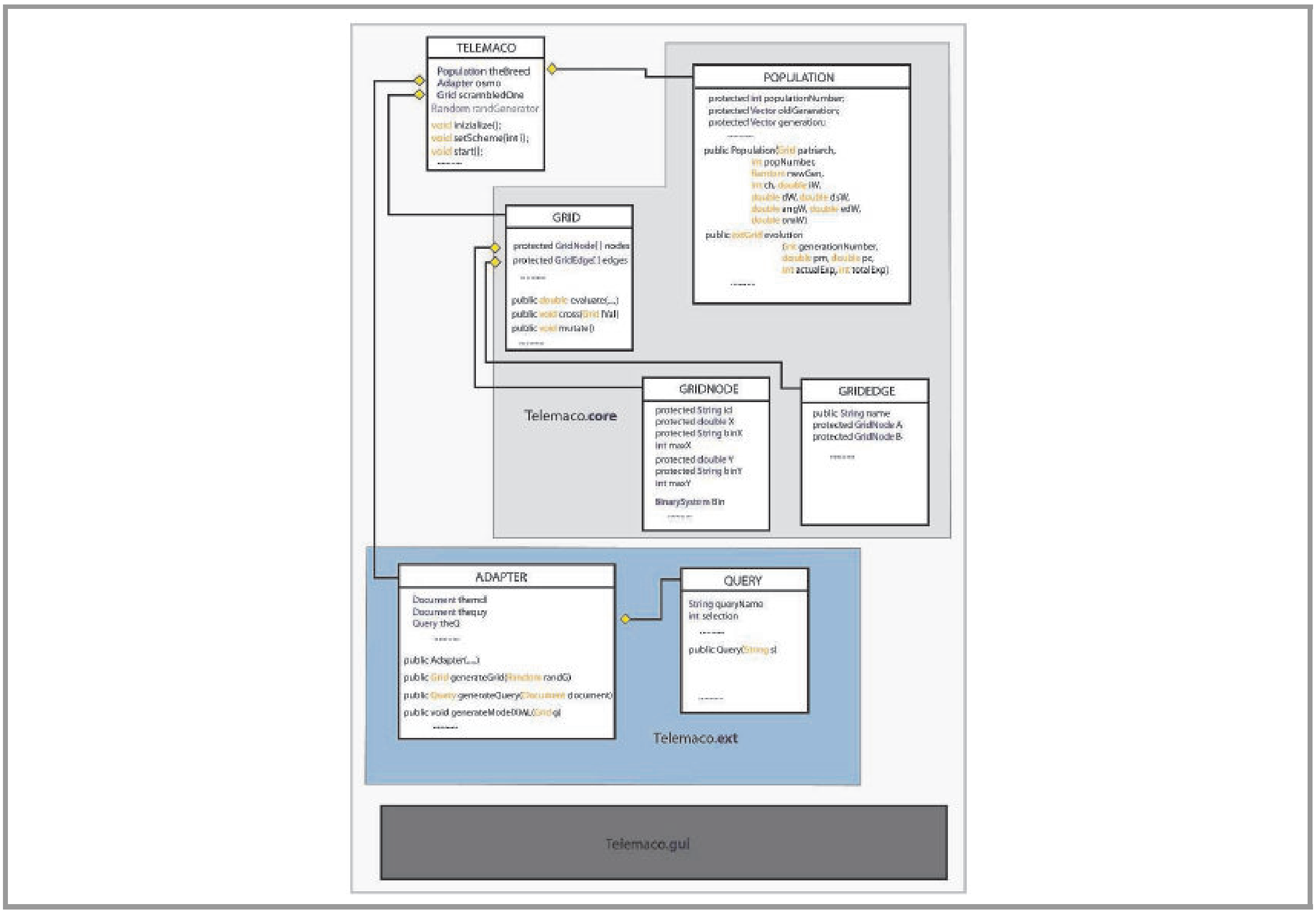

Fig. 4. A class diagram view.

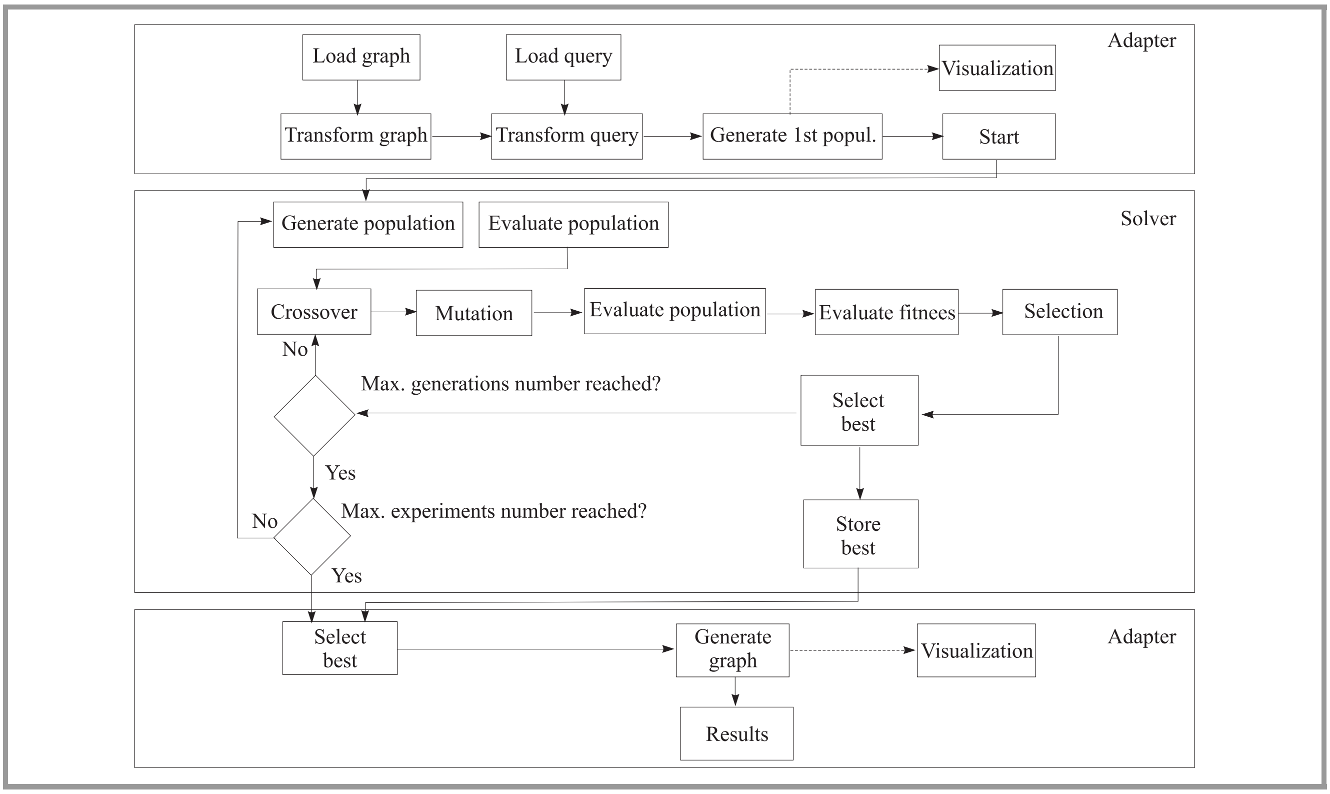

Fig. 5. Execution steps of Telemaco.

diagram in Fig. 4. Figure 5 presents a sketch of the operations performed by the tool.

4.4. An Application

In order to show how the tool works, we present here an example of the application of two queries, namely distinter and intersection, to the optimization of a graph representing a GSPN. The intersection query minimizes the intersection between arcs while the distinter query also aims to maximize distances between nodes. The initial situation is showed in Fig. 6 and both it and the query are described as follows:

```
<?xml version="1.0" encoding="UTF-8"
 standalone="yes"?>
<mdl type="FLAT">
  <GSPN fdl="GSPN.xml" name="Net2"
    Area="511">
    <Place name="P1" Tokens="2" X="10"
        Y="100"/>
    <Place name="P2" Tokens="0" X="20"
        Y="20"/>
    <Place name="P3" Tokens="0" X="300"
        Y="20"/>
<!--  TimedTransitions  -->
    <TimedTransition name="T1" X="30"
        Y="50"
     Rate="1.000000e+00" ServerType="0"/>
    <TimedTransition name="T2" X="80"
        Y="70"
     Rate="1.000000e+00" ServerType="0"/>
    <ImmediateTransition name="t3" X="25"
        Y="120"
     Weight="1.000000e+00" ServerType="1"
     Priority="2"/>
    <ImmediateTransition name="t4" X="30"
        Y="400"
     Weight="1.000000e+00" ServerType="1"
     Priority="2"/>
<!--  Arcs Section  -->
    <Arc name="Arc1" Weight="1" from="P1"
        to="T1"/>
    <Arc name="Arc2" Weight="1" from="T1"
        to="P2"/>
    <Arc name="Arc3" Weight="1" from="T2"
        to="P1"/>
    <Arc name="Arc4" Weight="1" from="P3"
        to="T1"/>
    <Arc name="Arc5" Weight="1" from="P2"
        to="T1"/>
    <Arc name="Arc6" Weight="1" from="P2"
        to="t3"/>
    <Arc name="Arc7" Weight="1" from="P1"
        to="t4"/>
  </GSPN>
</mdl>
```

```
<?xml version="1.0" encoding="UTF-8"
 standalone="no"?>
<mql rdlref="GSPN.rdl" mdlref="Nets.xml">
  <result name="distinter"/>
  <result name="intersection" />
</mql>
```

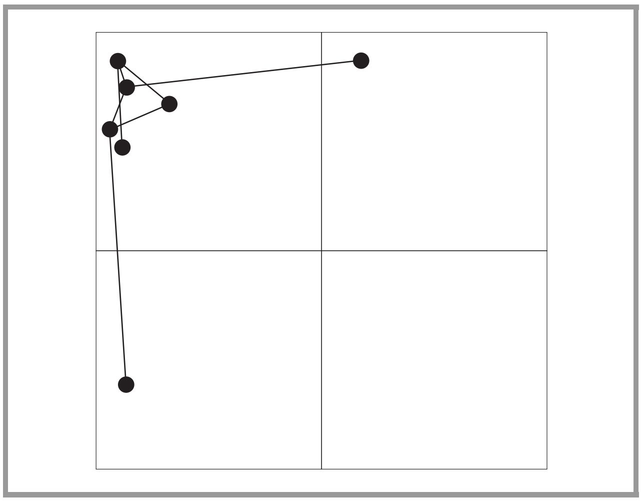

Fig. 6. Initial unoptimized graph.

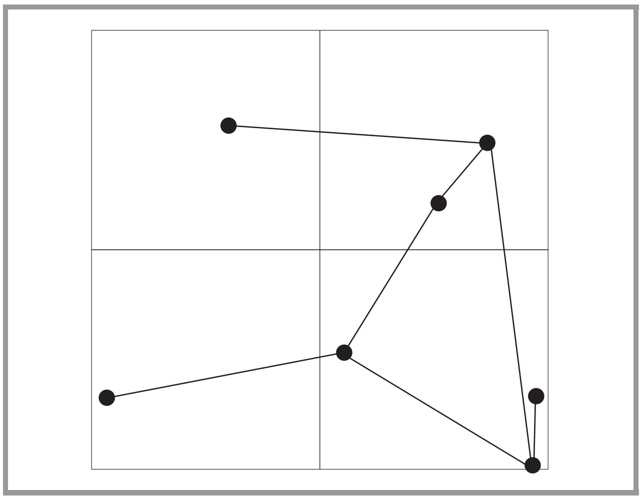

Fig. 7. Graph optimized according to distinter metric.

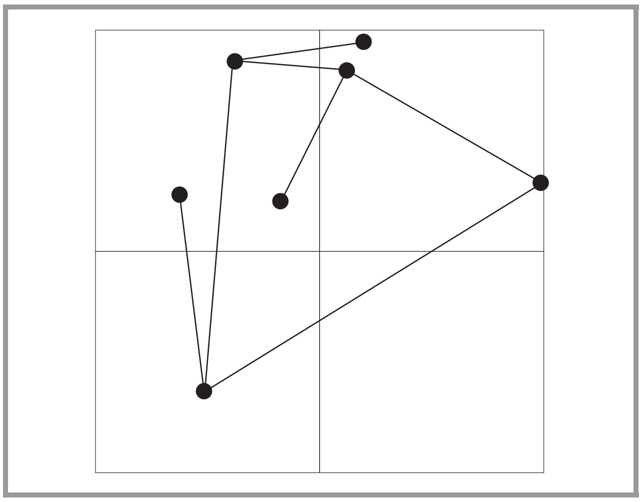

Fig. 8. Graph optimized according to intersection metric.

Figures 7 and 8 show the outputs of the two queries respectively after the execution of the genetic algorithm over few executions on several generations.

5. An Example – WSN Deployment

To demonstrate the use of Telemaco, a Wireless Sensor Networks (WSN) coverage example from the literature has been chosen. WSN are used to easily deploy sensors in areas of every dimension to collect data about the environment. In a WSN, sensors form a wireless ad-hoc network in order to vehiculate, with the lowest possible power consumption for transmissions, data towards some specialized nodes, known as High Energy Communication Nodes (HECN) and characterized by higher performances and connection to a communication channel towards the user. Optimal deployment of sensors is a well known problem [30] that depends on applications, and can be considered as the foundation of optimal dynamic relocation in Mobile Sensors Networks, see [31]–[34] for an introduction.

Our example is taken from [35], in which Jourdan and de Weck tackle a coverage problem for a military WSN with three different examples. Our example is a variation of the last of them, in which the WSN must cover at best an area by using sensors positioned so that no discontinuity can exist between their coverage. The area must be completely covered and at least one sensor must be connected to the HECN, which in turn has sensing capabilities. The authors design a multi-objective genetic algorithm approach specialized for this kind of applications. The presented example just aims to show the flexibility of Telemaco in facing problems for which it is not intentionally designed, by extending the tool for this purpose and comparing in general its results to the optimal solution obtained in the reference paper.

In order to capture the fact that a WSN is a specialized graph with two different node types (HECN nodes and sensor nodes) a proper formalism can be derived from the base Graph formalism. The new node types have a characteristic attribute, that is the coverage radius. Coverage radius will be described as an integer without loss of generality, and will be set to a default of 0 (non-working node). Each single sensor can have a different radius, in order to model heterogeneity of sensors or different sensing capabilities. The new WSN formalism will have three element types, two of which have been introduced and the third of which is the arc element, which actually represents the (fixed) connection that nodes will use to communicate. The two nodes will be specialized from Graph.Node (that is, the Node element of the Graph formalism) while the Arc will be derived from Graph.Arc.

The complete description of the formalism is as follows:

```
<?xml version="1.0" encoding="UTF-8"
 standalone="no"?>
<formalism parent="GRAPH" name="WSN"
 type="formalism">
  <elementType parent="" name="WSN">
    <elementType parent="GRAPH.Node"
        name="HECN">
      <propertyType name="Coverage"
            type="integer" default="0"/>
    </elementType>
    <elementType parent="GRAPH.Node"
        name="Sensor">
      <propertyType name="Coverage"
            type="integer" default="0"/>
    </elementType>
    <elementType parent="GRAPH.Arc"
        name="Arc">
     </elementType>
  </elementType>
</formalism>
```

The example is based on a WSN composed by a single HECN, with a bigger coverage radius, and five sensors, all with the same coverage radius. The arcs configurations is given and includes a loop containing the HECN in order to simulate redundant routing for better availability. The description of the WSN is as below:

```
<?xml version="1.0" encoding="UTF-8"
 standalone="yes"?>
<mdl type="FLAT">
  <WSN fdl="WSN.xml" name="WSN NET"
    Area="511" type="WSN">
    <Sensor name="P1" Coverage="100"
        X="10" Y="100"/>
    <Sensor name="P2" Coverage="100"
        X="20" Y="20"/>
    <Sensor name="P3" Coverage="100"
        X="300" Y="20"/>
    <Sensor name="P4" Coverage="100"
        X="500" Y="300"/>
    <Sensor name="P5" Coverage="100"
        X="350" Y="350"/>
    <!--  Energy Communication Node  -->
    <HECN name="T1" X="30" Y="50"
        Coverage="200"/>
    <!--  Arcs Section  -->
    <!-- Arcs -->
    <Arc name="Arc1" from="P1" to="T1"/>
    <Arc name="Arc2" from="P3" to="P2"/>
    <Arc name="Arc4" from="P3" to="T1"/>
    <Arc name="Arc5" from="P2" to="P1"/>
    <Arc name="Arc6" from="P3" to="P5"/>
    <Arc name="Arc7" from="T1" to="P4"/>
  </WSN>
</mdl>
```

Since the problem is about optimal coverage, the native metrics of Telemaco do not fit, being designed for the optimal visualization of graphs rather than to ensure the continuity of the covered area of sensors. The effects of native metrics is shown in Fig. 9 where the application of an intersection metric is presented. It is evident that the result is definitely inadequate in confront of one example layout (manually generated) depicted in Fig. 10.

In order to obtain a good result, Telemaco has been extended with two additional metrics. The first focuses on distributing the nodes at a distance that is as close as possible to the coverage radius. The second keeps the nodes

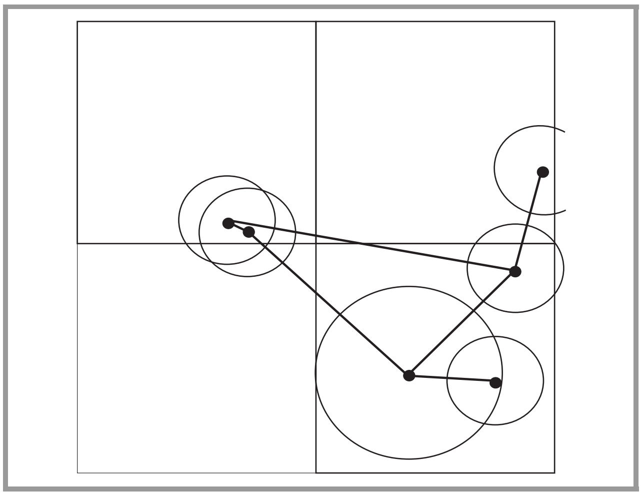

Fig. 9. WSN wrong optimization according to intersection metric.

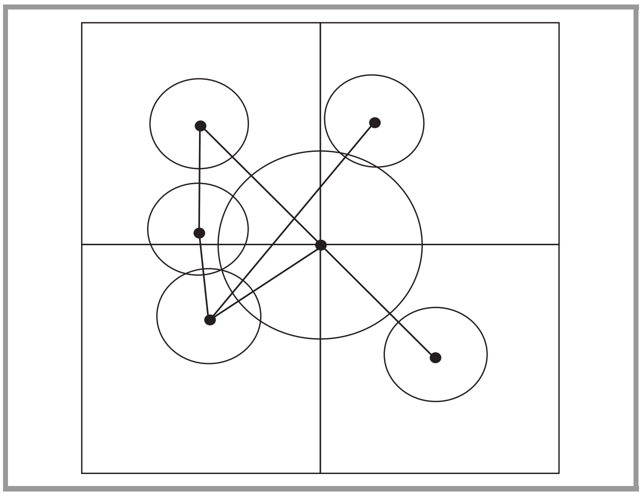

Fig. 10. WSN best layout.

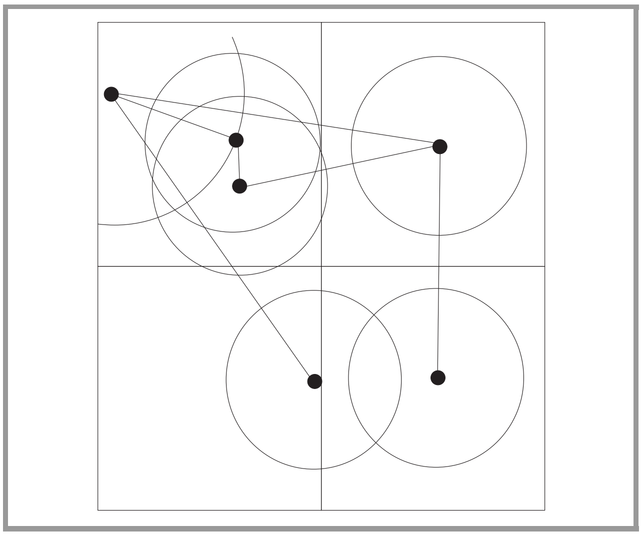

Fig. 11. Application of sensor_range metric to WSN.

coverage inside the overall area that represents the environment in which the WSN operates. Notice that while the

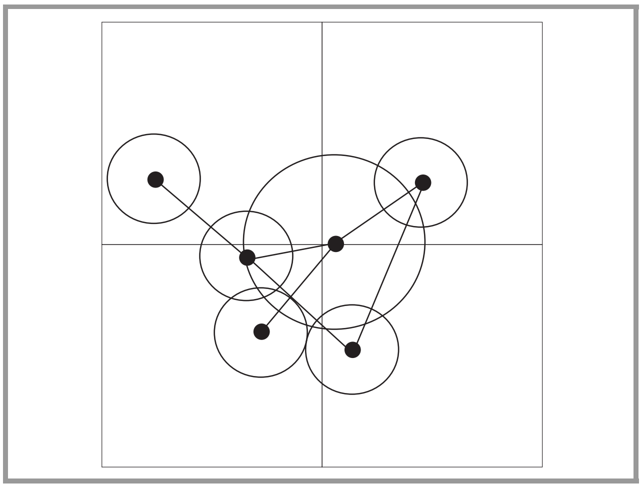

Fig. 12. Application of in_area metric to WSN.

standard metrics automatically still operate on the new formalism, the new ones exploit the new characteristics of the formalism. In consideration of the fact that the length of an arc is in some way a relative measure of the power needed to transmit data between two connected nodes, both these new metrics have been combined with the intersect metric in queries, since arcs crossings generally imply longer paths. The queries are as follows:

```
<?xml version="1.0" encoding="UTF-8"
 standalone="no"?>
<mql rdlref="WSN.rdl"
mdlref="WSN_net.xml">
<result name="sensor_range"/>
<result name="in_area" />
</mql>
```

Results of the application of the new metrics are shown in Fig. 11 and in Fig. 12.

6. Conclusions and Future Works

In this paper we defined a framework and described a tool aimed to solve the problem of graph layout optimization in its most general form by means of Model-driven Engineering, an emerging discipline of software engineering. By means of a formal languages definition every graph based model can be easily inherited from a graph on which layout optimization can be seamlessly performed.

This modeling technique is well defined in the OsMoSys framework, that greatly exalts the modeling features of the approach (by means of OMM) and the solution facilities (by means of OMF). Due to its generality, this approach takes the best from the use of genetic algorithms in order to generate most fit solutions, i.e., best graph layouts.

Several extensions are possible from this work both on framework and tools, extending fitness functions and improving the power of the genetic engine, and on applications, studying other application fields and deepen Wireless Sensor Network example.

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Paper

Recent Developments in Mobile Cloud Scheduling: State-of-the-Art, Challenges and Perspectives

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Abstract—Cloud computing became recently one of the most popular multi-layer distributed computational and data processing environments with various types of services, distributed data storages and resources. With rapid development of mobile technologies, computational clouds have been transformed into the systems with dynamically changing topology and flexible infrastructure through integration with the mobile devices and mobile users as the whole system nodes and actors. The aim of this paper is to provide a comprehensive study and critical comparative analysis of the recent developments in the Mobile Clouds with a new energy optimization criterion scheduling.

Keywords—energy awareness, mobile cloud computing, scheduling.

1. Introduction

Computational Cloud (CC) can be considered as a computational environment that contains many servers distributed at geographical locations [1]. The complex structure can be modeled as a hierarchical architecture composed of three main layers, namely: Application as a Service (AaaS), Platform as a Service (PaaS) and Infrastructure as a Service (IaaS) [1].

In today's cloud computing the role of the mobile devices is not limited just to the client devices for the cloud users. The mobile platforms can be also considered as data storage and access nodes of the cloud system. In mobile cloud environment, the remote server access with mobile devices is mandatory [2]. All conventional and recently developed features of traditional CC such as privacy, security, energy awareness, resource reliability, system reliance, etc., must be analyzed differently in case of Mobile Cloud (MC) computing systems [3]. A fair optimization of the energy consumption, mainly because of short battery life, seems to be a crucial for the design of modern mobile cloud schedulers. The main aim of this paper is to present comparative analysis of CC and MC systems with a focus on energy awareness in resource and cloud application management and scheduling.

The paper is divided into five sections. Section 2 discusses the generic model of MC environment, including the detailed characteristics of the system and different types of MC. Section 3 discusses wide range of scheduling problems in MC and addresses the most important scheduling criteria such as energy consumption. There is short comparative analysis of main CC and MC features. This chapter contains also some recently developed methodologies for complexity reduction of the processes implemented and executed at the mobile devices. Those methodologies are named the offloading methods and can be applied for users' applications computational time reduction, modifications of the authentication and authorization procedures. Dynamic Voltage and Frequency Scaling (DVFS) method implemented in IaaS cloud layer is also presented in this charter as a basic hardware methodology for Mobile Cloud energy consumption optimization. Section 4 presents the selected energy-efficient algorithms in MC systems, namely Scavenger [4], MobSched [5] and Scheduler of Mobile Cells [6].

2. Mobile Cloud Environment Generic Model

Similarly to traditional Cloud system Mobile Cloud (MC) environment can be modeled as multi-layer system. However in this model each cloud layer is in fact a combination of classic infrastructure with mobile devices. Cloud platforms are therefore extended by the mobile services and mobile client applications. Mobile databases must be integrated into cloud data centers, while mobile applications are usually sent to the cloud system for execution in order to offload the mobile system nodes. This last feature is very important especially due to mobile devices limited data storage and computational performance. Mobile cloud devices (e.g. smartphones, tablets, etc.) are used by mobile users who often change their location, leave and/or join the systems dynamically and may have limited access

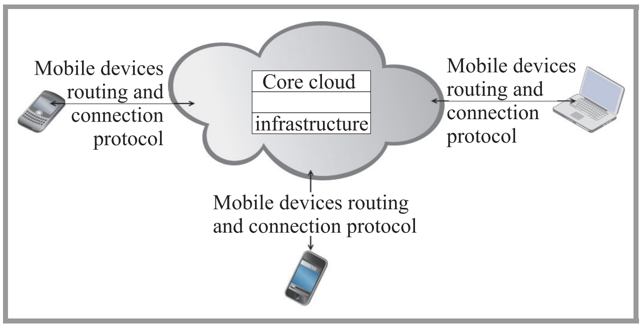

Fig. 1. Mobile Cloud system generic model.

to cloud data and services. Simplified Mobile Cloud system is shown in Fig. 1.
Brief characteristics of the system main components will be provided in the following subsections.

2.1. Mobile Cloud Core

The core of the Mobile Cloud is traditional cloud infrastructure, which is usually modeled as a 3-layer system. It consists of System as a Service (SaaS), Platform as a Service (PaaS) and Infrastructure as a Service (IaaS) layers [2].

Software-as-a-Service layer can be defined as a computational environment providing online services for the Web browsers. SaaS structure delivers many different applications that can personalize cloud system according to individual user's requirements. Gmail and Amazon are examples of the SaaS [7]. SaaS layer provides specific software packages for the cloud users.

Platform-as-a-Service is development environment for cloud applications. It is independent from the hardware, so the applications can be more flexible. This platform makes data and applications portable, so teams of engineers that are in different geographical locations can concurrently work using PaaS. They can see all the changes the other did and cooperate without any distance borders. This technology is also very useful for big corporations with branch offices - employees still can work on the same project, it is easy for them to share the results of their work and to do backups. Google App Engine is the example of PaaS [7].

Infrastructure-as-a-Service is the last main layer of cloud service, which is also the core of the whole cloud structure. IaaS is based on physical networks and hosts data and databases and running applications. IaaS is a physical layer of the cloud and is composed of the physical computational unit-connecting cluster and devices. The main IaaS of the resource management problem is to provide users access to virtual machines at the same time. Another issue is effective task scheduling, in order to provide clients and tasks distribution between virtual machines to achieve the best system performance [3].

2.2. Mobile Cloud Nodes and Users

Mobile devices can be considered as integral part of the cloud network. It means that some numeric operations can be performed on those devices. The main limitation of cloud mobile nodes is limited computational capacity and data storage, as well as relatively short battery life. It means that only lightweight numerical tasks and very simple databases can be implemented there. Different mobile devices can be included in or dropped from the cloud physical layer very often. MC users' mobility management is supported by geolocation systems and communication technologies such as GSM, Wi-Fi, GPS. All such methods may increase energy consumption, which results in battery life decrease. Recently two peer based techniques, namely "Escort" [8] and "Virtual Compass" [9] have been proposed as lightweight alternative for such geolocation problems.
Three cloud's layers short comparison of the CC and MC systems main features are presented in Tables 1 and 2.

Table 1
Two cloud types system layers short comparison

System's layer	Computational Cloud (CC)	Mobile Cloud (MC)
SaaS	Software applications are not located on the user's devices	Applications are used by users through the vendor's website
PaaS	Static	Dynamic
IaaS		

Table 2
Two cloud types system layers short comparison

Technology/ features	CC	MC
Security	Might be on an high level	High level is difficult to achieve because of many external users
Energy consumption	High, as computational unit is powered from AC mains	Low but mobile units are supplied from internal battery, which limits their operating time
Scheduling	Difficult to manage	Very difficult to manage because of Mobile Cloud structure

Information presented in Table 2 show that the whole resource management and scheduling issues are much more complex in MC than in classic CC.

2.3. Characteristics of the Mobile Cloud

2.3.1. Security

CC achieves high security level if data is stored in the Private Cloud. MC has dynamic structure where users can join in or drop off the cloud very often. This feature of MC can cause several issues of data security even if information is stored in the mobile Private Cloud. On the other hand it is obvious that as long as memory for data storage in mobiles is not large, many users store their personal data in the cloud. The main security issues include the following:

– privileged users access,
– data location,
– data segregation,
– recovery,
– investigative support,
– long-term viability.

One of the most interesting recent solutions for cloud systems security is model based on “weblets” technologies [10]. In this model, security framework that includes cloud manager, cloud node manager and cloud fabric interface consists of the following steps:

– check if framework is safe for the system,
– authenticate between weblets,
– perform authorization,
– establish connection and perform verification.

The safe weblets container is implemented on the mobile device and in the cloud. The authentication and authorization sessions supports access control with some specified privileges. MobiCloud is one of examples of weblets based system [10]. The architecture of MobiCloud security system is based on Mobile Adhoc Network (MANET) framework. Its main features are:

- MobiCloud is used as a link in authentication process, while cryptography is used for data access and its detachment;
- protection of the information in the system is based on VirtualTrust and Provisioning Domains that contain nodes responsible for message’s security system in MobiCloud;
- MobiCloud is used to define the risk level. The risk management service finds malicious nodes that may cause some problems [10].

2.3.2. Privacy

The privacy issue in MC relates the type of cloud (hybrid, private, public, etc.) and with the type of applications that are executed on mobile devices. When the device connects to the MC, applications and user get access to the data stored in the cloud. It is important to provide way to protect data and at the same time let user to decide about privacy level. Some examples of such methods are discussed below.

In Mobile Cloud network architecture and design must be flexible and adaptive to all changes in its topology. The users often join or leave the network so some nodes are activated or deactivated by the system. Network random users are never fully trustworthy. The authentication process must be therefore adapted to the dynamic structure of the cloud. Even when the authentications requests are very frequent because of the high traffic in the cloud, the securing process needs to be completed. It can charge the system significantly, so it is important to avoid asymmetric key operations and sending large messages. The solution is the PKASSO protocol that is based on Public Key Infrastructure [11]. PKASSO offloads the mobile devices in executing complex operations. However, this method requires fast Internet connection.

Another solution for privacy in mobile cloud nodes is Onion Routing Model [12]. It provides messages multiple encryption, as every message transport chain element adds its own level of encryption. Nodes task is to remove the last layer of encoding, read routing instructions, and send data to the next router with encryption. The intermediary nodes cannot see neither sender nor recipient of the message. This solution is a type of anonymous protocol, which is sometimes unreliable as intermediary node topology can change dynamically. Anonymity is the core issue of the problem. On the other hand, it should be always considered that the higher level of privacy is implemented in the public mobile cloud, the higher are transmission and computation costs [12].

As regards the privacy issues it might be important to let users set privacy level they want to use in the public Mobile Cloud. As it was mentioned, some of the applications may use information about users of the cloud. In [13] authors discussed Privacy Rights Management for Mobile Applications system allowing users to decide what information they want to share and what privacy level they want to keep on their device.

2.3.3. Energy Utilization

Energy consumption is a crucial issue for Mobile Cloud systems. Devices connected to the cloud are battery-powered, therefore it is important to use energy saving techniques. To optimize energy consumption in the cloud, information of node performance and tasks load is mandatory.

There are two main methodologies energy consumption measurement: hardware-based and software-based. The first category includes:

- Power usage map tool, that allows programmers to modify their source code to do it more energy efficient. However it is possible only with open source operating systems, because kernel modification are required. Consumption of energy is measured by instrumentation. Optimization is manual, based on energy measurement in reference to source code [14];
- Power usage framework, where power consumption is measured by sensing power supply current via shunt resistor. To use it the test software script must be first downloaded on the mobile device from the server. The framework sends power consumption results back to application run on server, and optimization can be done automatically. After the test the results are sent back [15].

It seems that energy usage related to the tasks is not convenient and in many cases not possible as it requires hardware that usually is not available within the nodes. The software techniques based on tasks time execution let avoid this limitation. An example of the software-based methodology is PowerSpy [16] implemented for Microsoft Windows operating systems.
Some examples of energy-aware "green" mobile projects are presented in the Table 3.

Table 3
Green mobile projects examples

Name	Region	Target
Earth	Europe	Mobile network
Green IT	Japan	Commercial IT
Green Grid	Global	Data centers
OPERA-Net	Europe	Mobile networks
Green500	USA	Supercomputer

2.3.4. Cost Specification

As it was mentioned, the main issue in Mobile Clouds is the dynamics of the system topology. At any moment the architecture of the cloud can change, some devices can join or drop from the cloud. In addition the number of servers or type of executed task can change at any moment. That is why it is so difficult to optimize execution time or to set the schedulers.
The MC costs can be divided into two main groups: Total Cost of Ownership (TCO) and utilization cost [17]. There are four factors influencing cost analyze time, energy, execution time and storage time.
TCO is a cost of keeping and managing the IT infrastructure in the cloud, i.e. energy, software, servers, network support, cooling system, etc. Utilization costs are the resources used by application or user. It is important to be aware that because of the cloud architecture the number of resources (devices, servers) is dynamically changed.
Additional (surrogate) servers, used in cloud for offloading tasks from mobile devices increase system cost. The cost models, created for cloud cost optimization use resource monitoring and profiling; methods to balance the cost vs. the benefit from offloading task to surrogate servers. The cost models try to predict energy consumption, video quality, and performance are predicted for every surrogate server.

2.3.5. Main Mobile Cloud Features

As it was discussed in previous subsections MC has several problematic features. To address them the implementation of specific algorithms is required, like presented in Table 4.

Table 4
Mobile Cloud implemented algorithms examples

Issue	Problem	Algorithm
Energy consumption	Video coding and data transmission	Adapts to the video and underlying network traffic to minimize the total energy consumption [18]
Video quality	The Mean Square Error between the original video frames and the decoded video frames [19]	Adopted a power rate-distortion model to capture the trade-off among the encoding rate, energy consumption of the encoder, and the video quality [20]
Power saving system	Consumption of energy during computing	Power-saving access point (PSAP) used for solar/battery powered applications [21]
Context-aware system	Different mobile devices (battery capacity, type of display), different users	Spans the application layer, Middleware layer and network layer [22]

2.4. Main Types of MC Systems

Similarly to CC systems, there are several types of the Mobile Clouds that can be based on the management and include the access policies to the cloud services and data [2].

Private Mobile Cloud – is a type reserved for the organizations. It is possible to completely control the privacy and security of data stored in such type of a cloud. Organizations can decide who and when can access the cloud. The high level of the security increases costs for the users.

Community Mobile Cloud – is shared by different organizations that have the same or similar security requirements. Shared data lowers security level compared to Private Mobile Cloud. It is required that all organizations that share the cloud trust each other. Because there are many partners the costs are shared among several users and can be significantly reduced.

Public Mobile Cloud – can be used by external organizations and single users. Companies are able to connect their own services to such cloud, with reduced costs and efforts. The problem is that nobody is able to control the structure, so it is not useful to store security sensitive data, e.g. medical information.

Hybrid Mobile Cloud – is a private cloud that is connected to public cloud service. This is a good solution for companies that want to have remote access to some resources. All of the company data is protected because is kept on private servers. This solution provides easier access for trusted users (in this case employees), while retaining data secured.

3. Scheduling Problems and General Scheduling Criteria

For traditional cloud systems, the following scheduling criteria can be considered:

- operational level issues (transmission protocols, scheduling algorithms, energy consumption),
- privacy, security and trust management,
- context-awareness,
- data access,
- Quality of Service.

Each of them can be used also in Mobile Cloud systems. However, due to limitation in the computational and storage capacity of mobile devices, all those criteria should be considered together with energy consumption. It means that scheduling methodologies in mobile cloud environments must be based on the idea of lightweight operations performed by mobile devices. One of the techniques which results in reduction of energy consumption is offloading operation.

3.1. Offloading Methods

In offloading methods, the computation is not executed on the mobile devices but it is distributed among the servers. Sometimes offloading can be done by sending the pointer to the file instead of sending all the data (minimizing of migration), which leads to further energy savings.
Offloading process in mobile systems can be performed in three areas: Client-Server Communication, Virtual Machine migration, Mobile agents [3].

Client-Server Communication is the transmission between offloader (mobile device) and another node. Typical communication protocols include the Remote Procedure Calls (RPC) and Remote Method Invocation (RMI) [3]. They are considered as stable protocols.

Virtual Machine migration – the memory image file of a Virtual Machine is transferred from a source server to the destination server. It is not necessary to stop application execution during migration process [23]. However, Virtual Migration method needs a long time to copy large VM data, so this method can be too slow for mobile devices.

Mobile agents – this method is used for the parallelization of mobile applications processes execution. Agents evaluate servers speed to assess the costs. The process uses special benchmarks.

Another problem is offloading procedures activation moment. Data storing, access and processing in the cloud physical and virtual networks may also increase the overall power consumption in the whole system. As it is presented in [6] offloading is only worth to be implemented in energy aware context, when there is a large number of computation data with proportionally small communication required in the system.

3.2. Dynamic Voltage and Frequency Scaling

Dynamic Voltage and Frequency Scaling (DVFS) is a hardware method used to reduce energy consumption in computing systems. Implementation of this method in cloud systems requires additional hardware circuit that can control power consumption of each node. This can be done by changing the clock speed [24] and/or supply voltage. This approach requires trade off: the best way is to increase clock and supply voltage when processor is busy, and decrease them to the lowest possible level when processor is idle. The power-frequency relation is not always linear, and is different for every platform, so optimum is difficult to find. [6]. Considering the fact that even if server is idle it still consumes energy, the sleeping modes are another solution. This can be applied to the whole system including not only servers but also all separated mobile devices.

4. Energy Efficient Scheduling Algorithms

The scheduler of Mobile Cloud Computing is implemented in PaaS layer. The data is shared by all the users connected to the Cloud at the same time, therefore it might cause some latency. The Quality of Service (QoS) is also a big problem as the number of users and the size of Mobile Cloud increase. Nowadays the solutions for data management in schedulers are well developed but still there are some issues regarding handling the data-intensive applications.
This section presents several heuristic methods implemented for scheduling problem in MC. Heuristic scheduling algorithms can be divided into two groups: heuristic and meta-heuristic. In heuristic methods every task gets priority, then is executed in decreasing order. This method is efficient but doesn't have to be optimal on global scale. Meta-heuristic methods, often called "genetic algorithms", are less efficient; however they find good solutions for complicated problems.
Dynamic structure and limited computing power of the devices in Mobile Cloud requires implementation of special schedulers.

4.1. Mobile Cells Scheduler

In this solution a model similar to mobile agents concept is proposed [25]. The whole cloud structure is build of cells, which are divided into two groups: UMSC (User Mobile Service Cell) and CMSC (Cloud Mobile Service Cell). The scheduling process is divided into three parts:

– UMSC collects requests from users;

– UMSC migrates to CMSC;

– CMSC goes to the MC and looks for the cloud's units able to handle the user requests. Then CMSC sends the information back to the mobile host.

Such methodology eliminates errors caused by turning off the Wi-Fi network or dropping off the mobile devices for Mobile Cloud users, because information is kept in the MC until it is restored. The disadvantage of this methodology appears when the mobile host is not connected while cloud cell wants to receive information. Also the cell region can lost connection with mobile host while it migrates to another cell region. The list of connection information for cloud units can solve this problem. If the wireless connection is fast the workflow is finished by cloud units, in other case the CMSC is saved until the connection is resumed.
As the model explained above, there is implemented the Genetic Algorithm (GA) based on the concept of UMSC and CMSC. Authors in [25] proposed the UMS (Universal Mobile Service) as a group of cells C with variable as arrival time, priority, state. The status of the C cell can have two values: migrated or divided.
The Mobile Cloud nodes are defined as N. The GA operates on couples like (C, N). The first step is to define initial population taking into account priority of the cells and efficiency of the computing system. Next the fitness function F minimize the computational time of the scheduling:

$$F(s) = C + D + R, \tag{1}$$

where C represents the computational time spent on executing cell C by node n, D is transmitting time of the data between cells and R is migrating time between nodes.

Next step is crossover. The representation of population is divided into two parts, for both there are solution S_1 and S_2. Next the cell P is chosen by the random function. The position of P is P_1 in S_1 and P_2 in S_2. Then S_1 and S_2 are divided into two halves and crossed like in traditional GA. Finally the last step is mutation. There are chosen two genetic representations G1 and G2 with the same cell priority. Next they are switched each other [25].

4.2. Scavenger

Scavenger provides cyber-foraging via Wi-Fi and scheduler for costs estimation. The method of estimate costs uses benchmark method to measure the speed of surrogate servers and decides if the offload process should be executed or not. The partitions, jobs, and workflows are provided by mobile code. This framework gives the possibility to offload the mobile devices to several surrogate servers. Executing the applications on a few surrogate servers as a parallel execution is more efficient. The scheduler algorithm includes:

– speed and utilization level of the surrogate server,

– latency of the surrogate and network capacity,

– complexity of the task that is described as a time needed to complete the task on the surrogate.

For every task there are created two profiles: with global task weight and for every task-device couple.
The global task weight is based on the Eq. 2 [4].

$$T_w = T_d(P_s/P_a), \tag{2}$$

where T_d = time needed to complete the task, P_s = NBench result, P_a = number of other tasks that executes at the same time on the same mobile device.

4.3. MobSched

MobSched is a scheduler providing optimization for power consumption and network capacity. It provides at the same time maximum QoS and maximum bandwidth of the computational model. The general model is presented in the Formula 3 [5].

$$\alpha(p_1x_1 + \ldots + p_Nx_N) - \beta(t_1x_1 + \ldots t_ix_i + \ldots t_Nx_N), \tag{3}$$

where α and β are the optimization parameters, p is the power consumed by every node, x is fraction of the work that was dedicated for each node and t is capacity for each node. The optimizing parameters are $0 < \alpha$ and $\beta < 1$ and $\alpha + \beta = 1$.
As the scheduler provides optimization of two (or even more) factors at the same time, there are implemented two optimization parameters (α and β). The whole system is limited by maximum tolerable error. This can be for example a number of lost packets during period.

5. Conclusions

In this work the general model of Mobile Cloud system was presented. For better explanation the MC system was compared to the traditional Computational Cloud. The paper discusses important problems related to MC, i.e. security, privacy, energy consumption and cost analysis. All of those factors contribute to the efficient scheduling. The paper also presents some examples of already implemented energy efficient schedulers, such as MobSched, Scavenger and the Mobile Cell Scheduler. As it can be noticed, the problem of energy efficient scheduling in Mobile Clouds is complex. Still the optimal solution for all of the issues in the Mobile Cloud is not found, which makes the research area wide open.

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Paper

Adaptive Distributed Data Storage for Context-Aware Applications

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Abstract—Context-aware computing is a paradigm that relies on the active use of information coming from a variety of sources, ranging from smartphones to sensors. The paradigm usually leads to storing large volumes of data that need to be processed to derive higher-level context information. The paper presents a cloud-based storage layer for managing sensitive context data. To handle the storage and aggregation of context data for context-aware applications, Clouds are perfect candidates. But a Cloud platform for context-aware computing needs to cope with several requirements: high concurrent access (all data needs to be available to potentially a large number of users), mobility support (such platform should actively use the caches on mobile devices whenever possible, but also cope with storage size limitations), real-time access guarantees – local caches should be located closer to the end-user whenever possible, and persistency (for traceability, a history of the context data should remain available). BlobSeer, a framework for Cloud data storage, is a perfect candidate for storing context data for large-scale applications. It offers capabilities such as persistency, concurrency and support for flexible storage schema requirement. On top of BlobSeer, Context Aware Framework is designed as an extension that offers context-aware data management to higher-level applications, and enables scalable high-throughput under high-concurrency. On a logical level, the most important capabilities offered by Context Aware Framework are transparency, support for mobility, real-time guarantees and support for access based on meta-information. On the physical layer, the most important capability is persistent Cloud storage.

Keywords—Blobseer, brokers, context, distributed, mobile devices communication.

1. Introduction

Today smartphones are becoming commodity hardware. They are seen everywhere, as more people realize that having more sensing and computing capabilities in every-day situations is attractive for many reasons. Smartphones are in fact already used to optimize (e.g., by helping organizing tasks, contacts, etc.) and assist (e.g., with navigation, find information more quickly, access online data, etc.) users with their everyday activities. Their success is the basis for a shift towards developing mobile applications that are capable to recognize and pro-actively react to user's own environment. Such context-aware mobile applications can help people better interact between themselves and with their surrounding environments. This is the basis for a paradigm where the context is actively used by applications designed to take smarter and automated decisions: mute the phone when the user is in a meeting, show relevant information for the user's current location, assist the user find its way around a city, or automatically recommend events based on the user's (possibly learned) profile and interests.

This vision is supported today by the inclusion of context as an active operational parameter of service provisioning. For many mobile applications an important requirement is represented by the active sensing of the operational environment, as users expect to receive only relevant content back on their mobile devices (e.g., when the user accesses a transportation service he expects to receive a route relevant for current location). The advances in mobile technologies support the inclusion of context as an active operational parameter for such applications, as today's mobile devices allow the users to acquire and manipulate complex, multifaceted information in real time, and to interact with each other in seamless ways. The range of applications using context is increasing rapidly, and it spans from urban navigation, cultural heritage to entertainment and peer-to-peer communication. The challenge for such pervasive applications is how to make them continuously adapt to dynamic changes in the environment, even when people move and when the underlying network architecture can offer only limited services.

To support this, the authors designed the Context Aware Framework as a middleware to automate the provisioning of context data to mobile applications. It sits between a persistence layer, where data is actually stored in a Cloud storage system, and the actual context-aware application running on the user's mobile devices, masking the complexity of managing the data. In our vision, a truly context-aware system is one that actively and autonomously adapts and provides the appropriate services or content to the users, using the advantages of contextual information without too much user interaction. Thus, providing efficient mechanisms for provisioning context-sensitive data to users is an important challenge for these systems. Context Aware Framework is designed to support the storage of context data for such context-aware systems. It manages every problem related to, for example, the unpredictable wireless network connectivity and data privacy concerns over the network, providing transparent access to the data to such systems.

The contribution of this paper is twofold: we first introduce the Context Aware Framework middleware, together with the design requirements that motivated our choices; we then present experimental results, supporting our decisions and illustrating the performance obtained when using Context Aware Framework.

The rest of the paper is organized as follows. Section 2 presents an overview of the main relevant work in the field. Section 3 makes an analysis of the main requirements for context-aware storage systems and presents the architecture of the Context Aware Framework that we devised following this analysis. Section 4 presents details of the implementation, while Section 5 presents evaluation scenarios and results illustrating the overall obtained system performance. The paper is concluded in Section 6.

2. Related Work

The ubiquity of mobile devices and sensor pervasiveness call for scalable computing platforms to store and process the vast amounts of the generated streamed data. Cloud computing provides some of the features needed for these massive data streaming applications. For example, the dynamic allocation of resources on an as-needed basis addresses the variability in sensor and location data distributions over time. According to the Association for Computer Operations Management (AFCOM), in year 2011 90.9% of data center sites used more storage space than they did three years ago. During that same three-year period, 37% were able to reduce their staff, and 29% kept their staffing levels the same. This trend is in large part due to the development and implementation of new tools and processes that have allowed IT departments and data centers to store massive amounts of data efficiently and inexpensively. The advent of cloud-based storage systems has had since then a profound impact on the way businesses collect and store their information. However, today's cloud computing platforms lack very important features that are necessary in order to support the massive amounts of data streams envisioned by the massive and ubiquitous dissemination of sensors and mobile devices of all sorts in smart-city-scale applications.

Several cross-device context-aware application middleware systems have been developed previously. In their majority these were Web service-based context-aware systems, especially the most recent ones. However, there has been a big variety of middleware systems, developed mainly in the early 2000, that do not rely on Web service technologies and are not designed to work on Web service-based environments [1]. In this work the authors began by studying several popular context-aware platforms, considering their provided functions and particular characteristics. From the category of non-based on web service context-aware platforms the following could be mentioned.

RCSM [2] is a middleware supporting context sensitive applications based on an object model: context-sensitive applications are modeled as objects. RCSM supports situation awareness by providing a special language for specifying situation awareness requirements. Based on these requirements, application-specific object containers for runtime situation analysis will be generated. RCSM runtime system obtains context data from different sources and provides the data to object containers which conduct the situation analysis.

The **JCAF** (Java Context Awareness Framework) [3] supports both the infrastructure and the programming framework for developing context-aware applications in Java. Contextual information is handled by separate services to which clients can publish and from which they can retrieve contextual. The communication is based on Java RMI (Remote Method Invocation). An example of application that use Java RMI is MultiCaR: Remote Invocation for large scale, Context-Aware Applications [4]. This application also address the issue of big data analytics.

The **PACE** middleware [5] provides context and preference management together with a programming toolkit and tools for assisting context-aware applications to store, access, and utilize contextual information managed by the middleware. PACE supports context-aware applications to make decisions based on user preferences.

CAMUS is an infrastructure for context-aware network-based intelligent robots [6]. It supports various types of context information, such as user, place and environment, and context reasoning. However, this system is not based on Web services and it works in a close environment.

SOCAM is a middleware for building context-aware services [7]. It supports context modeling and reasoning based on OWL. However, its implementation is based on RMI.

Web service-based context-aware platforms include the following.

CoWSAMI is a middleware supporting context-awareness in pervasive environments [8]. The **ESCAPE** framework [1] is a Web services-based context management system for teamwork and disaster management. ESCAPE services are designed for a front-end of mobile devices and the back-end of high end systems. The front-end part includes components support for context sensing and sharing that are based on Web services and are executed in an ad hoc network of mobile devices. The back-end includes a Web service for storing and sharing context information among different front-ends.

The **inContext** project [1] provides various techniques for supporting context-awareness in emerging team collaboration. It is designed for Web services-based collaborative working environments. inContext provides techniques for modeling, storing, reasoning, and exchanging context information among Web services.

Being context-aware allows software not only to be able to deal with changes in the environment the software operates in, but also being able to improve the response to the use of the software. That means context-awareness techniques aim at supporting both functional and non-functional software requirements. Authors of [9] identified three important context-awareness behaviors:

- the representation of available information and services to an end user,

– the automatic execution of a service,

– the tagging and storing of context information for later retrieval.

For massive context data streaming applications, M3 [10] is a prototype data streaming system that is being realized at Purdue using Hadoop. M3 eliminates all of Hadoop's disk layers, including the distributed file system (HDFS), and the disk-based communication layer between the mappers and the reducers. It proposes a hybrid memory-based and disk-based processing layer, includes dynamic rate-based load-balancing and multi-stream partitioning algorithms, and fault-tolerance techniques. However, M3 can handle only streaming data and does not handle queries that mix streaming with disk-based data. A context awareness extensible layer for M3 has been demonstrated separately in Chameleon [11]. However, Chameleon lacks general context-based indexing techniques for realizing context awareness, thus when the context changes the system cannot easily augment the query being executed by additional predicates to reflect that change.

Similar to Context Aware Framework, the author of [12] present a large scale system, called Federated Brokers, for context-aware applications. In order to avoid the centralized design (single point of failure) found in previous papers, the authors propose a context-aware platform that includes multiple brokers. The main difference compared to our system is that our architecture implies the existence of a metadata manager for all stored information, which relieves much of the burden impose for managing data on the mobile application, and that we offer prediction capabilities based on the provisioned data. Also notable is that the evaluation of Federated Brokers was conducted over a small-size homogeneous environment. The authors actually tested Context Aware Framework over a large grid environment, with more brokers and clients, considering a large distributed storage configuration.

From an utility point of view, our platform can also be compared with Google Now [13] and Microsoft On{X} [14]. Google Now [13] is able to predict what information an user need, based on his previous searches and on his context data. Microsoft On{X} lets the user set actions for states defined by his context data. When a certain state (previous defined by the user) is reached, a trigger is fired. This platform supports this kind of approaches, being build as a framework for developers, not as a stand alone application, like Google Now or Microsoft On{X}. Unlike previous work, the platform presented in this paper is specifically designed for the management of large-scale pervasive environments. The platform is designed to support scenarios with potentially thousands of sensors working together for the fine-grain analysis of phenomena. For example, the platform can cope with Smart City context-aware applications and services, providing citizens with real-time information, regardless of their mobility constraints, about current traffic values or other events of interests.

3. Architecture

3.1. Analysis of Requirements for Context-aware Applications

In a typical context-aware application users rely on their mobile devices to receive information depending on their current environment. Such applications can help users augment their reality: they could receive information about neighboring places or buildings in a typical tourism application; they could receive recommendations or navigation data that could help a driver to more easily navigate around a city. In such scenarios users are generally *moving*, and typical context data includes elements such as locality, time, user's status, etc. *Proximity* is important for provisioning – the amount of data is potentially too large to be served entirely on the user's mobile device, thus a selection of only the most relevant context data, from the immediate surrounding environment, is preferred. Therefore, Context Aware Framework should be able to support user's mobility and provisioning of data according to his locality.

Except for mobility, context-aware applications should provide real-time guarantees for data provisioning. The user should not receive events that happened too far in the past, as such events might not even be valid for his current interests. For example, if a tourist is looking for information about objectives near him, he might not be so happy receiving recommendations for a trip taken some while back. The same might happen when users expect alerts about potential congestions in traffic: if he receives such alerts when is already in the congested road, the information is not so valuable anymore. To cope with such a requirement, Context Aware Framework should support fast dissemination of data between users.

We also acknowledge the imperfections of today's wireless communication infrastructures. Context Aware Framework will not assume the user is always connected to the Internet (e.g., in situations where a wireless connection is not available). It will support the use of context data even when an Internet connection is not available, and in this case we look at alternatives such as opportunistically using the data accessed by others from distributed caches using only short-range communication (in form of Bluetooth and/or Wi-Fi).

The system should allow efficient access to the data – in terms of speed of access, as well as support for complex queries. Applications should be able to express their interests using complex queries, in forms of naturally-expressed filters. For example, the application will be able to request the data using an expression similar to "get prediction of my friends' location, but only for those in town". Or, an aggregated request could be expressed as "get prediction of road traffic on a particular street".

For context-aware applications we consider that the system should support discovery and registration of data sources (e.g., sensors and external services such as a weather service), access to data using different granularities, and the aggregation of information.

The framework needs to support scalability. For a typical collaborative traffic application, the number of users could potentially be in the range of millions. The data should be persistently stored. The history of data should be preserved for traceability and advanced data mining processing.

Except for these theoretical requirements, the functional demands coming from a context-aware application was also analyzed. For this we analyzed CAPIM [15], a platform designed to support context-aware services. Such services are designed to help people in an university, who may be endowed with a portable device, on top of which they run an application which facilitates the access to information by automatically reacting to changes of context. CAPIM brings support by:

– providing different information contents based on the different interests/profiles of the visitor (student or professor, having scientific interests in automatic systems or computer science, etc.), and on the room he is currently in;

– learning, from the previous choices formed by the visitor, what information he is going to be interested in next;

– providing the visitor with appropriate services – to see the user's university records only if appropriate credentials are provided, to use the university's intranet if the user is enrolled as staff;

– deriving location information from sensors which monitor the user environment;

– provide active features within the various areas of the university, which alert people with hints and stimuli on what is going on in each particular ambient.

In addition to the previously identified requirements, this scenario validated several new ones in terms of data accesses: users write frequently, while they read the data in a sparse way. They have also an interest in storage of large data volumes, for mining and processing relevant high-level context information.

To cope with these requirements, Context Aware Framework should include several layers. First of all, for persistence, collaborative and mobility support, the data should be stored remotely to any mobile device. A typical relational database has the disadvantage that accesses to the same data units should be synchronized for strong consistency guarantees. This cannot support well a typical scenario envisioning millions of concurrent users writing their context data.

BlobSeer [16] is a large-scale, distributed, binary storage service. It keeps versions for all records, so that concurrent read/write accesses are facilitated without affecting the high throughput of the system. BlobSeer allows concurrent accesses to the data, and for that it uses a versioning mechanism. Another benefit is that BlobSeer allows fine grain access to the data. It is possible to access small chunks, without having to read the entire Blob for example. BlobSeer also offers high throughput for read and write operations. Clients can write new information in a chunk while others can read the old information, without needing to synchronize.

Thus, BlobSeer [16] offers an appropriate alternative, as it provides real-time guarantees, large concurrent access guarantees, and support for eventual consistency through an advanced versioning mechanism. This is what motivated our choices in what follows.

3.2. Architecture

Based on the identified requirements, we propose the architecture presented in Fig. 1 is proposed.

The architecture includes several components (see Fig. 2): Data and Metadata Clients, Brokers, the Metadata Manager, and a Cloud-based storage layer.

The **Metadata Client** and **Data Client** connect the Context Aware Framework with the third-party context-aware applications. In the practical implementation (detailed in the next Section) both these software components are integrated into the context-aware application in need of their support (they offer an API for such applications).

The **Metadata Client** is responsible for creating and accessing the metadata information that describes the context data schema used by a particular application. In fact, we acknowledge that various context-aware applications have different requirements in terms of the data scheme used internally. Consequently, each application can use a different data schema to model the context.

A **Data Client** can write, retrieve, and store context data necessary to a particular context-aware application. Here we assume a one-to-one relation, each application being served by a dedicated Data Client. Each Data Client is responsible for supporting the mobility of the user, supporting seamless access to the nearest **Broker**. The Data Client works with its own local cache, used for offline situations, when the user cannot access anymore the data from its Broker.

A dedicated **Discovery Service** is also used for the registration and discovery of the existing Metadata Manager and Brokers. In the architecture we assume the existence of one Metadata Manager, but several Brokers. The Discovery Service is, therefore, also responsable for finding the Broker most convenient for a particular Client.

The Metadata Manager manages the connections between the meta-information describing the data, and the information regarding the actual physical data storage. When a new context-aware application is registered for the first time, the Metadata Client connects to the Metadata Manager and writes meta-information describing that particular application. The information contains, among others, the datatype formats to describe the context data collected/stored by the application. Next, when the Data Client writes context data, it connects to the nearest Broker. The context data is sent to the Broker, which in turn writes it to the **Persistence layer**. The process involves two steps: first the Client writes the data, and next asynchronously the Broker handles

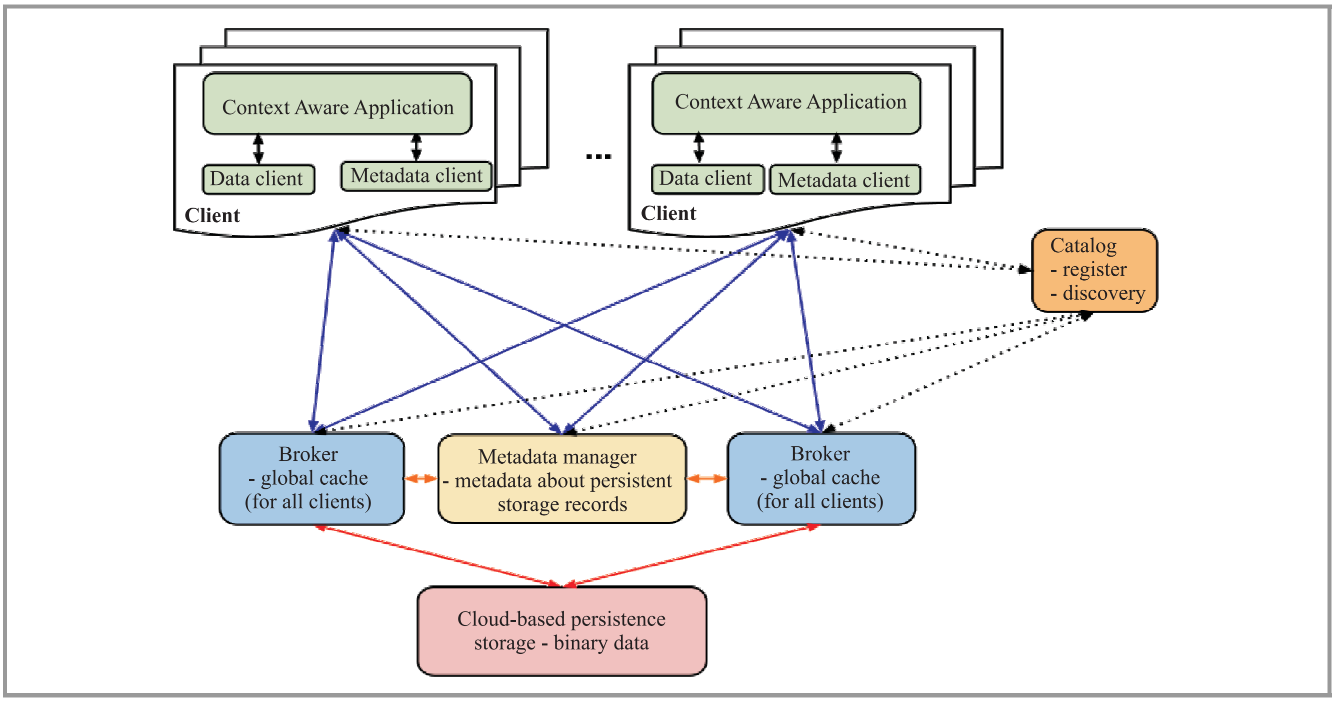

Fig. 1. The proposed architecture.

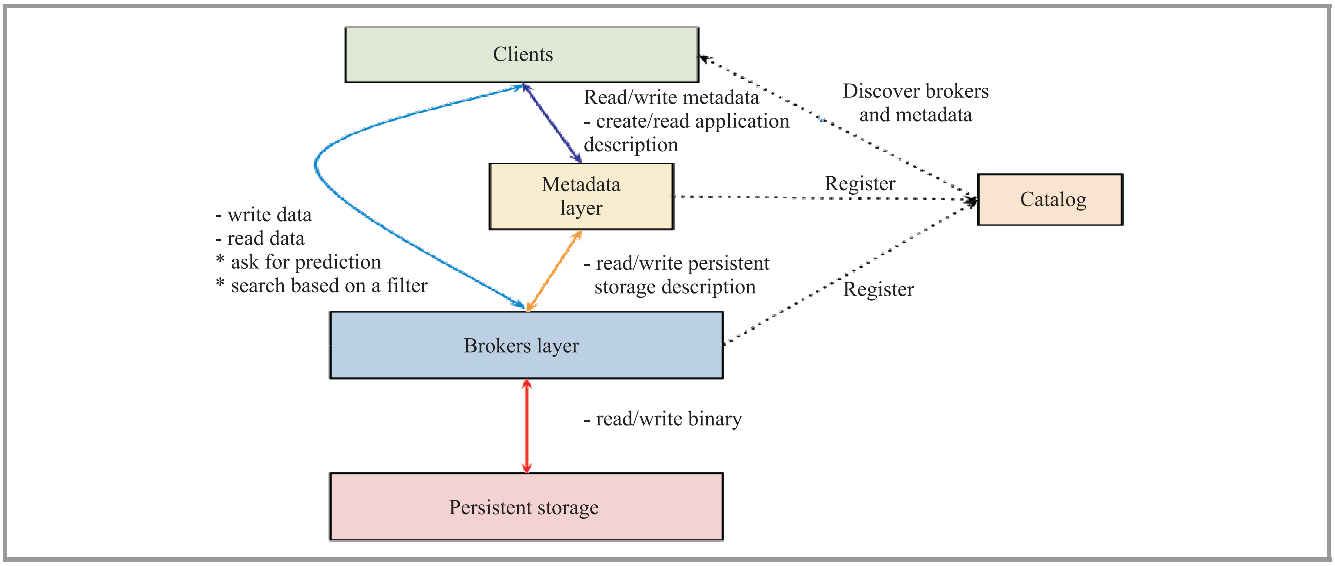

Fig. 2. Architecture layers.

transparently (in background) the actual writing into the Persistence layer.

The Broker also writes information describing the physical storage parameters to the Metadata Manager. For persistent storage we decided to adopt the use of Blobs. A Blob stores the context data needed by one context-aware application. The Metadata Manager actually links the meta-information all the way to a particular Blob and to an offset inside it where the particular data resides. The Metadata Manager also manages the relation between the persistent Blobs (the ones used for history preservation of context data) and the Brokers, where the real-time information is preserved.

The Context Aware Framework comprises several distributed Brokers. The Broker is responsible for handling real-time guarantees specified when an application wants to access context data. The Broker handles requests coming from a limited number of users, grouped based on their locality. It supports distributed writing of data, and processing of requests coming from clients.

For accessing the context data, a Client application generates a filter (expressing the parameters of interest) for finding it. This filter is further received and processed by the Broker. The resulting data is sent back to the Client, and is also temporarily stored in a cache, local to the Broker. This cache is used to speed up the response time for subsequent requests for similar data. If another client sends a similar request, the Broker is capable to reply directly with the data from its own cache unless the data was invalidated by a subsequent write.

For the Persistence layer we use the BlobSeer [16] storage distributed system. BlobSeer consists of a set of distributed entities that cooperate to enable a high throughput storage. Data providers physically store the blocks corresponding to the data updates. New providers may dynamically join and leave the system. The provider manager keeps information about the available storage space and schedules the placement of newly generated blocks, according to a load balancing strategy. Metadata providers store the information that allows identifying the blocks that make up a version of the data. The version manager is in charge of assigning version numbers in such a way that serialization and atomicity are guaranteed. In addition, clients can access the Blobs with full concurrency, even if they all access the same Blob. One can get data from the system (Read), update it by writing a specific range within the Blob (Write) or add new data to existing Blobs (Append). Rather than updating the current pages, each such operation generates a new set of pages corresponding to a new version. Metadata is then generated and "weaved" together with the old metadata in such way as to create the illusion of a new incremental snapshot. This actually shares the unmodified pages of the Blob with the older versions.

For our Context Aware Framework Clients can sometimes access the second to the last version of the context-aware data until one write-in-progress operation is finished. Context Aware Framework uses this to provide to the higher-lever applications eventual consistency support for read/write operations.

Typical context-aware applications [15] usually generate big amounts of unstructured or semistructured data. Applications can interpret this data in particular ways, by defining appropriate meta-information associated with it. The applications can decide on their own different granularities – for example, an application can write several chunks of data at once, for the data corresponding to several events, and define one single meta-entry to describe this. It is entirely left in the responsibility of the application to define its use and schema corresponding to the context data and associated model.

3.3. Overall Architectural Benefits

The architecture of Context Aware Framework brings several capabilities mentioned below.

The framework is designed for context-aware applications that work with data represented mostly as time-based series, where entries are in the form $\langle timestamps, Object \rangle$.

The architecture supports scalable applications. Once deployed, the system can support a large number of applications, involving potentially large number of users, each with its own context data. This is because each application runs in a separate environment.

Context Aware Framework provides Locality, Mobility and Real-time access guarantees. In order to have good response times, a client will connect to the closest Broker before launching a request. All read operations are cached on two levels: one is on the Data Client side, and one on the Broker side. If two clients issue the same request, the response for the second one will be fetched from the Broker's cache. This ensures both a good response time.

Persistence is also supported. Clients write their data, which in turn is saved in the storage system (where we use BlobSeer).

Later on, clients can ask for data, through complex search filters. Also, prediction is supported. In order to benefit from the large amount of stored data, clients can activate predictions for a specific set of data (see Section 4.2). When this is happening, the context data is pre-fetched on the Broker cache, based on complex prediction algorithms. This can be used to cache in advance data for certain data types.

4. Implementation

Next a pilot implementation of the Context Aware Framework architecture previously described was developed.

As explained, the Metadata Manager is responsible for handling the logical relation between the description of the context data and its actual physical storage. In example in a large city many users might send GPS data to collaboratively support an application capable to aggregate this information and offer a traffic model. Some users are capable to also send data about pollution (they have sensors for monitoring the air quality). We assume this information is sent and stored using the previously described Context Aware Framework.

First, the Client will write in the Metadata Manager the datatypes used by the application:

```
object Location {
    float lat, long;
    string hw_description;
}

object COLevel {
    float level;
    string hw_description;
}
```

Next, different Clients will write the actual context data, which is similar to:

```
array{Timestamp, Location} ==> {
 {243452343L, {14.5, 34.45, 'Nexus Galaxy'}},
 {243452354L, {14.51, 34.467, 'Nexus Galaxy'}},
 {243452368L, {14.53, 34.473, 'Nexus Galaxy'}}
}

array{Timestamp, COLevel} ==> {
 {243452344L, {45.3, 'Air Quality Sensor'}},
 {243452360L, {45.4, 'Air Quality Sensor'}},
 {243452412L, {45.37, 'Air Quality Sensor'}}
}
```

The data is written in a Blob, inside the Persistence layer – the actual data is stored in BlobSeer. In this example, the data is written in bursts. We support this feature in cases, for example, when a car can collect data and sent it only when a Wi-Fi connection becomes available. The actual information used to describe the physical storage looks similar to

```
{UUID, BlobID, BlobVers, BlobOffs, Size}
```

where UUID refers to the application id that generated the data, BlobID, BlobVers, BlobOffs and Size identify the blob, its version, the data offset and size in the Blob where the information was written. Next, the Metadata Manager adds an entry linking the UUID to the

```
{TimestampStart, TimestampEnd, DataType, UUID,
 BlobID, BlobVers, BlobOffs, Size, NoRecords}:
```

(e.g.,

```
{243452343L, 243452368L, 'Location', 0x242,
   213412L, 34, 0, 1234402L, 3}).
```

The actual implementation of the Metadata Manager uses Mongodb [17], a flexible open source document-oriented NoSQL database system. Mongodb includes support for master-slave replication and load balancing. For searching, it also supports regex queries. For Context Aware Framework, the database system was preferred for several reasons: The number of entries kept by the Metatada Manager – entries previously described – is small. Each entry follows a structured object-oriented data schema. Consequently, an object-oriented database model is preferred.

Also, when the number of metadata access requests becomes high enough, the system should be able to scale. MongoDB, the distributed object relational database, is the natural choice, because it support distributed deployment and high scalability [18].

The Metadata Manager is also collaboratively used by different applications. For security and management reasons, in the actual implementation each application stores its related data in separate sandboxes.

4.1. Filtering

As previously described, for accessing the data the Client builds a search filter. This can include different custom data types defined by an application. The filter specifies the restrictions for searching particular datatypes. For instance, a filter can include restrictions for retrieving specific location and pollution levels. In this example, the filter looks similar to:

```
class Filter {
    Location l;
    COLevel c;
    ...
    bool filter() {
        return  l.lat > 10.53 and
            l.long < 20.45 and
            c.level < 15;
    }
}
```

The filter result is in format (timestamp, location, level).

The Client sends the serialized version of this filter class, and the Broker loads it and instantiate it with values that match the implementation of the filter instance.

4.2. Prediction

Prediction is done using linear interpolation (in the pilot implementation Lagrange interpolation was used). The prediction module is extensible, and the user/application can easily replace it.

For predicting a future value based on a time-dependentł;inebreak series, the user specifies several parameters. The predictability pattern specifies how the data varies (possible values include daily, weekly, or hourly patterns). For example, a daily pattern considers that data is similar for the same hours each day, while a weekly pattern assumes data is similar for the same days each week.

To optimize the prediction process, only a subset of all data in history is used by the prediction algorithm (a time-window like approach, considering only the last most relevant values). The interpolation considers the set of last values and depends on the type of prediction pattern selected.

To use this facility, the API allows the user to specify *N*, and two timestamp values. *N* specifies the number of predicted values the user is requesting – and it is used to define the granularity of the sampling history data. The two timestamps specify the interval in the future of interest for the prediction – the prediction returns in this case the *N* values spread over the requested interval, by mediating the obtained predicted values. Obviously, if the prediction cannot be performed (or the error is too high), the returned answer can be also none.

We applied the prediction facility to implement an adaptive cache. Such cache is filled with values that it predicts the user will need in the near future – thus, it can support the losing of the connectivity, or it can support an optimization by requesting data asynchronously from the persistence layer before the actual request for data takes place.

Let's consider the example of an application requesting data about weather. In this case weather is considered to be a function of hour and location. A predictive cache could predict the location of the user in the near future, as well as the time moment he will get there. Thus, it will be able to further interrogate a weather service and request the weather values in advance. We tested this assuming that a client requests a new weather value every 30 minutes, and the cache replenishes the weather values in advance, such that by the time client makes the actual request, the cache is able to opportunistically serve him the data (i.e., even in case an Internet connection is no longer in place).

5. Experiments, Evaluation and Results

5.1. Experimental Setup

For evaluating Context Aware Framework, the following scenario is used: Many taxis from a city are equipped with mobile devices that run a context-aware application. This application collects GPS data, and sends it to a server. Clients are presented with context-aware capabilities, such as searching for nearby free taxis or inspecting routes (for example, the municipality can learn the popularity of routes).

As input data, a real-world dataset publically available on CRAWDAD is used [19]. The dataset contains mobility traces of taxi cabs in San Francisco, USA, in the form of GPS coordinates for approximately 500 taxis collected over 30 days in the San Francisco Bay Area. It includes approximately 11 millions unique entries.

These taxis were considered as clients for our context-aware middleware. They were able to write data, and use different access patterns to obtain context-based information. Each client runs on a different node inside a distributed system. For these experiments we used Grid'5000 [20], a large-scale distributed testbed specifically designed for research experiments in parallel, of large-scale distributed computing and networking applications.

To evaluate the performance of Context Aware Framework we had to first filter the data for each unique taxi in the experiment. Therefore, 500 different input files was used, with an average of approximately 20,000 records per file. Each record is specified as [latitude, longitude, occupancy, time]. For example, a record is expressed as [37.75134 –122.39488 0 1213084687], where latitude and longitude are in decimal degrees, occupancy shows if a cab has a fare (1 = occupied, 0 = free) and time is in Unix epoch format. For the storage layer, we used BlobSeer. The total data written by each taxi is approximately 5 MB.

In Grid'5000 112 dedicated parallel nodes for the clients was used, and 4 other dedicated parallel nodes for 4 Brokers. One other dedicated node was used for the Metadata Manager, and another one for BlobSeer. In these experiments we used an increasing number of Brokers – ending with the 4-based Broker experiment. We assume the city is equally split between these Brokers – if a taxi always connects to the nearest Broker, the mobility data is equilibrated such that we obtained an approximately even number of data sent to each Broker. Thus, the number of clients distributed per broker is uniform.

During the experiment configuration data was varied, such as: the parameters used for BlobSeer configuration (number of data providers, and page size was progressively in-

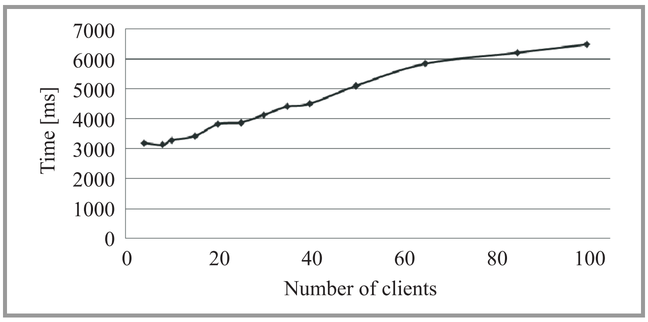

Fig. 3. Write test.

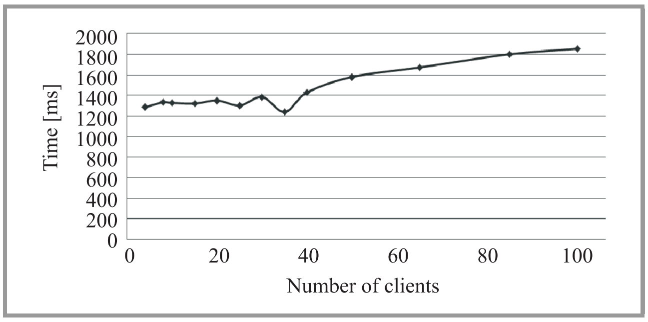

Fig. 4. Simple search.

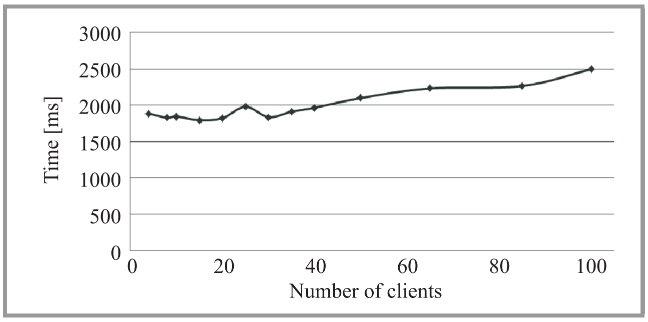

Fig. 5. Complex search.

creased up to 12 MB), the number of clients and brokers, the maximum records written per chunk. We were particularly interested in time taken to perform different operations (to illustrate the capability to support real-time traffic), as well as in the consumed data traffic, to evaluate the optimization obtained when adding the caches.

First, the writing performance was evaluated. For this the authors conducted several experiments where the number of clients that write data (entire input files) to Context Aware Framework was increased. Since 112 dedicated nodes is used, the evaluation is relevant up to this limit – the results are presented in Fig. 3. Figures 4 and 5 show the result obtained for different read access patterns. Compared to these figures, the write operation is more time consuming. Still, the time increases by small amounts, thus the system shows good scalability results.

For evaluating the read operations, we considered two different scenarios. First, a simple search consists in a query where a driver wants to obtain all data relevant for a particular location (given as latitude and longitude limits) and time period. A more complex search operation is one where a client queries the system for the nearest free taxi considering a particular time moment and location. For such query the system has to aggregate data from two different data types.

Again, we varied the number of clients assumed in the experiment, up to 112. The experiment ran until Context Aware Framework has all the context data persistently written. When all data is written, next all clients issue a filter such that all queries will always return results. In a first experiment, we used the same filter, but the caches will return always the value and the time penalty is minimal. We

next assumed that each client issues a unique filter, thus each query is served by questioning the last layer: BlobSeer. We were interested to see the Broker's capability to support parallel client requests. Figures 4 and 5 show the results obtained in this case.
Again, in this case Context Aware Framework is able to successfully handle the queries coming from distinct clients. The results show that time increases by small amounts, thus the system shows again good scalability results.

5.2. Evaluation of Prediction

Next, the prediction component is evaluated. In this experiment the prediction is activated for each taxi within the dataset. The input data file is splitted in two parts: 80% of the data was used as input for learning, and then 20% of the data was used for the evaluation of the prediction accuracy. The predictor in this case uses the data to predict where a cab will be for future time moments, considering daily repeatability patterns – it can be used by a client to search for the nearest taxis, for example, at a future moment of time. In this case, as mentioned, the authors were particularly interested to measure the prediction accuracy.
The results in Fig. 6 show a cumulative graph for number of values passing a prediction acceptance threshold. A threshold of 10% means, for example, that for a variation range of 40 km, a value predicted with a 4 km error is still accepted as being correct. For the experiment, a 10% threshold means that a value is predicted such that, when compared to the real observed value in the input file, it gives a variation of no more than 10% of the entire city area, assuming that each car drives through the entire city during its experimental lifetime and has an equal probability to be at a certain moment of time in any of the next probable locations.

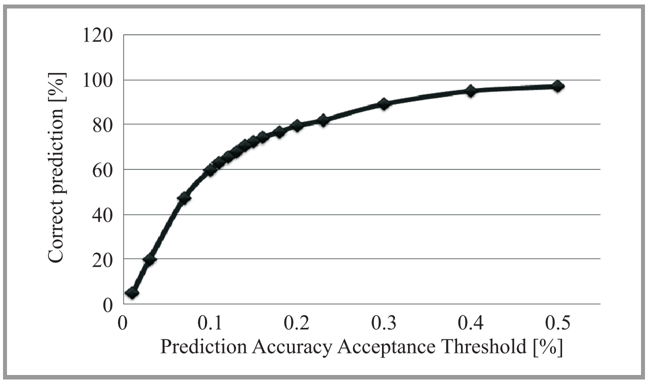

Fig. 6. Accuracy acceptance.

Looking at Fig. 6, when accuracy acceptance percent goes down, there is a random factor that determines some of the values to still be correct. When the percent goes up, there are some "unpredictable" values that make the prediction slightly lower then 100%.
Also, it can be observed that a good threshold is around the value 0.2 for accuracy acceptance, where the prediction becomes very good, yielding approximately 80% correct predicted results.
Next the prediction type was varied (considering hourly or weekly patterns). It can be observed that the correct prediction behavior is similar, but it depends on the nature of the dataset and assumed prediction pattern. For example, predicting with one hour pattern for a too large time interval results in inaccurate prediction results, because the cabs' moving patterns is not hourly based (8 am traffic, for example, is different than the 11 am one).

5.3. Predictive Cache

A good use of the prediction module consists in the implementation of a predictive cache that can be used by a mobile application. The cache sits on the mobile device, tightly coupled with the application, and uses prediction to obtain in an opportunistic way data from the storage layer.
First an experiment was designed to test the prediction accuracy of the predictor in a real application. In this experiment, from time to time (e.g., once an hour), a background process asks for a predicted value for the location (e.g., using a pattern such as predict my location after an hour). Then, the process uses this predicted location to ask further for the weather, having both the location and the time for the next interval (hour). Then, the answer is saved into the cache. So, after an hour, the user can ask for the weather in his locations, and the answer can be found in the cache with a 80–100% location prediction accuracy.
These experiments were done on a machine with the following characteristics: Intel Core i5 processor, 2.5 GHz, 4 GB RAM. Unlike the next series of experiments, in this case we assumed that all Clients are always connected to the Internet (and, thus, can access at all times the Broker). To test the implementation, we have used one Broker. Clients are periodically (every 30 minutes) asking for their predicted location. The obtained results (Fig. 7) show that the answer time increases logarithmicaly with the number of clients. Thus, that the predictive cache scales very well for a big number of clients.

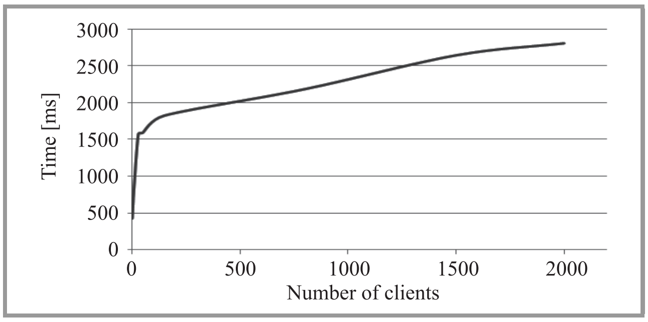

Fig. 7. Predictive cache.

For evaluating this capability, next an application that runs on the user's mobile device and present him with traffic information is simulated. This kind of data is context-aware because the user is interested to receive traffic information depending on his both time and location.

The scenario consists in cabs from San Francisco moving inside the town and trying to acquire the traffic information using a public service. Their only way to connect to the Internet is using Wi-Fi hotspots (3G/4G is too expensive for a large scale system), distributed in a grid configuration through all the town (see Fig. 8).

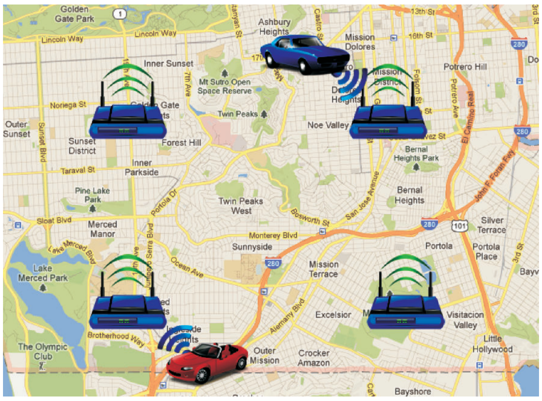

Fig. 8. Taxis connect to the nearest Wi-Fi access points.

The grid was chosen because it provides a good covering of the town with fewer resources than other configurations. The active area of the town is around 250 square km, taking into account our users moving pattern from the input dataset.

We assume that the prediction component is available in the form of a Web service, reachable over the Internet. In our scenario we assumed users ask for new traffic information every 30 minutes (access the Web service).

We also assumed the traffic data does not dependent necessarily on the real-time information. For a typical traffic navigator, the traffic data is generally served by aggregating the traffic data for a certain period of time. This assumption was needed because we assume the information is still valid, even if it is kept in cache for 30 minutes.

Next, the following scenario is envisioned.

From time to time (30 minutes), the user tries to access relevant traffic information, related to his time and location. In the implementation this is accomplished by a background process continuously waking up periodically in order to ask the Context Aware Framework about the most "possible" future location of a particular car. This process, which actually simulates the behavior of the Client cache, then downloads traffic information related to his future predicted location – for this, it sends to the traffic prediction Web service the time in future for which it wants the information. The request will be served only if there is an Internet connection available at the moment the request is issued. If not, the service will fail to bring results. If successful, the returning pair (future time, future location) will be locally cached (on the Client cache).

When the client will actually need the traffic data, if the predicted value for its future location was computed correctly it will actually use the cached data. This means that this client will not need an Internet connection to access this new data, and it will have it fast (since it is already cached locally).

Because traffic information is very sensitive to the current user location, in experiments relevant only location values predicted with an accuracy error lower than 5% was considered.

Considering the scenario and the experiment conditions described above, the average time for one request is plotted, having the predictive cache on or off. In the experiment, the number of hotspots in the town's Wi-Fi grid is varied, from 40 to 120 different access points. The obtained results are presented in Figs. 9 and 10.

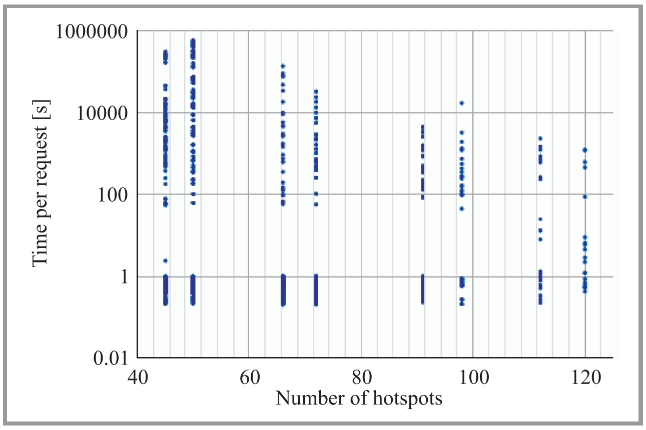

Fig. 9. Predictive cache on.

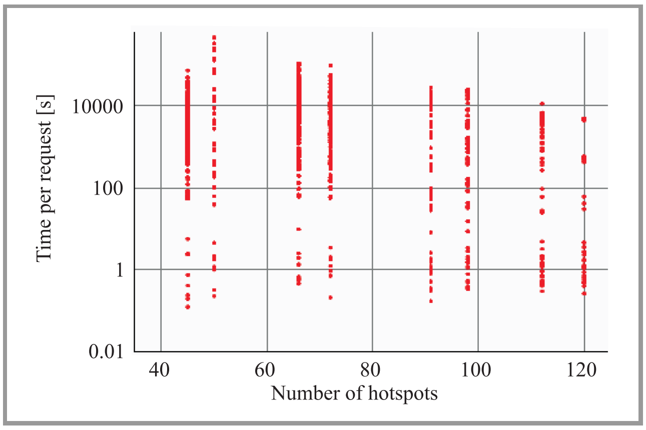

Fig. 10. Predictive cache off.

When cache is active the time necessary to serve each request is actually decreasing. A large amount of requests are finishing under 1 second, with or without Internet connection. When the cache is stopped, only requests which are issued by clients within Wi-Fi coverage zones are still served within good time limits, while for the others taxis are not capable to acquire the data until they reach Internet connectivity.

Since there is a compromise between the time for a request and the density of Wi-Fi hotspots, as seen in the plots, the number of hotspots has an important impact over how well the queries are served when using the predictive cache. However, we cannot assume a too much density of such hotspots, considering our scenario that covers only approximately 250 sq·km. For instance, a more detailed analysis for 66 hotspots, when we vary the acceptance level of the prediction to 0.13 (if the predicted location doesn't need to be so precise) revealed that only 12.5% of the requests need an Internet connection. The rest of them were cache hits. This, combined with the fact that only 20% of requests are issued when cabs have Internet connection, leads to only 10% probability for a request not to be solved at the moment it was made.

6. Conclusions and Further Work

In this paper we presented a platform designed to support several hot challenges related to big data management on clouds by focusing on a particular class of applications: context-aware data-intensive applications. A representative application category is that of Smart Cities, which covers a large spectrum of needs in public safety, water and energy management, smart buildings, government and agency administration, transportation, health, education, etc. Today, many Smart City applications are context-based and event-driven, which means they react to new events and context changes. Such applications have specific data access patterns (frequent, periodic or ad-hoc access, inter-related data access, etc.) and address specific QoS requirements to data storage and processing services (i.e., response time, interrogation rate). With the advent of mobile devices (such as smartphones and tablets) that contain various types of sensors (like GPS, compass, microphone, camera, proximity sensors), the shape of context-aware or pervasive systems changed. Previously, context was only collected from static sensor networks, where each sensor had a well-defined purpose and the format of the data returned was well-known in advance and could not change, regardless of any factors. Nowadays, mobile devices are equipped with multimodal sensing capabilities, and the sensor networks have a much more dynamic behavior due to the high levels of mobility and heterogeneity.

Context Aware Framework is designed to support such requirements. In a pervasive world, where the environment is saturated with all kinds of sensors and networking capabilities, support is needed for dynamic discovery of and efficient access to context sources of information. Such requirements are mediated in our case through a dedicated context management layer, which is responsible for discovering and exchanging context information. The authors presented the context storage system architecture for data management that includes an additional set of components. This supports the mapping between meta-information (describing the context) and the actual context data stored in BlobSeer, data caching and handling requests coming from a distinct set of users or city area, and connecting the metadata management layer to context-aware applications. In addition, we presented a layer that is responsible for creating and accessing the metadata information that describes the context data schema used by a particular application and allows the mobile application to write, retrieve, and store context data. It is also responsible for supporting user's mobility. The components support several requirements: user's mobility and provisioning of data according to his locality; real-time guarantees for data provisioning; allow efficient access to the data in terms of speed of access, as well as support for complex queries; discovery and registration of data sources and access to data using different granularities; and scalability.

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Paper

Recognition of the Numbers in the Polish Language

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Abstract—Automatic Speech Recognition is one of the hottest research and application problems in today's ICT technologies. Huge progress in the development of the intelligent mobile systems needs an implementation of the new services, where users can communicate with devices by sending audio commands. Those systems must be additionally integrated with the highly distributed infrastructures such as computational and mobile clouds, Wireless Sensor Networks (WSNs), and many others. This paper presents the recent research results for the recognition of the separate words and words in short contexts (limited to the numbers) articulated in the Polish language. Compressed Sensing Theory (CST) is applied for the first time as a methodology of speech recognition. The effectiveness of the proposed methodology is justified in numerical tests for both separate words and short sentences.

Keywords—Automatic Speech Recognition, Compressed Sensing, Sparse Classification.

1. Introduction

Automatic Speech Recognition (ASR) can be defined as a real time computer-driven transcription process of a spoken language into a readable text. The main idea of ASR research is to define efficient speech recognition methodologies, implemented and automatically driven by computers, which work in a real time and allow to transform spoken language pronounced by native speakers. The accuracy of the speech recognition technology depends usually on the size of the implemented vocabulary, noise or speaker accent and individual pronunciation features. If the vocabulary size is sufficiently large and the system is properly trained to learn an individual speaker's voice, The recognition accuracy can exceed 90%. There are three most important features of the ASR systems: large vocabularies, continuous speech capability and speaker independence. Most of the speech recognition technologies has been initially designed for the visually disabled people as voice recognition support, which may be helpful for those patients, who suffer from eye diseases or some cerebral impairment.

In the recent years, thanks to intelligent and mobile technology, advanced speech recognition methods have been used by the users of mobile devices. Automatically created messages or documents from dictation – ASR technology makes their life easier. ASR has been also used for practical purposes and over the last several years it has been an area of great interest and activity to the signal processing. It seems that the perspective of development of ASR systems is optimistic – future applications of automatic speech recognition will contribute substantially to the quality of life of the society [1]–[3].

There are two main categories of speech recognition technologies: acoustic-phonetic approach, and pattern recognition-based approach. In the former, a continuous speech spectrum can be divided into several segments, which are defined as phonetic units (with unique labels), based on particular speech features. Speech stream is therefore defined as the continuous stream of such phonetic units. Then, the sequences of phonemic units are mapped into the sequences of words through a lexical decoding.

In the pattern recognition-based approach, the basic speech units are modeled by using acoustic lexical description of all words in the vocabulary. The acoustic-phonetic mapping is generated as a result of a finite discrete training process of a set of utterances. This mapping generates speech units which are defined based on the acoustic descriptions of linguistic units represented in the words occurring in the training set. This methodology seems to be the most effective speech recognition technique so far [1], [3]–[5].

Speech recognition problem is very complex for solving by using classical learning or pattern recognition methodologies. First, signal segmentation process generates a huge number of phonetic units, that must be processed. There are several methods of digital parametric representation of the speech signals, e.g., short-time spectral envelope method, Linear Predictive Coefficients (LPC) method, or Mel-Frequency Cepstral Coefficients (MFCCs) methodology, which can support the complex signal processing and classification. However, this support seems to be insufficient in the light of many types of the noise factors and individual speaker's articulation abilities, and differences between the spoken and written language or difficulties in the recognition of the words in the speech context [6].

Although speech recognition is a difficult classification task, Polish language morphology makes this problem even more challenging in the case of the Polish language, also in the simple cases of limitation of the analysis to the specific types of words such as numerals. Speech recognition applications used to recognize Polish languages usually include voice user interfaces such as voice dialing, appliance remote controlling, searching specific information, preparation of structured documents or speech-to-text processing. Algorithms which are used in those systems are based on several formal and heuristic learning-based or optimization methods, such as neural networks (usually as hybrid method with

Markov models), statistics (N-grams) and rule-based (Finite State Traducers) language models or dynamic Bayesian networks [4], [7]. Developing an effective model for the Polish language recognition can be useful for other recognition systems in which sounds are not yet classified [6].

The main objective of this work is to develop a speech recognition model based on the *Compressed Sensing* and *Sparse Classification* methods for the analysis and pattern recognition of Polish numerals, both as separated words and as words in a context. In this case Sliding Window technique [8] is used as support mechanism. All above mentioned methods have been previously known as effective tools for face recognition [9]. The developed technology allows to recognize speech of virtually any speaker. Its efficiency has been justified in series of numeral tests presented in the experimental work section.

This paper is structured as follows. In Section 2 the recent solutions for ASR with a special focus on the Polish languages approaches are surveyed. Details of the Automatic Speech Recognition general problem with a short explanation of the main steps in speech recognition process are defined. In Section 3 the authors defined the proposed methodology and implementation details. Section 4 presents the results of simple empirical analysis. The paper ends with conclusions and general future research remarks.

2. Automatic Speech Recognition

Speech recognition is a type of pattern recognition problem. The design of a pattern recognition system essentially can be realized in the following four steps [2]:

- data acquisition and preprocessing,
- data representation, necessary for pattern recognition,
- training, for instance imparting pattern class definition into the system, usually by showing a few typical examples of the pattern,
- decision-making process, generation of a pattern class by means of a training set of examples.

2.1. Problem Statement

The ASR process can be realized in the following three main steps [2]–[3], [10]:

- preliminary processing (pre-processing),
- parametric encoding,
- classification.

2.1.1. Pre-processing

The signal pre-processing stage is usually recorded and subjected to pre-treatment, such as the use of pre-emphasis filter and removal of the silence at the beginning and end of the recorded words. Then, the signal is divided into short time periods, called frames, by using of windowing method (such as Sliding Window technique). The preliminary processing stage precedes the process in which these features are isolated from the signal, and they are later used for classification. Pre-processing stage is crucial, because it usually has a great impact on the efficiency of the whole speech recognition system. The pre-processing phase flow is presented in Fig. 1.

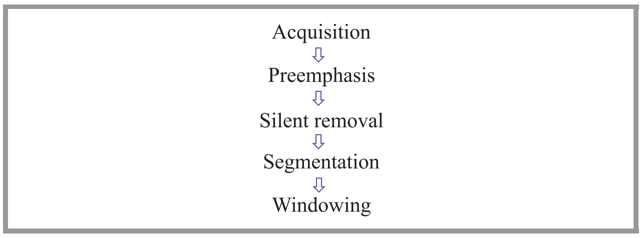

Fig. 1. Preliminary processing stage.

The first step in the pre-processing is an acquisition (recording the signal). The analog speech signal is usually recorded by a microphone and stored at the computer hard disc. However, before digital conversion, the strength of the signal may be improved by filtering through anti-aliasing filters. The analog-to-digital conversion processes is therefore defined as a sequence of sampling, quantization and signal encoding procedures.

After the acquisition, the signal strength is modulated by using a pre-emphasis filter. In the signal spectrum, the high frequency units are reinforced. Pre-emphasis is applied in order to reduce the phenomenon that the higher frequency has lower amplitude than the lower frequency (spectral tilt). In the process of segmentation the speech signal is divided into short segments called frames. Each frame contains specified number of samples. The frames can overlap. The typical length of the frame is 20–40 ms, which gives 320–640 samples at sampling 16,000 Hz [2].

In the segmentation process some gaps may occur in the processed signal. Those gaps may be the main reason of disruptions in Fourier transform (the additional harmonic components appear in the temporary spectrum). In order to cope with that, the value of each frame should be multiplied by a *window function*. It makes the time-frame at the end of frames more smooth. The most commonly used window function is *Hamming window mapping*, which can be defined as follows [3]:

$$w(n) = 0.54 - -0.46 \cdot cos\left(2 \cdot \pi \cdot \frac{n}{N-1}\right), \qquad (1)$$

where

$$0 \leq n \leq N-1\,. \qquad (2)$$

2.1.2. Parametric Encoding

In the parametrization stage the characteristic features are isolated from the signal. A *feature vector f* is defined for each frame created during the preliminary processing

stage f. The frame containing time course of the amplitude undergoes spectral analysis. This is usually difficult to obtain proper features of the frames which would not be dependent on the particular speaker. Indeed, the energy flow for the same word can be very different for various speakers.
There are two main methods of such frequency analysis: linear prediction model, and analysis based on Mel-frequency Cepstral Coefficients (MFCC) [11]. The authors used the second method in this work.

Mel-frequency Cepstral Coefficients (MFCC)

The purpose of the mel-cepstral analysis is to represent how the speech signal is considered by the human auditory system. First, the signal is transformed into the field of frequencies by using of the Fast Fourier Transformation (FFT). Then the signal is filtrated through the regular triangular filters in mel-scale, which reflects the nonlinearly perception of the frequencies (typical of human ear). Then, the discrete cosine transform is used to step into the cepstral field.
Mel-scale reflects, how human hear perceives the speech signal (how it reacts nonlinearly to the frequencies of the signal). At low frequencies (below 1 kHz) the changes are easier to detect than at higher frequency ranges. The higher the frequency, the lower the accuracy. The gaps between the series of ranges should be expanded in order to compensate for the nonlinearly. This may be achieved by using a proper set of filters for the successive frequency ranges. The filters are nonlinear according to the mel-scale [6].

2.1.3. Classification

The classification stage is defined as a comparison process of the signals' characteristics with respect to the recognition of characteristics stored in a classification local database, where a signal sample is assigned to a specific class. One of the simplest methods of classification algorithm is "k-nearest neighbors" methodology. Some other example can be sparse classification.

2.2. Efficiency Measures

There are various methods of the measurement of the efficiency of speech recognition techniques. The most popular metrics can be defined as follows [1].
Word Error Rate (WER) defined as

$$WER = 100\frac{S+D+I}{N}[\%]. \tag{3}$$

Word Accuracy (WAcc) defined as

$$WAcc = 100 - WER = 100\frac{N-(S+D+I)}{N}[\%], \tag{4}$$

where: S – denotes a number of substitutions, D – stands for the number of deletions, I – is the number of insertions, N – stands for the number of words to recognize.
WER defines how much the recognized series is different to the original one. It is based on the Levensthein distance (the degree of similarity of two sign chains) [3]. However, instead of letters words are used as the units to measure. What is interesting, is the fact WER can exceed 100%, so the value of WAcc can be less than zero.
The metrics defined above are based on the following recognition errors indicated for series of words in the context:

– *substitutions* – in the series of words the word is recognized as another word,

– *deletions* – the word appearing in the original series does not appear in the recognized series,

– *insertion* – the recognized series contains a new word between two words of the original series.

3. Implementation Details

In this section we define methods and algorithm we have developed for the implementation of our Polish numerals recognition system.

3.1. Database of Samples

We created a database of the 825 signal samples, which consists of wave type files containing numerals spoken by 80–83 speakers (depending on number there were 80 or more speakers). Experiments have been provided for each sample of from this database. To provide the independence of the speaker recognition, the samples were disposed which have already been uttered by the currently tested speaker. All tests except for comparing the effectiveness of various mitigation methods were carried out using the Least Angle Regression (LARS) Least Absolute Shrinkage and Selection Operator (LASSO) method.
The validation process is performed before classification. The results of the validation is usually a simple classification of the audio samples, which can be defined in the following way:

– properly validated samples – samples in which SCI rate exceeded the assumed threshold,

– correctly classified samples – samples that have been assigned to the correct class, excluding the validation,

– samples properly validated and classified,

– incorrectly validated samples – samples in which SCI did not exceed the assumed threshold, although it belongs to one of the classes from the database,

– samples that were incorrectly classified with the correct validation – samples that were incorrectly classified even though the validation proved positive,

– samples that were incorrectly classified and rejected with validation – samples, which in the absence of validation would be misclassified.

Table 1
Results of the recognition of sequences of the numerals

Number of iterations	30			50			70		
SCI	0.15	0.35	0.50	0.15	0.35	0.50	0.15	0.35	0.50
All samples	825	825	825	825	825	825	825	825	825
Properly validated samples	813	684	516	793	570	327	765	438	140
Correctly classified samples	767	767	767	777	777	777	777	777	777
Samples properly validated and classified	761	672	512	757	567	326	739	437	140
Incorrectly validated samples	12	141	309	32	255	498	59	387	140
Samples that were incorrectly classified with the correct validation	52	12	4	36	3	1	27	1	0
Incorrectly classified samples	58	58	58	48	48	48	48	48	48
Samples that were incorrectly classified and rejected with validation	6	46	54	12	45	47	21	47	48

Table 1 presents results of the recognition of sequences of the numerals – the results taking account of the SCI and the number of the iterations in which the LARS LASSO method was used.

3.2. Detection of the Initial and Terminal Elements in the Word

The isolated words are recognized and classified in the speech detection process. If the speech sequence is not divided into units it is hard to determine the boundary between the words. In both cases one should prepare the model samples to reflect the pure signal. The initial and final silence should be removed from the recording. The background noise makes it very difficult to determine the initial and final part of the word, because it disrupts the proper speech signal [7].
There are many algorithms for Voice Activity Detection. In the respective research the silence removing algorithm was used which makes use of signal energy, and the algorithm making use of zero crossing density. They can be used together. In both methods the first step is to divide the signal into equally long frames. Then for each frame the decision rule is made up in order to classify the segment as the speech signal or silence (background noise).

Signal energy algorithm. That particular algorithm is based on assumption that the speech signal energy in much higher than the background noise energy. For each frame the energy is calculated according to the formula [7]:

$$E_n = \sum_{n=1}^{N} [s(n)]^2 , \tag{5}$$

where N is the number of samples is the timeframe.

Then the result is compared to the threshold – if it is higher than the assumed value, the frame is classified as the part containing speech signal.

Zero crossing density algorithm. In that algorithm, the number of so called zero crossings of the signal is calculated for each frame, according to the [2]:

$$ZCR_n = \sum_{n=1}^{N} \left| sgn\left[x(n)\right] - sgn\left[x(n-1)\right] \right| , \tag{6}$$

where

$$sgm[x(n)] = \begin{cases} 1 & x(n) \geq 0 \\ -1 & x(n) < 0 \end{cases} . \tag{7}$$

One can assume, that if the zero crossing number is high the signal is background noise. Otherwise, it is speech.

3.3. Mel-filters Pool Used in the Implementation

For the purposes of the tests one use the pool of 40 FB-40 triangular filters which covers the frequency range [133, 6854] Hz at the assumed sampling frequency 16 kHz. Middle frequencies of the first 13 filters are distributed according to the linear relation (linear scale) in the range between [200, 1000] Hz in the interval of 66,67 Hz. Other 27 filters are distributed according to the logarithmic relation (logarithmic scale) in the range between [1071, 6400] with the step logStep = 1.0711703 [12]:

$$\text{logStep} = \exp\left(\frac{\ln\left(\frac{f_{c40}}{1000}\right)}{\text{numLogFilt}} \right) \tag{8}$$

where f_{c40} denotes the middle frequency of the last of logarithmic filters, numLogFilt is the numbers of logarithmic filters.

3.4. Compress Sensing Methodology – Main Concept

Compressed Sensing (or Compressive Sampling) is an emerging field that has attracted considerable research interest over the past few years and then has already become

a key concept in various areas of applied mathematics, computer science, and electrical engineering.
Compressed Sensing has recently emerged as an efficient technique for sampling a signal with fewer coefficients than the number dictated by classical Shannon/Nyquist theory. The assumption underlying this approach is that the signal to be sampled is sparse or at least compressible, i.e., it must have a concise representation in a convenient basis. In Compressed Sensing, sampling is performed by taking a number of linear projections of the signal onto pseudorandom sequences. Therefore, the acquisition presents appealing properties such as low encoding complexity, since the basis in which the signal is sparse does not need to be computed, and universality, since the sensing is blind to the source distribution. Reconstruction of a signal from its projections can be done e.g. using linear programming, with a complexity that is $O(N^3)$, with N the number of samples to be recovered [13]–[15].

3.5. Implementation-Compressed Sensing Used in the Process of Classification

The classification method by means of compressed sensing is called sparse classification. The idea is to represent the feature vector of the examined sample as the linear combination of the feature vectors of relatively few samples taken from the teaching set. Then, the quality of that representation for each class is examined.
Because the samples can have various length but they are divided into frames of equal length, feature vectors can be given various length (and they are usually are given). In sparse classification we need feature vectors of equal length. In the implementation we used a method of changing the vector length (normalization) based on the linear interpolation.

3.5.1. Sparse Representation

Sparse representations are representations that account for most or all information of a signal with a linear combination of a small number of elementary signals.
The examined sample y can be represented as the linear combination of the samples taken from the teaching set $d_{i,n}$ where the first index $(1 \leq i \leq l)$ stands for one of the I classes, whereas the second $((1 \leq n \leq N_i)$ refers to the particular sample taken from and i class which consists of N_i samples [16]–[18]:

$$y = \sum_{i=1}^{1} \sum_{n=1}^{N} \alpha_{i,n} d_{i,n} \,. \tag{9}$$

As each sample is represented by k-elements feature vector, the base can be represented as the matrix.

3.5.2. Il-minimization

Usually, the number of teaching samples is much more numerous than the length of the feature vector for the single sample. In other words, it has more unknowns than equations [13]. Such a system of equations, called undetermined, has infinite number of solutions. We assumed, however, that x is a sparse vector (it contains mainly zero elements). Therefore we choose only the most "sparse" results from the all available, according to the compressed sensing theory. Because minimalization of such a solution in NP-complete problem we stick to the minimalization in the norm l^1. In other words, we strive for making the sum of absolute values of the x vector as small as possible [19]. In the perfect case, we achieve a vector which have nonzero components at positions pertaining to only one class. There are many methods how to find the sparse result of the set of equations. In our experiment, the so called LASSO method was used (Least Absolute Shrinkage and Selection Operator).

3.5.3. LASSO Solutions

LASSO solutions can be calculated by standard numerical algorithms. In the tests the method Least Angle Regression (LARS) was used with the modification that allows for solving the problem of the recognition of Polish numerals.
LARS algorithm is a modification of Forward Stepwise Regression [9], [17]. One single modification of the enables solving the LASSO problem. If the non-zero coefficient exceeds zero, the predictor is removed from the set of predictors and the direction is calculated once again. LARS method seek for the solution iteratively. Each step it adds one value to the vector of solutions. Therefore, after k-steps only k-elements of the resulting vector x are non-zero.
Before applying the LASSO, the y vector and every matrix A column are normalized. Each element of the y vector is divided by the vector length:

$$y_{norm} = \frac{y}{||y||} = \frac{y}{\sqrt{y_1^2 + \ldots + y_N^2}} \,, \tag{10}$$

where N stands for the number of elements of the vector.
LASSO is useful for us to solve the linear regression problem, yet it restricts the solution – the sum of absolute values of the resulting vector must be less than the given value.
The model of the linear regression is determined as [8]:

$$y_i = a_i x_{i1} + \ldots + a_n x_{in} + \varepsilon_i = A x_i + \varepsilon_i, \quad i = 1, \ldots, k \,. \tag{11}$$

Vector of the prediction is defined as

$$\hat{\mu} = \sum_{j=1}^{n} a_j \hat{x}_j = A\hat{x}, \tag{12}$$

and the square error of the prediction is defined as

$$S(\hat{x}) = \left\| y - \hat{\mu} \right\| = \sum_{i=1}^{k} (y_i - \hat{\mu}_i)^2 \,. \tag{13}$$

May $T(\hat{x})$ be the L_1 of the vector $(\hat{x})$

$$T(\hat{x}) = \sum_{j=1}^{n} \left| \hat{x}_j \right| \tag{14}$$

The result of the LASSO method is the vector $(\hat{x})$, which minimizes the square error $S(x)$ with respect to the condition on $T(\hat{x})$:

$$minimizeS(\hat{x})forT(\hat{x}) \leq t\,. \tag{15}$$

If the value of the parameter t is high, it does not affect the above-mentioned restriction. But for the smaller values of t most of x_i coefficients have mostly zero value (their number is affected by t parameter). Therefore, sparse vector x being the solution of the set of equations is obtained. We will make use of that vector during the subsequent stages (validation and classification).

3.5.4. Classification

Having already sparse solution of equations, can be properly classified. For each class, a check is made as to its exact samples can play test sample. This error is called the residue. To retrieve to the sample with copies of only one class is defined function $\delta_i(x)$, which leaves unchanged the elements vector x associated with the class, and resets the remaining. The reconstituted vector can be defined as follows:

$$y_{i^R} = A\delta_i(x)\,, \tag{16}$$

where reproduction error is the difference between the test vector y and the vector reconstituted y_{i^R} in standard L^2.

3.5.5. Classification of Continuous Speech

Classification of continuous speech is a task much harder than the classification of the isolated words. In conducted tests, this would require taking a break while making each part number, which would be an unnatural phenomenon for the speaker. In order to identify all the spoken numbers without having to take breaks, we apply a sliding window method, called sliding window [8], while the same classification we use sparse classification method, which was presented in the previous parts of this work.
The sliding window technique consists in the fact that the speech feature vector of the test select fragment, called a window, comprising the features specified amount of time frames and the this fragment is subject to classification. At the beginning of the first T is selected frames of speech, then the window is moved to the end of the vector, taking into account the specific step Δ. It is important that the coefficients T and have a significant impact on the effectiveness of recognition. Longer step reduces the required amount of computation, while reducing effectiveness. The classification for each window, based on the values of SCI and residuum. It is easy to calculate that the number of windows, and thus the maximum number of classification is W = I – T + 1. In practice, the number of classifications is much smaller, because not all windows are validated.
For samples of each word conduct identical process as in the case of the test sample. Window sizes in both cases must be the same, while a step shift can vary, e.g., to reduce the number of copies of the training we can choose to move more than one frame.
To sum up: each part of the number is treated as a single word. A word can be spoken by many speakers. Each word consists of a number of copies produced in the process of moving the window on the vector characteristics of speech.

3.6. Validation and Sparsity Concentration Index

Validation is the process which allows for checking if the given voice sample is the proper sample from the base. Validation precedes the classification process, as only validated samples undergo classification. The SCI coefficient is applied to validate samples. Sparsity Concentration Index stands for the level of concentration of the non-zero coefficients of the x vector pertaining to one class. SCI = 1 means, that the tested sample is represented only by items from one class. SCI = 0 means, that the non-zero coefficients are set evenly for all classes.
In the experiments the threshold was adopted $\tau \in (0,1)$, which determines whether a sample is classified as unrecognized. If $\mathrm{SCI}(x) < \tau$, the sample must be rejected, otherwise it should be classified [8].

4. Empirical Analysis

Testing methodology – isolated numbers. All test programs were implemented in the C++ language. The base of samples consisted of wave files containing the numbers spoken by speakers, which gives the total number of 825 files. The tests were conducted for each sample from the test base by means of cross-validation. During the recognition of the given sample from the test base, all the samples formerly recorded by the tested speaker were removed in order to ensure, that the recognition level is not dependent on the particular speaker. All the test, except the comparison of the efficiency of various minimalization methods, were conducted by virtue of LARS-LASSO method.

4.1. Testing Methodology – Series of Numbers

We tested series of numbers (0-9) spoken by 16 speakers. The test base was the same as for the isolated numbers, but with some additional samples reflecting background noise (silence), cut from the series of numbers. It turned out that the following parameters affected the recognition efficiency:

- window length,
- shifting the window for the samples for recognition,
- shifting the window for the model samples,
- number of iterations for the LARS LASSO method,
- SCI threshold enabling for determining if the given window should be ascribed to a class or if it should be removed as a gap between words (classes).

4.2. Experimental Results

The results according to the SCI threshold and the number of LARS Lasso method iterations. All the samples used in that test were proper samples taken from the base. The feature vector length was 35 frames. According to the conducted tests, SCI threshold = 0.35 at 70 iterations is enough for the validated sample to be correctly recognized (at least, it is highly probable). However, the higher is the threshold, the more severe are the validation criteria. As a result, the number of correct samples which are removed increases (Fig. 2).

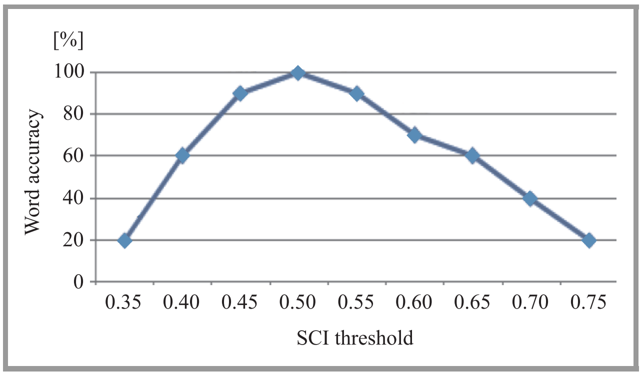

Fig. 2. The effect of SCI threshold on the efficiency of recognition.

4.2.1. Comparison of Sparse Classification to the *k*-nearest Neighbors

Figure 3 presents the percentage of the correctly and incorrectly recognized samples for two classification methods, namely Sparse Classification – minimization by virtue of the LASSO method at max. iteration number = 50, and k-nearest neighbors – for $k = 1$ or seeking only one nearest neighbor [20]. In both methods the feature vectors had the same length (35 frames). The classification was preceded by validation.

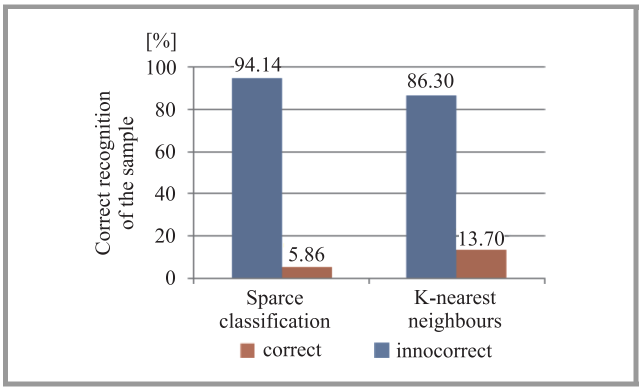

Fig. 3. The comparison of the various classification methods.

Sparse Classification method turned out to be more efficient than the standard k-nearest neighbors algorithm. It is because in the SC the model samples are considered more globally, whereas in the k-nearest neighbor method single incorrect sample from the base can lead to recognition error. The LARS LASSO method provides us with much better results, then. But there is also additional advantage of that algorithm. It is the time factor, which is so crucial in speech recognition.

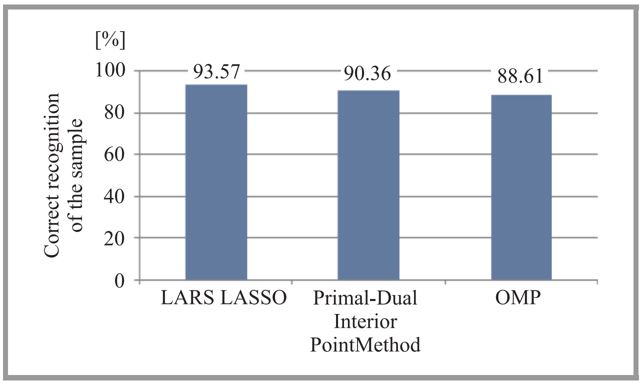

Fig. 4. The comparison of the efficiency of the various minimization.

Computation by means of LARS LASSO method, at the given parameters, took 0.14 s. OMP algorithm is faster (0.11 s) but its efficiency is much worse (Fig. 4). For the available Primal-Dual Interior Point Method it took terrible 11.3 s, despite the fact, we significantly shortened the feature vector.

4.2.2. Results of the Recognition of Sequences of the Numerals

The recognition of sequences of the numerals experiments have been conducted under the set of the key parameters presented in Table 2.

Table 2
Key parameters for the recognition of sequences of the numerals

Parameter	Value
The window size	45
Move the main window	2
Move the window of the samples	5
The maximum number of iterations of LARS LASSO	50
Threshold of SCI	0.5

Detailed results of the recognition os sequences of the numerals are presented in Table 3.

The technique of “sliding window” in conjunction with the method of classification “Sparse Classification” and validation based on the ratio of SCI, gives good results for the recognition of strings of digits.

Table 3
Results of the recognition of sequences of the numerals

Filename	WAcc [%]	Recognized string
Adrian_G_80514_22_10.wav	80	0 1 2 3 4 9 6 7 4 8 9
Adrian_S_80577_21_0_10.wav	90	0 1 2 3 4 5 6 7 8 4 9
Krzysztof_U_80582_22_0_10.wav	80	0 1 2 3 4 5 6 7 8 5 9 5
Lukasz_K_80534_21_0_10.wav	80	0 1 2 3 4 6 9 5 6 7 8 9
Maciek_M_80541_21_0_10.wav	90	0 1 2 3 4 5 6 7 8 9 5
Marcin_K_80523_21_0_10.wav	80	0 2 3 4 9 6 7 8 9
Marek_W_80589_21_0_10.wav	70	0 2 4 5 6 7 8 9 5
Michal_J_80519_21_0_10.wav	100	0 1 2 3 4 5 6 7 8 9
Michal_M_82432_23_0_10.wav	100	0 1 2 3 4 5 6 7 8 9
Michal_P_80558_22_0_10.wav	80	0 1 2 3 4 5 6 7 1 8 9 5
Pawel_M_80545_22_0_10.wav	80	0 1 3 4 5 6 5 8 9
Piotr_B_80497_22_0_10.wav	80	0 1 2 3 4 9 5 6 7 7 8 9
Radoslaw_S_80567_22_0_10.wav	70	0 4 2 4 4 9 6 7 8 9
Witold_P_80556_22_0_10.wav	100	0 1 2 3 4 5 6 7 8 9
Witold_U_80583_22_0_10.wav	70	0 2 3 4 6 7 8 5
Wojciech_Z_80602_21_0_10.wav	70	0 1 2 4 6 6 7 8
Average:	83	

5. Conclusions

According to the experiment results, the Compressed Sensing and Sparse Classification methods are very efficient in recognition of the isolated words and can find multiple applications f.ex. in recognizing voice commands in remotely controlled appliances.

Moreover, the SCI coefficient-based validation is quite effective. The application of the above-described validation method and Sparse Classification in the voice identification systems is also worth the separate research. One can also examine so called Sparse Imputation technique enabling for the recognition of words in a very noisy background. It is crucial, how to prepare the pool of model samples. The number of good quality speech samples should be vast. According to the experiment, the more model samples in the base, the more efficient is the recognition system.

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Paper

Evaluation of the Energy Harvestable from an Airless Tire Equipped with Piezoelectric Bimorphs on the Lamellar Spokes

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Abstract—In this work, one evaluates the electrical power generated by an airless tire equipped with piezoelectric bimorphs on both lateral surfaces of the radially distributed lamellar spokes. Such sheet-like spokes are hinged both toward the wheel drum at the inner annular band, and toward the wheel tread at the outer annular band. Since the hinged spokes are able to transmit tension forces but unable to transmit compression forces, bending and buckling of the spokes occur in the region of contact between the tire and the road. Models for the rolling friction of the airless tire, for the bending and buckling deformation of the spokes, and for the electrical power generated by the airless tire are suggested. Variation of the curvature radii and bending deformations for the spokes in the region of contact with the road are illustrated for various values of the rolling friction coefficient and spoke length. Then, variation of the generated electrical power versus the length of contact is obtained for various travel speeds of the vehicle. One observes that the generated electrical power increases at augmentation of the rolling friction coefficient, spoke length and travel speed. Although the obtained electrical power for the proposed harvesting system is relatively modest, it is not depending on the road roughness, i.e. harvesting becomes possible even on smooth roads, such as highway surfaces.

Keywords—airless tire, bending and buckling, energy harvesting, generated electrical power, piezoelectric bimorph, spoke.

1. Introduction

Wheels are considered as a major source of excitation in motor vehicles [1], the excitation frequency increasing with the vehicle speed and decreasing with the wavelength of the road roughness. While a vehicle is resting or running, its supporting tires receive on their hubs (wheels) a load proportional to the gross mass of the vehicle, and then tires transfer the load to the ground (road, railway, etc.) as a contact pressure [2]. Tires should endure such contact pressure, to provide for a buffering function, i.e., to absorb the energy of shock and vibrations produced by the rough road, and also to maintain a proper quasi-circular shape. Pneumatic tires utilize the internal inflation pressure as mean of carrying the load. Such tire is inflated through a valve, which is positioned either on an inner elastic tube, or on a rim in the case of a "tubeless" type (Fig. 1). The most serious problem of air-filled tires is the puncturing, which leads to a flat tire or even to an explosion at high traveling speeds. This might cause severe traffic accidents.

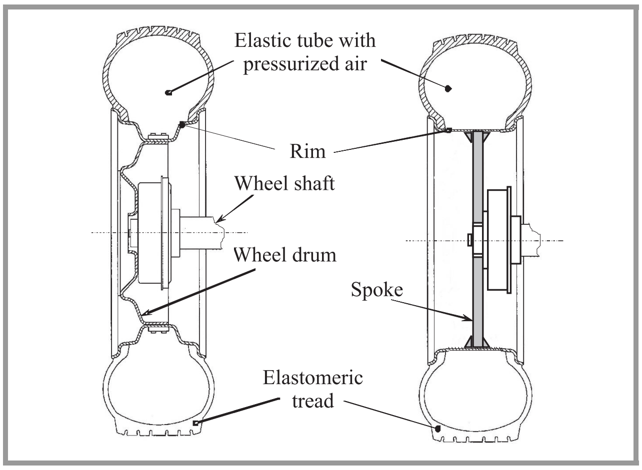

Fig. 1. Schematic view of a classical pneumatic tire with the rim supported by a wheel drum (left side) or radially distributed spokes (right side).

Standard pneumatic tires [3], [4] are also susceptible to depressurization due to the air leakage at the valve as well as around of the wheel rim, and due to the oxygen absorption into the elastomeric part. Loss of pressure causes the tire to flatten in the area where the load is applied, producing a larger contact area of the tire with the ground with every revolution. This leads to increased friction and wear at the contact region with the road, i.e., to larger loss of energy. On the other hand, increased bending and buckling deformations in the contact region produces temperature augmentation and consequently, quicker degradation of the tire.

In order to avoid the disadvantages of standard pneumatic tires (puncturing and depressurization), non-pneumatic or airless tires were developed [3]–[6]. A non-pneumatic or airless tire does not utilize the internal inflation pressure to carry the load. Instead, load is supported by solid structural elements (tread, annular bands, spokes, etc.). Some of these elements are working in compression and the other in tension. Principles for carrying the load include the column, arch, and buttress for distribution of load which have been largely used for building cathedrals, monuments, roads, bridges, etc.

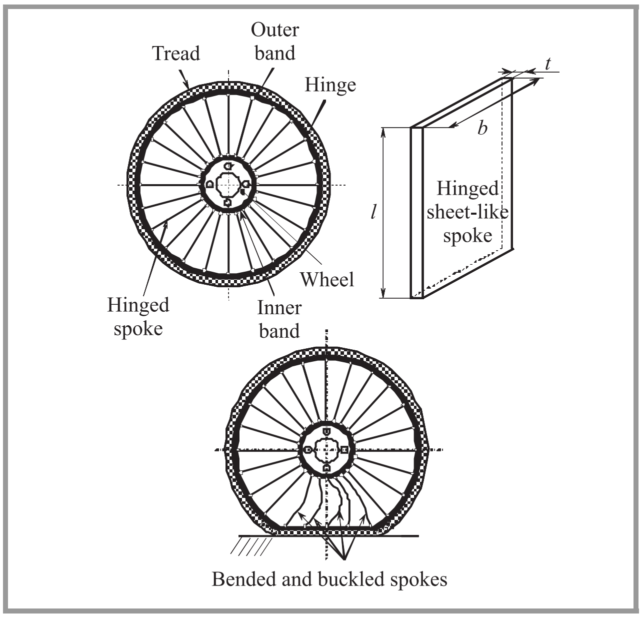

Fig. 2. Structure of an airless tire (upper left side), geometry of the hinged sheet-like spokes (upper right side), and bending or buckling deformation of the spokes on the contact surface with the road (lower part).

A non-pneumatic tire is usually made of five parts bonded together (see the upper left side of Fig. 2): the hub or wheel having an axis of rotation, a spoke section, an annular outer band that surrounds the spoke section, an elastomeric tread portion that surrounds the outer band and contacts the ground, and a mounting inner band at the radial inner end of the web spokes [5]. The mounting band anchors the tire to the hub or wheel. Elastomeric tread has an annular shape and a thickness of about 10 mm. It integrally covers the annular band and has a grooved pattern on the outer circumferential face similar to that of a classical pneumatic tire. Sometimes a side-cover (not illustrated in Fig. 2) is used to keep the tire spokes from filling with mud and dirt thereby maintaining wheel balance and ride smoothness. Usually the lamellar or sheet-like spokes (upper right side of the Fig. 2), uniformly distributed in the radial direction are employed; they may be axially aligned, or may be oblique to the tire axis. In order to obtain an almost uniform pressure on the contact surface with the road, sheet-like spokes are hinged both toward the wheel drum at the inner annular band, and toward the wheel tread at the outer annular band. Since the hinged spokes are able to transmit tension forces but unable to transmit compression forces [5], [6], bending and buckling of the spokes occur in the region of contact between the tire and the road (see the lower part of the Fig. 2).

In this work, the electrical power generated by an airless tire equipped with piezoelectric bimorphs of thickness $H = 0.43$ mm is evaluated. Bimorphs are bonded on both lateral surfaces of the lamellar spokes (Fig. 3). Electrical

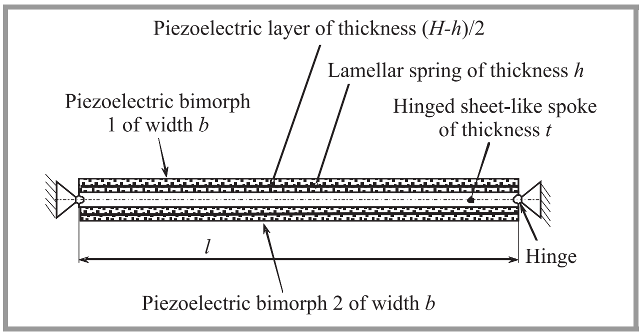

Fig. 3. Structure and dimensions of an energy harvesting spoke coated with piezoelectric bimorphs.

power is obtained due to the cyclical bending or buckling of the piezoelectric bimorphs together with the associated spoke in the region of contact with the road [7]. Such piezoelectric bimorph [8]–[10] consists of a lamellar metallic spring of thickness $h = 0.1$ mm whose both lateral surfaces are coated by a piezoelectric ceramic material, as for example, M1 piezoelectric ceramic supplied by Fuji Ceramics Ltd. [11]. First, in this theoretical investigation, models for the rolling friction of the airless tire, for the bending and buckling deformation of the spokes, and for the electrical power generated by the airless tire are suggested. Then, variation of the curvature radii and bending deformations of the spokes, on the contact surface with the road, is investigated for various values of the rolling friction coefficient and spoke length. In the end, variation of the generated electrical power versus the length of contact is investigated for various travel speeds of the vehicle.

2. Model of the Rolling Friction for an Airless Tire

Figure 4 illustrates the behavior of a tire with a rotational direction, the rolling being represented in idealized form on a flat underlying surface, on which a flattened contact area is produced. Length $2C$ of the contact area is measured from the entry into the contact region up to the exit from contact area; it depends on the tire characteristics, on the wheel load, the tire pressure, the driving state (longitudinal and/or lateral forces), the travel speed and the road status (dry, wet, etc.). Thus, the tire has a bending deformation from a quasi-circular shape to a flattened shape at the entry into the contact area, and a bending deformation from a flattened shape to a quasi-circular shape at the exit from the contact area. At entry into and exit from the flat contact with the ground, the centrifugal acceleration suddenly changes, and the repeatedly flexed tread increases its temperature.

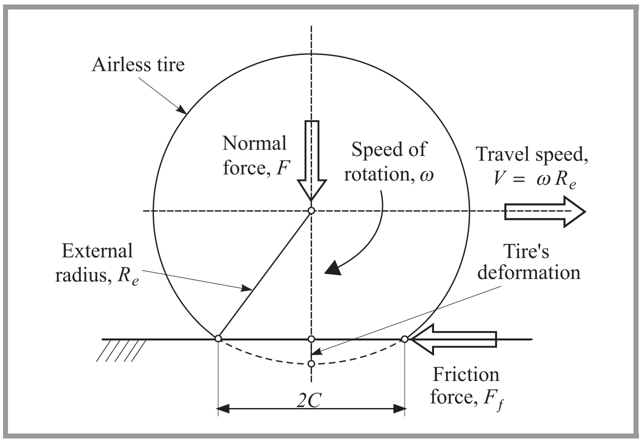

Fig. 4. Model of rolling friction for an airless tire.

Rolling friction coefficient μ_r of an airless tire is defined as the ratio of the friction force F_f and the normal force F_n (Fig. 4):

$$\mu_r = \frac{F_f}{F} = \frac{C}{\sqrt{R_e^2 - C^2}} = \frac{\overline{C}}{\sqrt{1-\overline{C}^2}}\,; \quad \overline{C} = \frac{C}{R_e}\,. \quad (1)$$

Here, the dimensionless contact length $\overline{C} = \frac{C}{R_e}$ is taken as the ratio of the contact length $2C$ to external diameter $2R_e$ of the airless tire. One observes that $\mu_r \cong \overline{C}$ for usual values of the dimensionless contact length ($\overline{C} = 0.2 - 0.4$).

3. Model for the Bending or Buckling Deformation of the Spoke of an Airless Tire

As Fig. 5 illustrates, the lamellar spoke of length $l = R_e - R_i$ occurs as planar in the upper part of the tire, between the entry into and the exit from the flat contact region with the ground ($x = \pm C$). Since the hinged spokes are able to transmit tension forces in the upper part of the tire, but unable to transmit compression forces in the region of contact between the tire and the road, the following assumption can be made concerning the bending or buckling deformation of the spokes: length of the deformed (curved) spoke equals the length $l = R_e(1 - \overline{R}_i)$ of the spoke before bending or bucking, as shown on lower part in Fig. 5.

In such circumstances, variation of the spoke dimensionless deformation $\overline{\delta}$ versus the dimensionless coordinate $\overline{x}$ taken along the contact region can be expressed by the following relationship:

$$\overline{\delta} = \frac{\delta}{R_e} = \overline{\rho} - \sqrt{\overline{\rho}^2 - 0.25\left(\sqrt{1-\overline{C}^2 + \overline{x}^2} - \overline{R}_i\right)^2}\,. \quad (2)$$

Here, the dimensionless curvature radius $\overline{\rho}$ of the spoke, the dimensionless coordinate $\overline{x}$ and the dimensionless radius of the wheel drum are defined as follows:

$$\overline{\rho} = \frac{\rho}{R_e}\,; \quad \overline{x} = \frac{x}{R_e}\,; \quad \overline{R}_i = \frac{R_i}{R_e}\,. \quad (3)$$

Dimensionless curvature radius $\overline{\rho}$ of the spoke is computed by numerically solving the following equation, derived on the basis of the above-mentioned assumption concerning the length of the bended spoke:

$$\sin\frac{1-\overline{R}_i}{2\overline{\rho}} = \frac{\sqrt{1-\overline{C}^2 + \overline{x}^2} - \overline{R}_i}{2\overline{\rho}}\,. \quad (4)$$

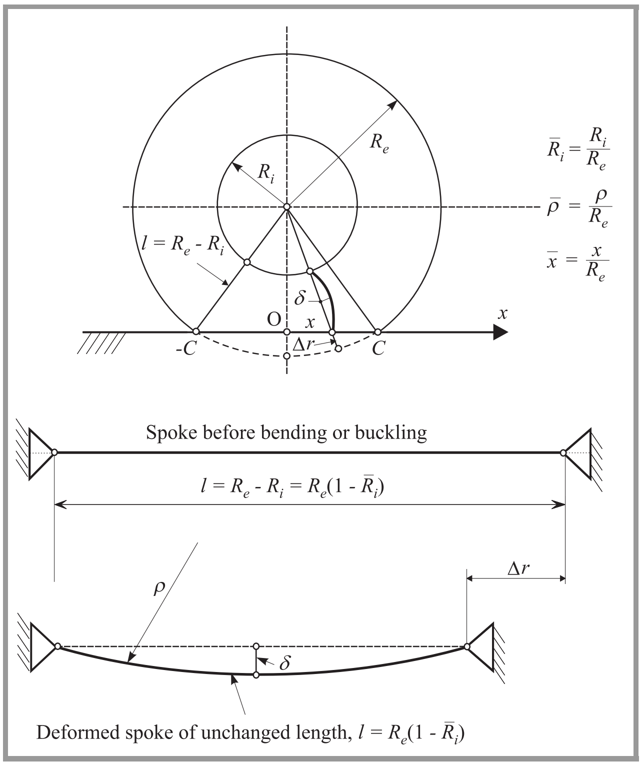

Fig. 5. Model for the bending or buckling deformation of the spoke in the contact region of the airless tire with the ground.

4. Model of the Electrical Energy Generated by Lamellar Spokes Coated by Piezoelectric Bimorphs

For a four wheels vehicle equipped with airless tires employing lamellar spokes coated by piezoelectric bimorphs, as already illustrated in Figs. 2 and 3, the generated electrical power can be calculated as follows [11], [12]:

$$P = \frac{9}{8\pi^2} k_{31}^2 \Gamma \frac{1-\overline{R}_i}{1+\frac{h}{H}} nbHV^3 J\,; \quad J = \int_0^{\overline{C}} \overline{\delta}^2(\overline{x})d\overline{x}\,. \quad (5)$$

Here, V is the traveling velocity of the vehicle, n is the number of spokes included into the web connecting the inner and outer annular bands of the airless tire, Γ is the density of the ceramic material of the employed piezoelectric bimorph, k_{31} is the global electromechanical constant of the piezoelectric bimorph, and the integral J represents the mean potential energy of deformation of the bended spoke. The generated electrical power varies proportionally with the density of the piezoelectric ceramic material, the number of spokes, the spoke length $\frac{l}{R_e} = 1 - \overline{R}_i$ expressed in dimensionless form, the mean potential energy of deformation of the bended spoke J, the width b of the spoke, and the thickness H of the bimorphs. On the other hand, the generated electrical energy is proportional to the second power of the global electromechanical constant of the piezoelectric bimorph, and to the third power of the traveling velocity. In conclusion, usage of piezoelectric ceramic materials with high values for density and electromechanical constant is desirable. Additionally, the generated electrical power maximizes for nil thickness of the lamellar spring of the piezoelectric bimorph ($h = 0$), which means that the piezoelectric ceramic material should be directly applied on the lamellar spoke.

5. Numerical Results and Discussions

Figure 6 illustrates the variation of the dimensionless curvature radius $\overline{\rho}$ of the spoke (upper part) and the dimensionless bending deformation $\overline{\delta}$ of the spoke (lower part) versus the dimensionless coordinate $\overline{x}$ for the longest spoke

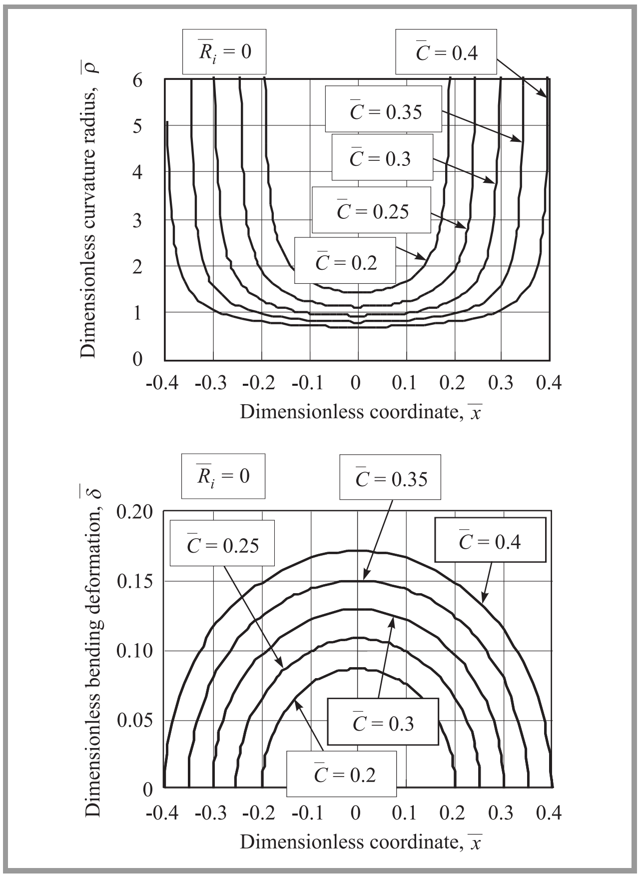

Fig. 6. Variation of the dimensionless curvature radius (upper part) and the dimensionless bending or buckling deformation (lower part) of the spoke versus the dimensionless coordinate along the contact region, for the longest spoke and various values of the rolling friction coefficient.

($R_i = 0$) and for various values of the dimensionless contact length $\overline{C} = 0.2$, 0.25, 0.3, 0.35 and 0.4. Additionally, Fig. 7 presents the variation of the dimensionless curvature radius $\overline{\rho}$ of the spoke (upper part) and the dimensionless bending deformation $\overline{\delta}$ of the spoke (lower part) versus the dimensionless coordinate $\overline{x}$ for the most probable dimensionless length of the contact region $\overline{C} = 0.3$, i.e., the most probable rolling friction coefficient μ_r of about 0.3, and various dimensionless radii of the wheel drum: $\overline{R}_i = 0$, 0.2, 0.4, 0.6 and 0.8.

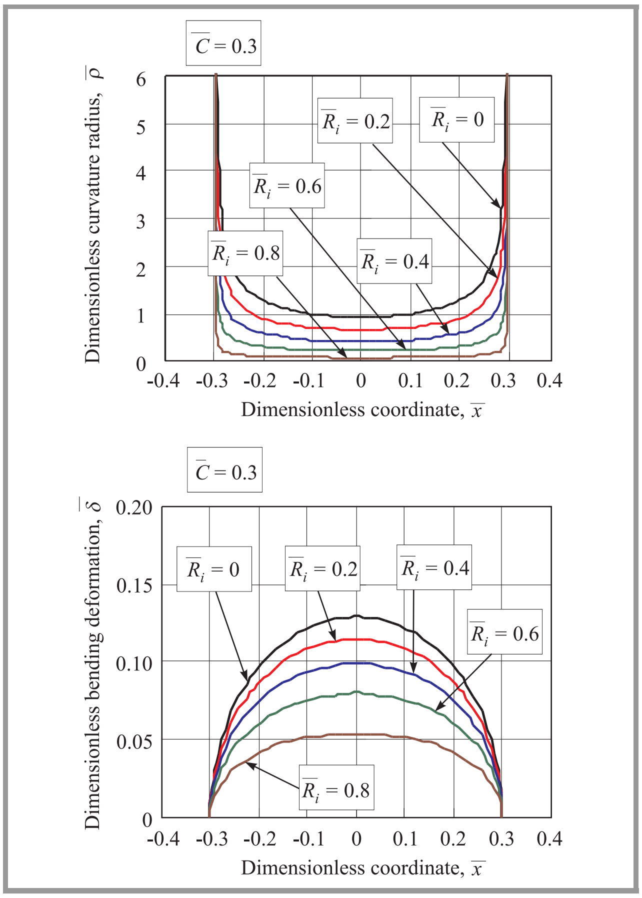

Fig. 7. Variation of the dimensionless curvature radius (upper part) and the dimensionless bending or buckling deformation (lower part) of the spoke versus the dimensionless coordinate along the contact region, for the most probable value of the rolling friction coefficient and various values of the dimensionless radius of the wheel drum.

From Figs. 6 and 7 one observes that at the entry into and the exit from the flat contact region with the ground ($x = \pm C$) the curvature radius of the spoke tends to infinity and the bending or buckling deformation of the spoke tends to zero. This can be explained by the fact that at these locations ($x = \pm C$) the curved spoke regains its planar shape, see the lower part of the Fig. 2. Additionally, at augmentation of the rolling friction coefficient or length of the contact region (Fig. 6), as well as at augmentation of the length of the spoke (Fig. 7), the maximum value of the bending deformation and also the variation rate of the bending deformation increases. Since the electrical power generation varies proportionally to both the maximum value and the variation rate of the bending deformation, one expects also increased efficiency of the proposed harvesting system.

Based on the results presented in the lower part of Figs. 6 and 7, one calculates the integral J, i.e. the mean potential energy of deformation of the bended spoke (Table 1).

Table 1
Variation of the integral J (mean potential energy of deformation of the bended spoke) for various values of the dimensionless length of contact and radii of the wheel drum

$J \cdot 10^{-4}$	$\overline{R}_i$				
$\overline{C}$	0.0	0.2	0.4	0.6	0.8
0.20	9.98	7.96	5.94	3.92	1.90
0.25	19.45	15.34	11.52	7.57	3.61
0.30	33.52	26.65	19.76	12.88	5.99
0.35	53.08	42.08	31.08	20.06	9.06
0.40	78.96	62.40	45.83	29.26	10.93

Next, Fig. 8 illustrates the variation of the generated electrical power versus the dimensionless contact length for various values of the traveling speed $V = 20$, 40, 60, 80 and 100 km/h, for a total number of spokes $n = 24$, for a width of the spoke $b = 0.1$ m, for a M1 piezoelectric ceramic material, supplied by Fuji Ceramics Ltd. [11], having a density equal to $\Gamma = 7{,}600$ kg/m^3, and a global electromechanical coefficient equal to $k_{31} = 6.5$, and for the most probable values of the dimensionless radii of the wheel drum, i.e., $\overline{R} = 0.2$ (upper part of the Fig. 8) and $\overline{R} = 0.4$ (lower part of the Fig. 8). Numerical results presented in Fig. 8 confirm that the generated electrical power increases at the augmentation of the spoke length and traveling speed of the vehicle. Although the obtained electrical power for the proposed harvesting system is relatively modest (a few tens of watts), it is not depending on the roughness of the road, i.e., harvesting becomes possible even on smooth traveling roads, such as highway surfaces.

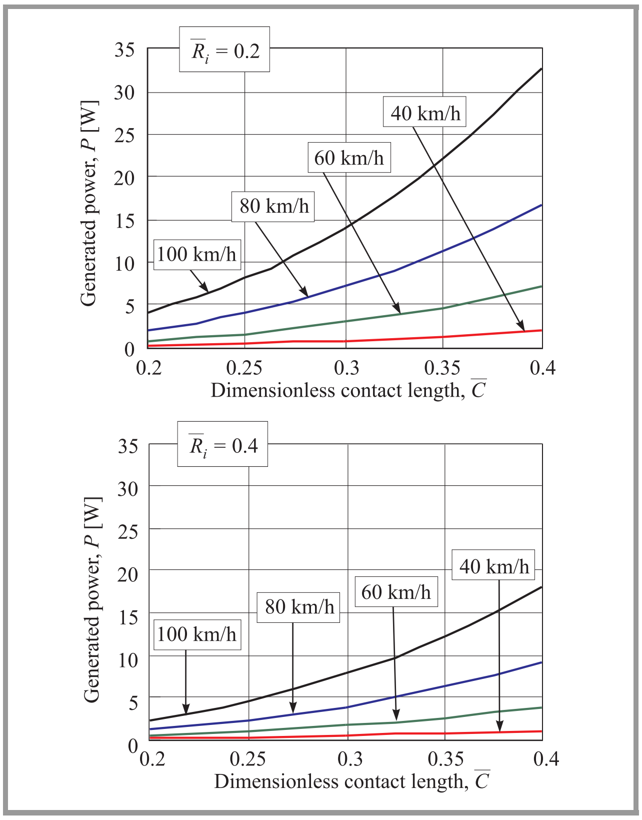

Fig. 8. Variation of the generated electrical power versus the dimensionless contact length for various traveling velocities of the vehicle and the most probable values of the dimensionless radii of the wheel drum, i.e. 0.2 (upper part) and 0.4 (lower part).

6. Conclusion

In this work, the electrical power generated by an airless tire equipped with piezoelectric bimorphs on both lateral surfaces of the radially distributed lamellar spokes was evaluated. Models for the rolling friction of the airless tire, for the bending and buckling deformation of the spokes, and for the electrical power generated by the airless tire were suggested. Variation of the curvature radii and bending deformations for the spokes in the region of contact with the road were illustrated for various values of the rolling friction coefficient and spoke length. Then, variation of the generated electrical power versus the length of contact was obtained for various travel speeds of the vehicle. Based on the theoretical analysis performed and the numerical results obtained, the following conclusions were inferred:

- Generated electrical power varies proportionally with the density of the piezoelectric ceramic material, the number of spokes, the mean potential energy of deformation of the bended spoke, the length and width of the spokes, and the thickness of the piezoelectric bimorphs bonded on the lateral surfaces of the spokes.
- Electrical energy is proportional to the second power of the global electromechanical constant of the piezoelectric bimorph, and to the third power of the traveling velocity of the vehicle. Thus, usage of piezoelectric ceramic materials with high values for density and electromechanical constant is desirable.
- Electrical power maximizes for nil thickness of the lamellar spring of the piezoelectric bimorph, which means that the piezoelectric ceramic material should be directly applied on the lamellar spoke.
- Electrical power increases at augmentation of the rolling friction coefficient, i.e., at augmentation of the length of the contact region between the airless tire and the ground. This behavior was explained by the increased maximum value of the bending deformation and also by the increased variation rate of the bending deformation in the contact region.

- Although the obtained electrical power for the proposed harvesting system is relatively modest (a few tens of watts), it is not depending on the roughness of the road, i.e., harvesting becomes possible even on smooth roads, such as highway surfaces.

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Paper

Application of High-Performance Techniques for Solving Linear Systems of Algebraic Equations

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Abstract—Solving many problems in mechanics, engineering, medicine and other (e.g., diffusion tensor magnetic resonance imaging or finite element modeling) requires the efficient solving of algebraic equations. In many cases, such systems are very complex with a large number of linear equations, which are symmetric positive-defined (SPD). This paper is focused on improving the computational efficiency of the solvers dedicated for the linear systems based on incomplete and noisy SPD matrices by using preconditioning technique – Incomplete Cholesky Factorization, and modern set of processor instructions – Advanced Vector Extension. Application of these techniques allows to fairly reduce the computational time, number of iterations of conventional algorithms and improve the speed of calculation.

Keywords—Advanced Vector Extension, conjugate gradient method, incomplete Cholesky factorization, preconditioning, vector registers.

1. Introduction

Solving linear systems of algebraic equations is a problem of linear algebra, which is common in many fields of science. The basic techniques and methodologies investigates for solving such a problem can be classified into two main categories, namely direct and iterative linear system solvers. Direct methods need computationally efficient resources (e.g., large RAM memory and fast CPU), which results in an inability to achieve a proper solution in a reasonable time [1]. For most types of the engineering problems modeled by the linear systems such methodologies are able to generate the exact ideal solutions.

Iterative methods are based on approximation of the exact solutions. In each iteration, the best achieved partial results may be improved by the implemented local optimizers. Final solution vector is generated as a results of the execution of the specified maximal number of iterations of the basic algorithm or in the case of achievement of the declared accuracy. Based on the formal definitions and analysis presented in [1], the efficiency of the iterative methods comes from their main features:

– the ability to solve larger problems, especially in three-dimensions,

– the development of highly effective preconditioners can enormously improve the speed and robustness of the iterative procedures,

– the ability to solve relatively large-scale problems in mini- and microcomputers,

– the possibility to vector and parallel programming.

The main factor affecting the performance of iterative methods is the number of the equation system expressed in the matrix, which may change frequently during the calculation. The problems presented in this paper are connected with solving ill-conditioned systems. The experimental part implements one of the best known and most efficient method for solving systems of linear algebraic equations – Incomplete Cholesky Conjugate Gradient (ICCG) method. Application of preconditions leads to reduction of the number of iterations. As a result the expected accuracy and simultaneously in reasonable time is obtained.

The proposed novel technology is based on operations on vector registers, to reduce the calculation time. To achieve this, a vector processing of multiple data sets procedure, namely Single Instruction, Multiple Data (SIMD) and Advanced Vector Extension (AVX) instructions implementation of the algorithms are applied.

This paper is organized as follows. In Sections 2 and 3 the conjugate gradient method and the preconditioning of the matrix are defined. Cholesky factorization is defined in Section 4. The methods of the utilization of processor registers and cache memory are presented in Section 5. AVX and optimization methods are characterized in Sections 6 and 7, and all the proposed techniques are evaluated experimentally in Section 8. The paper ends with short conclusions.

2. Conjugate Gradient Method

One of the most popular and effective method of solving the systems of equations is the Conjugate Gradient (CG) method, introduced by Hestenes and Stiefel in year 1952 [2]. The original model was then modified as iterative method for solving large systems of linear algebraic equations [1]. CG is a type of the Krylov subspace methods and usually it is applied to a system of equations defined by using the following matrix equation:

$$Ax = b\,, \tag{1}$$

where A is an $n \times n$ symmetric matrix of positive real numbers. The iterations number of CG should not exceed n without the round-off error. In practice, the number of iterations decreases depending on the specified accuracy level [2].

Following the formal definitions presented in [2] the general optimization problem for GC can be specified in the following theorem:

Theorem 1. If A is symmetric and positive definite, then the problem of solving $Ax = b$ is equivalent to minimizing of the quadratic form

$$q(x) := \frac{1}{2} x^T Ax - x^T b \,. \tag{2}$$

The idea of CG method is to generate a new vector x_{i+1} based on the best partial solution so far, which is defined as the vector x_i. This vector is characterized by two parameters, namely direction p_i and distance α_i, that is

$$x_{i+1} = x_i + \alpha_i p_i \,. \tag{3}$$

The coordinates of the *search directions vector* $p = [p_1, \ldots, p_i, p_{i+1}]$ are conjugate with respect to A. It can be defined as $p_{i+1} = r_{i+1} + \beta_i p_i$, where $r_{i+1} = b - Ax_{i+1}$ [1]. The values of the parameter α are estimated by using the Eqs. (2) and (3)

$$q(\alpha_i) = \frac{1}{2}(x_i + \alpha_i p_i)^T A(x_i + \alpha_i p_i) - (x_i + \alpha_i p_i)^T b \,. \tag{4}$$

Next, the partial derivative of α is computed:

$$\alpha_i = \frac{p_i^T r_i}{p_i^T A p_i} \,. \tag{5}$$

Now an approximation x_{i+1} from Eq. (3) and computation residual vector $r_{i+1} = r_i - \alpha_i A p_i$ is possible.

The next step is the calculation of a conjugation of directions:

$$p_{i+1} = r_{i+1} + \beta_i p_i \,, \tag{6}$$

where $\beta_i = \frac{r_{r+1}^T r_{r+1}}{r_i^T r_i}$.

Based on the above analysis, the CG algorithm can be defined by using Algorithm 1 [3].

Algorithm 1 Conjugate gradient method

Choose the initial approximation x_0 (e.g. 0)

$r_0 = p_0 = b - Ax_0$

For $i = 0, 1, \ldots, n-2$

$\alpha_i = \frac{p_i^T r_i}{p_i^T A p_i}$

$x_{i+1} = x_i + \alpha_i p_i$

$r_{i+1} = r_i - \alpha_i A p_i$

If the stop case is true – break

$\beta_i = \frac{r_{i+1}^T r_{i+1}}{r_i^T r_i}$

$p_{i+1} = r_{i+1} + \beta_i p_i$

End for

3. Preconditioning

The iterative methods are less demanded on computer resources than the direct methods, but unfortunately their accuracy is usually much worse. They cannot be successfully applied for some classes of the global optimization problems with many local solutions, where the iterative methods can be trapped in local optimum. One of the possible approach is the use of preconditioner, which is not always sufficient to achieve a convergence to the global solution in a reasonable time [4].

It allows to convert the matrix A from Eq. (1) to improve the distribution of its eigenvalues and reduce the condition number. It has a direct impact on the iterative methods convergence. Therefore, the matrix A preconditioning may be the key to an effective iterative method for solving systems of equations [5].

If a small change in the input causes a large change in the output, the problem is ill-conditioned, otherwise it is well-conditioned. In case of solving systems of linear algebraic equations problems, the lower condition number, the better conditioning task. In this context, the condition number of the matrix A is defined as [2]:

$$\kappa(A) = \|A\|_2 \cdot \|A^{-1}\|_2 \,. \tag{7}$$

To define the preconditioning algorithm first step is to reduce the condition number, and in the same time to reduce the number of iterations of the CG algorithm. To achieve this, the transformation of the linear system in Eq. (1) into the following one is provided:

$$M^{-1}Ax = M^{-1}b \,, \tag{8}$$

where M is a symmetric matrix of rational positive numbers.

Algorithm 2 Preconditioned conjugate gradient method

Choose the initial approximation x_0 (e.g. 0)

$r_0 = b - Ax_0$

$Mz_0 = r_0 \rightarrow z_0$

$p_0 = z_0$

For $i = 0, 1, \ldots, n-2$

$\alpha_i = \frac{z_i^T r_i}{p_i^T A p_i}$

$x_{i+1} = x_i + \alpha_i A p_i$

$r_{i+1} = r_i - \alpha_i A p_i$

If the stop case is true – break

$Mz_{i+1} = r_{i+1} \rightarrow z_{i+1}$

$\beta_i = \frac{z_{i+1}^T r_{i+1}}{z_i^T r_i}$

$p_{i+1} = z_{i+1} + \beta_i p_i$

End for

It is also assumed that M is well-conditioned, which means that $\kappa(M^{-1}A) \ll \kappa(A)$, where κ is a condition number of a matrix. The system $Mx = b$ is much simpler to solve compare with Eq. (1) [6]. However, the crucial issue here is to generate appropriate M – *preconditioned matrix*. The closer the M matrix to the original matrix A, the convergence of the method is better.

The modified CG method is called *a Preconditioned Conjugate Gradient* (PCG) method and it is defined in Algorithm 2 [3].

4. Incomplete Cholesky Factorization

In many mathematical tasks matrix is as a product of a number of other matrices. Solving linear systems of algebraic equations is the most important problem of these areas. In such cases the Lower Upper (LU) decomposition is useful.

LU is the decomposition of the matrix A as a product of the lower triangular matrix L and the upper triangular U : $A = LU$. The solution of $Ax = b$ systems reduced to two steps: solving of the $Lz = b$ with a respect to z, and solving of the $Ux = z$ with respect to x.

A special case of LU decomposition is when $U = L^T$. It is Cholesky factorization (decomposition), which can be formally defined as the distribution matrix for factors such LL^T.

Theorem 2. If A is real, symmetric and positive definite matrix, then it has a unique factorization, $A = LL^T$, in which L is lower triangular with a positive diagonal [2].

Cholesky factorization procedure is defined in Algorithm 3 [7].

Generation of the preconditioned matrix does not entangle complete factorization. The Cholesky factorization involves the solution of $Ax = b$ system, so in this case the incomplete Cholesky is used. This method returns the matrix close to A, a similar structured and characterized by lower expenditure of computing for solving the system of equations [8].

Algorithm 3 Cholesky factorization

For $k = 1, \dots, n$
$l_{kk} = \sqrt{a_{kk}}$
    For $i = k+1, \dots, n$
    $l_{ik} = \dfrac{a_{ik}}{l_{kk}}$
    End for i
    For $j = k+1, \dots, n$
        For $i = j, \dots, n$
            $a_{ij} = a_{ij} - l_{ik}l_{jk}$
        End for i
    End for j
End for k

Incomplete Cholesky Factorization (ICF) is one of the most important preconditioning strategy. This paper presents a variant of the ICF by position, as shown in Algorithm 4 [7].

Algorithm 4 Incomplete Cholesky factorization

For $k = 1, \dots, n$
$l_{kk} = \sqrt{a_{kk}}$
    For $i = k+1, \dots, n$
        If $a_{ik} \neq 0$
        $l_{ik} = \dfrac{a_{ik}}{l_{kk}}$
        End if
    End for i
    For $j = k+1, \dots, n$
        For $i = j, \dots, n$
            If $a_{ij} \neq 0$
                $a_{ij} = a_{ij} - l_{ik} - l_{jk}$
            End if
        End for i
    End for j
End for k

It should be noted that the Algorithm 4 is not always stable. As long as the M matrix is positive definite, it can be decomposed in to LL^T, it means that $M = LL^T$. In some cases, during the decomposition process, the matrix can be no longer positive definite. However, there is a solution which preserves the positive definiteness matrix during factorization process.

Algorithm 5 Stabilization of the incomplete Cholesky factorization

---

Start factorization with $\gamma = 0$
If during factorization process $a_{kk} < 0$
Return to initial state A
If $\gamma \leq 0$
        $\gamma = 10^{-20}$
Else: $\gamma = \gamma \cdot 10$
        End if
Correction matrix $A = D + S \cdot \dfrac{1}{1+\gamma}$,
        where: D – diagonal matrix, S – other elements
Restarting the factorization process
End if

Algorithm 5 solves the stability problem of Algorithm 4 by introducing a correction factor γ, which initially is equal 10^{-20}. In the case when the diagonal element (the second line) will be negative, calculation is interrupted, matrix returned to the initial state and all elements (except the diagonal) are multiplied by the value of $\frac{1}{1+\gamma}$. This procedure is repeated until the matrix is positive definite [9].

The application of Algorithm 5 allows to increasing the diagonal dominance of matrix A and is one of the possibilities to stabilize the factorization process.

PCG method (Algorithm 2) requires the solution of the $Mz_{i+1} = r_{i+1}$ equation. In this case, lower triangular matrix L for forward/backward substitution method could be used.

5. Processor Registers and Cache Memory

Typical computer processor CPU (Central Processing Unit) is composed of the Execution Unit (EU), and the Control Unit (CU) main modules.

The processor does not perform operations directly on the main memory, which is time-consuming. It has a number of small, high-speed memories, called registers. They are located in the EU and are used to temporarily storage of the results and control data. Number of available registers depends on the processor architecture. The internal memory of processor benefits from fast reading and writing.

Generally, memory stores data and programs. There are various types differ in cost and performance. The most important parameter is the access time (shorter access time increases cost). Therefore a hierarchy of memory was built. The highest levels of the memory are the fastest ones, but also the most expensive and smallest. The lower ones are slower, but larger and cheaper. Figure 1 presents computer memory hierarchy.

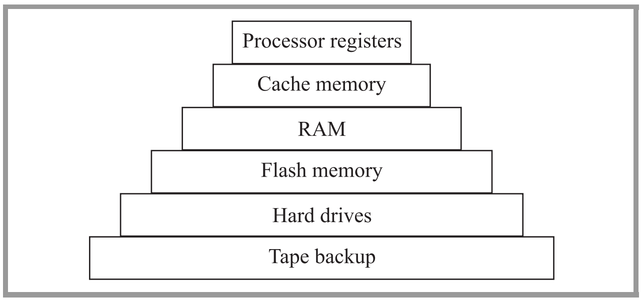

Fig. 1. Computer memory hierarchy.

Processor registers are located in the at the highest memory level. This is a static memory, and it is cleaned up in the idle mode of the computer. It has a very small capacity, e.g., 16, 32, 64, 128-bits, or 256 bits if CPU supports AVX instructions; access time is a fraction of a nanosecond. This is the fastest memory in the computer system [10].

Second in hierarchy is cache memory, which usually is a two or three-level (L1, L2 and L3) static memory with short access time. It is used to store a small amount of data, which are mostly used by the processor. Depending on the logical processor architecture, each level consists of the blocks (lines) in size 32, 64 or 128 bytes. The data between main memory and cache memory are transferred by same size blocks. Memory of the first level directly supports communication processor with main memory, while the lower-level memory (L2, L3) support the work of L1 cache. L2 and L3 – analogous to L1 memory – stores frequently used data in memory, and they are correspondingly larger. If the processor will not find the required data in L1, refers in the first instance to the L2 memory, and then – if it exists – to L3 memory. When the processor finds the requested data in the cache it is called the read hit, the opposite situation is read miss. Miss will reload the cache-line data. The new data is loaded, with completion of the cache line (up to the maximum) – because the tasks frequently cooperate with neighboring data [10], [11].

6. Advanced Vector Extension

In the Section 5, much attention has been paid to memories, including the fastest ones – CPU registers. The one type of registers is vector register that store the data processed by the SIMD architecture.

SIMD architecture is defined as systems which are processed multiple data streams based on a single instruction. Currently SIMD architecture is also used in personal computers. Processors use the extended set of SIMD instructions, such as MMX (MultiMedia eXtension), SSE (Streaming SIMD Extensions) or Advanced Vector Extension [13].

AVX is an extension of SSE instruction set that allows floating point operations on vectors of numbers using a special 256-bits processor registers (two times larger than previously used in processors that support SSE instructions). The introduction of new technology has forced changes in the architecture. Added 16 new registers are identified as YMM0, ..., YMM15. YMM registers are completely independent. It should be noted that the AVX instructions require support from the operating system. Older operating systems such as Windows XP or Windows Vista, even if the processor supports AVX instructions make impossible to use them [12], [13], [15].

AVX and previous technologies define two types of operations: packed and scalar. Scalar operations are present only on the least significant element of the vector (bits 0–63), while parallel operations on all elements of the vector in a single clock cycle [12]. The idea of operations on vectors is presented in Figure 2.

AVX has provided several new instructions, and now includes [13]:

- 19 instructions executable only on YMM registers,
- 12 multiply-accumulate instructions (MAC),
- 6 instructions support AES encryption,
- 88 instructions from the SSE instruction set, which may perform operations on vectors of floating point numbers stored in XMM/YMM registers,
- 166 instructions for 3- and 4-arguments operations.

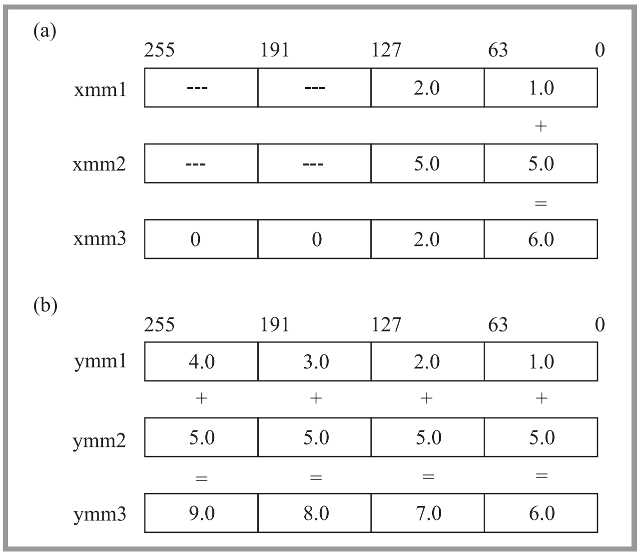

Fig. 2. Example of: (a) scalar and (b) packed multiplication.

A computer running in 32-bit mode has access to the first eight registers, in 64-bit mode to 16 registers. Due to the doubling of the size of registers, new data types are available:

– vector of eight single-precision floating-point numbers,

– vector of four double-precision floating-point numbers.

Most AVX instructions have their counterparts in special functions and data types used in C, C++, and Fortran programming languages. Using the appropriate functions and data types in C/C++ there is need to include library *immintrin.h* and compiler instruction: */arch: AVX* [13], [15].

7. Optimization Techniques

Loop unrolling is the first of the optimization techniques used in the implementation of ICF, CG and PCG methods. It allows reducing the number of hops by replicating code from loop body. Unrolled loop structure is closer to a more linear code and allows better use of the processor execution unit [14]. In implemented examples functions loops have been unrolled 8-times. This number was chosen because of the L1 cache-line size. Cache-line size of the computer where the experiment was performed is 64 bytes, while the size of one of a double is 8 bytes – thus in a cache-line fit in 8 double words.

The cache-line size is closely related to the second of the methods of optimization – data prefetching. It is realized by *void _mm_prefetch*(*char * p , int i*) function that loads a data block of size equal to the cache-line size [15]. The following example uses the prefetch function in combined with loop unrolling:

_mm_prefetch ((*const char* *) (&*vector*1[*i*+8]), *_MM_HINT_T0*).

The data are loaded from the shift of eight indexes of the double array to all levels of the cache. For single-precision data, shift will equal 16 indexes.

The application of data prefetching allows to hide the memory latency between sending and receiving a request for access to the memory. Processor must wait for data only in the first iteration of the loop [11].

The last of the optimization techniques are operations on registers (using AVX instructions). The introduction of operations on XMM/YMM registers forced to develop new types of data. In this paper two types of vector: _m256 and _m256d storing 8 float numbers and 4 numbers of double type respectively were used. Instructions for loading data into the vector registers (_mm256_load_ps/_mm256_load_pd), and unloading into RAM (_mm256_store_ps/_mm256_store_pd) require alignment of data within 32 bytes. Memory for all arrays is dynamically allocated and aligned by the function void * _aligned_malloc (size_t size, size_t alignment), where the first argument specifies the size of the allocated memory, and the second – the alignment (for instructions AVX – 32 bytes). The memory is release after performing of void _aligned_free (void *memblock) function. Static arrays have also been declared with the relevant directive: _declspec (align(#)) [11], [15].

8. Experimental Analysis

The next part of this work was the application of preconditioning in implementation of conjugate gradient method (Algorithm 2). The preconditioning method is stable variant of Incomplete Cholesky Factorization by position (Algorithms 4 and 5) and for comparison Conjugate Gradient method without preconditioning (Algorithm 1). The application is written in native C++ language.

The program consists of 16 functions, including: ICF and CG method in two versions: with and without preconditioning.

Solver has been tested on a server equipped with 16-cores, 64-bits AMD Opteron 6276 2.3 GHz processor based on Bulldozer microarchitecture. Opteron 6276 has three levels of cache, and the L1 memory is divided into data-cache (16 Kbytes) and instruction-cache (64 Kbytes). In the first-level cache is space for up 256 cache-lines 64 bytes each. Opteron 6200 series processors support MMX, SSE, SSE2, SSE3, SSSE3, SSE4, SSE4.1 + SSE4.2, SSE4a, AES, ABM, AVX, FMA4, XOP instructions. The server has 64 GB DDR3 ECC memory.

For the efficiency analysis of the proposed solution system of 512 linear algebraic equations expressed in the matrix which condition number is 2186 was used. Items are mostly floating-point numbers. The desired accuracy (set a priori) of the solution for each test case was set at value 10^{-6}.

First the impact of application of the ICF to obtain a solution of the system using the CG method was examined. For both, the Conjugate Gradient method with and without preconditioning allow to obtain the correct result with the expected accuracy.

The CG method gives the result with the expected accuracy in 79 iterations. The PCG, which use a ICF as preconditioner, reaches a result over five times faster – within 14 iterations (Table 1). For both methods the initial vector of solutions is the zero vector.

Table 1
Number of required iterations for obtaining the correct solution with the expected accuracy

Method (512 × 512 system)	Number of required iterations
Conjugate Gradient	79 iterations
Preconditioned CG	14 iterations

Figure 3 illustrates the process of reducing the error value with successive iterations. One can observe the PCG method is faster convergent than the CG.

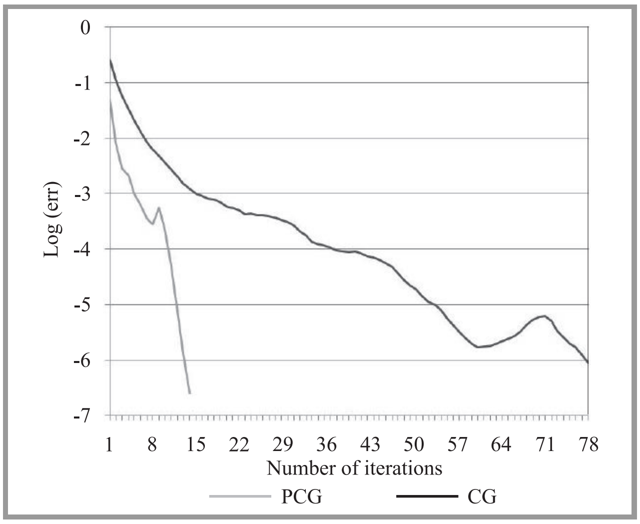

Fig. 3. Comparison of convergence of CG and PCG methods.

The second task of the experiment was to measure the computation time for the CG method with and without preconditioning in two options: standard and with using operations on YMM registers (AVX).

AVX instructions are used in: vector-matrix multiplication, scalar multiplication of vectors, calculating the Euclidean norm and the operation of the scheme: $\overline{z} = \overline{z} + (x \cdot \overline{y})$, $\overline{z} = \overline{z} - (x \cdot \overline{y})$ and $\overline{y} = \overline{z} + (x \cdot \overline{y})$.

For the PCG method additionally vector operations for the forward/backward substitution method was used, which solving systems of linear algebraic equations with triangular matrices.

Due to the small complexity of the task, calculations were repeated 500 times – in order to be able to observe changes in the time of obtaining solutions. All calculations were performed on the double precision numbers.

The results are shown in Table 2 and Figure 4. The system of equations was solved by CG over 8393 ms. Thanks to the vectorization computation task was solved three times faster compared to the solution without AVX instructions, which took 25569 ms.

Table 2
Times of obtaining solutions for all variants

Method (512 × 512 system)	Solution time	
	with AVX	without AVX
CG	8393 ms	25569 ms
PCG	4852 ms	14913 ms

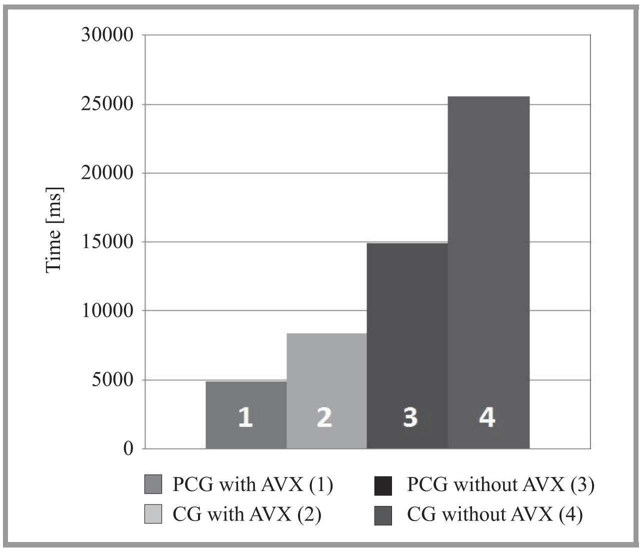

Fig. 4. Comparison of solving time for all variants.

For PCG method the expected results were received within 4852 ms for the AVX instructions and 14913 ms without.

Figure 4 illustrates the computational time differences between all solving methods.

The use of vector calculation and preconditioning technique resulted in the expected effect. Thanks to application of the AVX instructions and preconditioner, calculations were performed faster (81% less time). Even a small percentage increase in speed is important in solving large-scale complex mathematical problems.

9. Conclusions

The main aim of this research was to increase the speed of solving ill-conditioned systems of linear algebraic equations. These problems are characterized by slow convergence, and therefore, require many iterations to achieve the result with the expected accuracy. The author tried to solve this problem by using the CG, PCG and ICF methods dedicated for solving linear systems. The system matrix should be symmetric and positive-definite. The work focused on two factors that have a significant impact on the speed of the implemented algorithms:

- the number of required iterations,
- the time-consuming operations like matrix-vector and vector-vector multiplication and forward/backward substitution method.

The obtained reduction of the number of iterations enables to increase speed of convergence. The main factor affecting the number of iterations is the condition number. In order to minimize this parameter, the ICF procedure was implemented. This variant keeps the stability of the algorithm during the decomposition process at the fair level.

In simple experimental analysis, a system of 512 equations was defined. By using Incomplete Cholesky Decomposition the number of iterations decreased more than five times without having a negative impact on the result accuracy. It is worth to note that the ICF method with its variants is widely used in various fields to solve technical issues. In practice there are no specified the universal methodology to choose the best method of preconditioning. In the other words it is impossible to determine which form of preconditioning would be best for each problem [16].

Another issue is a performance of matrix-vector operations and forward/backward substitution. They are the most time-consuming steps of obtaining solutions of equation systems. In order to accelerate this operation three closely related methods are used: loops unrolling, data prefetching, vector operations.

The loops have been unrolled hence during one iteration data required for the next one can be loaded. While information is loading, operations are parallel performed. By using these two techniques the waiting time between a requesting and receiving access for data was eliminated.

The last method use innovative technology of modern processors – 256-bits vector registers YMM. During one clock cycle are performed parallel operations on all elements of the vector.

Application of operations on the most expensive and also on the fastest computer memory, allows to obtain significant acceleration of the calculation. Reduction of the solving time of conjugate gradient method by about 67% was observed. This is due to application high-performance techniques in the application.

Application of the preconditioning technique and AVX instruction allows to solve the problem more than five times faster and effectively reduce the number of iterations.

Despite the results, it should be mentioned that not all possibilities of increasing the efficiency have been used in this study. There are many other techniques, both related to the preconditioning methods and the use of modern IT solutions, e.g., implement specific algorithms for sparse matrices, which are increasing the efficiency. It is also possible to use ready-made high-performance libraries such as BLAS, LAPACK, MKL. It is worth to consider application of adaptation solutions for multi-threaded and distributed computing, or expand the use of AVX instructions.

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An Incentives Model Based on Reputation for P2P Systems

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Abstract—In this paper an incentive model to improve the collaboration in peer-to-peer networks is introduced. The proposed solution uses an incentives model associated with reputation issues as a way to improve the performance of a P2P system. The reputation of the all peers in the system is based on their donated resources and on their behavior. Supplying peers use these rules as a way to assign its outgoing bandwidth to the requesting peers during a content distribution. Each peer can build its best paths by using a best-neighbor policy within its neighborhood. A peer can use its best paths to obtain best services related to content search or download. The obtained results show that proposed scheme insulates the misbehaving peers and reduces the free-riding so that the systems performance is maximized.

Keywords—content distribution, incentive model, reputation, peer-to-peer networks.

1. Introduction

During the last years, content delivery over the Internet has gained significant popularity. Applications such as TV over IP, streaming and multimedia live streaming are examples of content distribution from one-source to multiple receiver-nodes. On the other hand, peer-to-peer (P2P) networks have attracted the attention from the research community who find in these systems a fast and efficient way to deliver movies, music or software files. A P2P communication infrastructure is formed by a group of nodes located in a physical network. These nodes build a network abstraction on top of the physical network, known as an overlay network, which is independent of the underlying physical network with regard to the P2P procedures. An important advantage of P2P systems is that all available resources are provided by the peers. In a P2P system each peer can take the role of both, a server and of a client at the same time. During content distribution, peers contribute their resources to relay the content to others. Thus, as a new peer arrives to the P2P system the demand is increased, but the overall capacity too. This is not possible in a client-server model with a fixed number of servers.

A challenge in peer-to-peer media streaming systems is how to select good peers in order to realize high quality streaming sessions. The selection of good peers can offer a manner to improve the quality of service via an optimal search or an efficient content delivery. However, this goal is difficult to be achieved because P2P systems can be affected by misbehaving peers and free-riding, which reduce the system performance. Reputation management systems have been proposed as promise methods to alleviate this problem [1]–[3]. A reputation management system allows individual peers to rate one to each other according to their past experience with each other. Reputation systems are proposed in [1], [4]–[6] with the purpose to ensure that peers obtain reliable information about the quality of the resources they are receiving.

On the other hand, locate a provider node does not guarantee that the service provided by it will satisfy user demands [7]. This is because some misbehaving peers may offer false information in order to maintain a cooperation impression. To minimize the effects of misbehaving peers, they must be detected and isolated from the system. In this paper an incentives model associated with a reputation scheme to reduce the negative impact of these peers in a P2P system is proposed. In a reputation management system, each peer can realize an optimal peer selection from which download a specific content. The goal in this work is to examine how incentives and reputation affect the performance of a P2P network. To reach this goal, both characteristics in order to obtain the following benefits is mixed. First, peers with high reputation can cooperate to realize an optimal search or a better content delivery. Second, an incentives-system can encourage the collaboration and exchange of data between peers [8], [9]. Finally, the isolation of misbehaving or non-cooperative peers can avoid the degradation of the system performance. In this way, the number of corrupted file downloads from malicious peers on the P2P network could be minimized, the number of cooperative peers increased and the number of success downloads improved. The author motivation to implement incentives and reputation issues on a P2P network are the following:

– selection of good peers could improve the content delivery services,

– simulate a P2P system that allows a fair distribution of the available resources in the networks,

– provide different access polices to the system resources,

– increase level of cooperation to provide high quality streaming to users in a large system,

– empirical validation of the effectiveness of our approach in a P2P network.

The rest of this paper is organized as follows. Section 2 gives a background about this work. The incentive and reputation schemes are introduced in Section 3. Then, the evaluation of proposed system and its results are described and discussed in Section 4. Section 5 concludes the paper.

2. Background

2.1. Media Content Delivery

Today, we witness an exponential rise in the number of Internet users. Many of these users generate contents that are accessed by other users interested in them. Multimedia contents has already become the most popular content to be distributed over Internet. This is due to the fact that multimedia contents are currently generated by a high number of applications such as videoconferencing, media broadcasting, e-learning, video streaming, etc.

Media can be transmitted in two different modes: download an streaming. In the download mode, the users have to download the entire media file before playing it. However, the media files generally are very large which require long transfer times and large storage capacities. The download mode requires patience from the users, who have to wait until the entire video has been downloaded before it can be viewed. Download also offers reduced flexibility, because the users must download the entire video before deciding if it is the wanted one [10]. In contrast, in the media streaming mode, the receiver can already consume the media file while part of it is being received and decoded. In others words, media streaming reduces the delay between the start of delivery and the beginning of playback at the viewer, and its requirements of storage are low, because only a small portion of the video needs to be stored by the viewer during media streaming. However, video streaming is sensitive to the delay [11], because the packets must arrive at the receiver before their play-out deadlines.

Media delivery can be realized using different communication techniques such as as: unicast, broadcast and multicast. Unicast represents a common communication form between two entities. Unicast communication also is known as point-to-point or one-to-one communication. Unicast communication can be simplex, half-duplex or duplex. Telephone conversations and video streaming over the Internet [12] are typical unicast examples. Broadcast means that the information emitted from a source will be received by all the other devices connected at the same network. Broadcast probably represents the most popular communication scheme due to its wide usage in broadcast television. Multicast is similar to broadcast except that the information emitted from a source is only received by a specific group of nodes in the network, which is called a multicast group. Multicast is an alternative to unicast that reduces the network traffic and optimizes the server resources. Multicast is a one-to-many communication scheme, while broadcast is an one-to-all communication. Videoconferencing is a multicast example, where a predefined group of devices/computers are involved to receive the same content.

2.2. P2P Content Delivery Topologies

The two most important types of technological solutions that have been proposed for content delivery on the Internet are Content-Delivery Networks (CDN) and Peer-to-peer (P2P) networks. A CDN is formed by content servers networked together across the Internet, which cooperate with each other to transparently distribute content to end-users. Typically, the content servers are located near the users, in order to be able to serve the requested content rapidly [13]. However, the CDN approach faces a number of problems such as single point of failure and costly access to high rate networks. In [14] is presented an extensive discussion about CDN. On the other hand, P2P networks has emerged as a promising infrastructure to the distribution of large-sized media content to a large population [15]. A P2P communication infrastructure is formed by a group of nodes located in a physical network. These nodes build a network abstraction on top of the physical network known as an overlay network, which is independent of the underlying physical network. The overlay network is established by each P2P system through TCP or HTTP connections. Due to the abstraction layer TCP protocol stack, the physical connections are not reflected by the overlay network [10]. Based on how the nodes in the overlay structure are connected to each other, P2P systems are classified mainly into two categories: unstructured and structured. Unstructured P2P systems can be further divided in [16]: centralized, pure and hybrid. Kademlia [17] is an example of pure P2P systems, while BitTorrent [18] is an example of hybrid P2P systems. Figure 1 shows this classification.

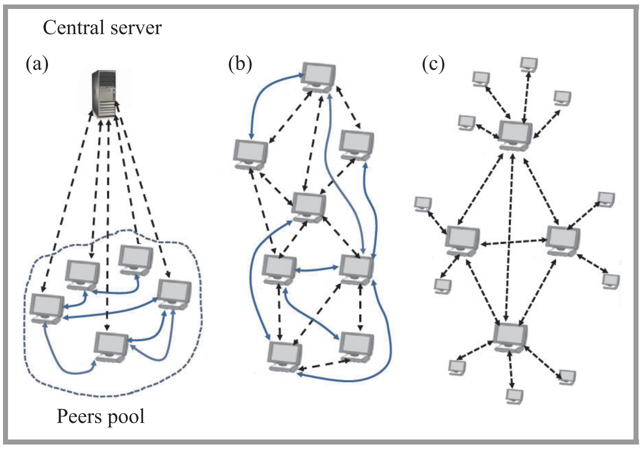

Fig. 1. A comparison among the different unstructured P2P architectures: (a) centralized, (b) pure, (c) hybrid.

Content delivery over P2P networks can be realized using three different schemes: tree-based, forest-based and mesh-based. These schemes are illustrated in Fig. 2. In a tree-based system, participating peers are organized in a hierarchical way in which source node is located at the root. The rest of peers are organized as interior nodes or leaf nodes into a single tree. Leaf nodes don't need to forward the receive packets [19]. In Fig. 1a, the source S sends the data to requesting peer R_1, which forwards the data to requesting peers R_2 and R_3. Then, a packet in this configuration is basically pushed from a parent node to their children node along a well-defined route. Thus, in Fig. 1a the upload capacity of peer R_1 is used by the multicast tree for content distribution, while the upload capacity of the leaf peers R_2 and R_3 is not used. Main drawback in a tree-based scheme is because all the burden generated by forwarding multicast messages is carried out by a relative small number of interior nodes.

Cooperation plays an important role in the P2P systems. However, content delivery based on a single tree does not match well for cooperative environments, because the forwarding multicast traffic is carried by a small number of interior peers, while the upload capacities of a large number of leaf peers is not used. To face these challenges, a forest-based overlay architecture for content delivery has been proposed. A forest-based overlay organizes participating peers into multiple trees [20], and distributes the forwarding load among them in an efficient manner. An example of a forest-based system is shown in Fig. 2b. Here, source S stripes its content and distributes the stripes using two separate trees T_1 and T_2. Each internal node in each distribution tree forwards any received stripe to all of their child nodes. However, the determination of the number of required trees to maximize the overall throughput of the system is a hard task.

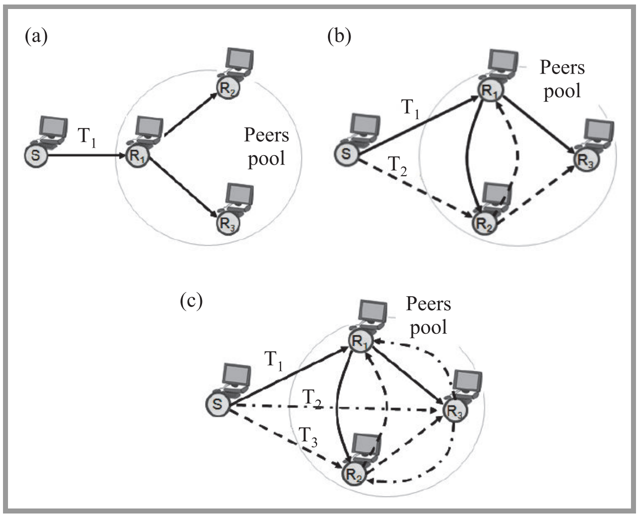

Fig. 2. P2P streaming delivery schemes: (a) tree-based overlay, (b) forest-based overlay, (c) mesh-based overlay.

Limitations of the forest-based systems have led to the emergence of a new approach called mesh-based systems. This type of P2P systems are formed by peers, which are interconnected via random connections. In this scheme each peer (except the source) tries to maintain a certain number of parent peers and also serves a specific number of child peers using a swarming mechanism for content delivery [21]. A mesh-based scheme is shown in Fig. 2c. In a mesh-based P2P topology, a peer can concurrently receive data from different sources, and send the received data to other requesting peers. Although mesh-based P2P systems are less vulnerable to network dynamic [13], they introduce long latency in media playback mainly due to periodic exchanges of buffer maps and transmission of data request.

2.3. Why Do We Need Reputation Management?

P2P networks are liable to be invaded by the malicious peer, this is due to the fact that any peer can join the network at any time to share or use any type of file. These malicious peers are computers that share inauthentic files or give false information about their resources (CPU, memory available, bandwidth, etc.). Examples of inauthentic files include corrupted files, virus-infected files or spam [22]. In order to minimize the effects of malicious peers on a P2P network, there is a need to isolate these peers from peers with good behavior who are sharing authentic files, which are high-quality, virus-free files. In addition, peers should be informed about the best sources from which download files. This information needs to have a way to be propagated through network so that all peers have a wide view of the reputations of all other peers in the network. In a P2P network with a reputation management system, each peer will be better able to make good decisions about which other peers are available to download files, thus minimizing the number of files downloaded from malicious peers on the network and increase the number of successful downloads.

2.4. Why Do We Need an Incentives Mechanism?

Most P2P systems are based on cooperation among interested users [19]. However, cooperation consumes user's resources and may degrade their performance, which could generate disincentives for cooperating in the users. As a result, each user's attempt to maximize its own utility effectively lowers the overall performance of the system. To avoid this scenario, there is a need to introduce incentives for the cooperation. To maintain satisfied peers in the system, the proposed strategy of upload allocation is to make that each peer's download rate be proportional to its upload rate (parity download/upload). The author used the reputation generosity rings to translate this approach in an incentive way to proposed model. To verify data integrity, there are many techniques, such as SHA1, which hashes all the pieces included in the file, and peers don't report that they have a piece until they have checked the hash. Selection of good choking algorithms in order to utilize all available resources, provide reasonably consistent download rates for all peers.

3. Related Work

P2P systems based on reputation have been proposed in several works [1], [2], [4], [5], [7]. In most of these

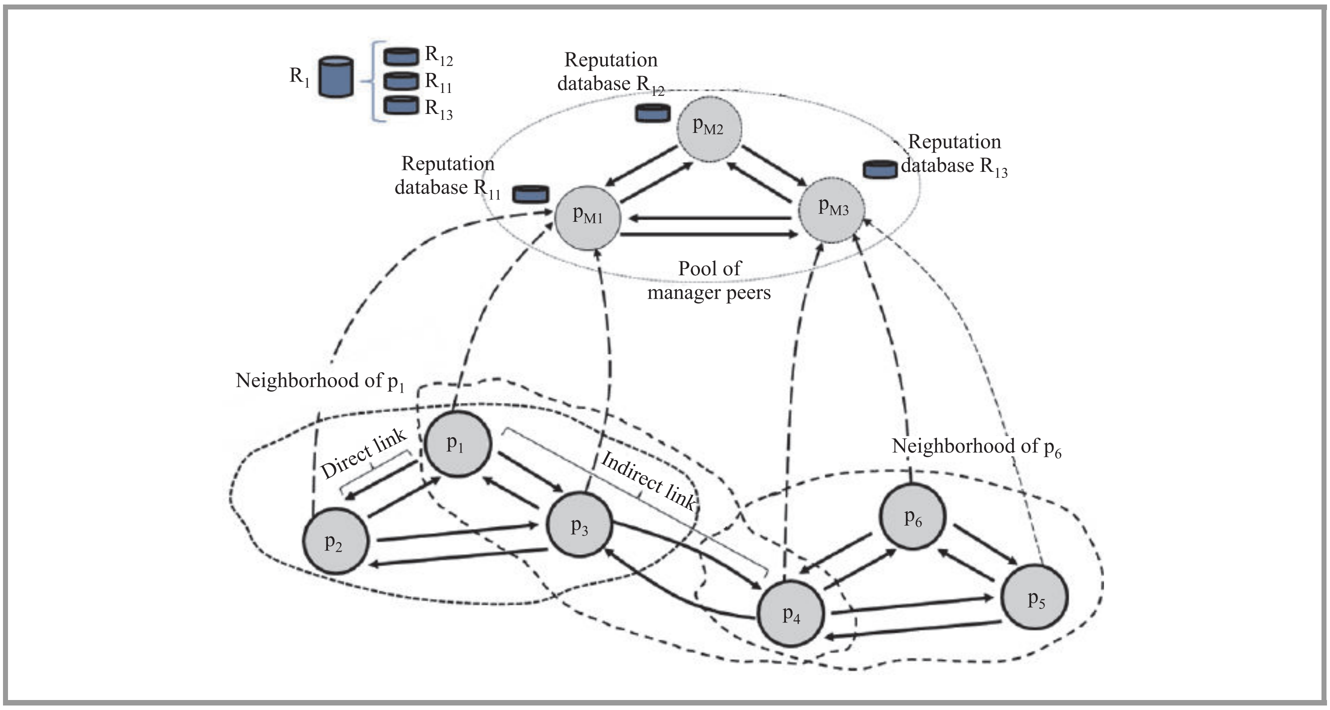

Fig. 3. The proposed reputation architecture with participating peers, neighborhood and pool of managers peers.

proposals, each node is associated with a reputation established based on the feedbacks from others that it has made transactions with. The reputation information helps users to identify and avoid malicious nodes [23]. On the other hand, independent mechanism or systems associated with reputation have been proposed in some recent works [23]–[29]. Authors in [24] show that a rank-based incentive mechanism achieves cooperation through service differentiation. In this framework, the contribution of a user is converted into a score, then the score is mapped into a rank, and the rank provides flexibility in the peer selection that determines the quality of a streaming session. The incentives mechanism reduces the data redundancy required during a streaming session to tolerate packet loss [24]. Without the incentive mechanism, it is required to send more redundant data to achieve the same QoS that the incentive mechanism can provided. In [25] is proposed a wage-based incentive mechanism for enforcing truthful report in self-interested P2P networks. Additionally, authors propose a set of incentive compatibility constraint rules including participation constraints and self-selection constraints. Incentive schemes have been used in the P2P systems to reduce free-riders and to encourage cooperative behavior in [27]. Transactions of a distributed P2P file-sharing system are modeled in [26], using feedback based on transaction outcomes and reputation issues, to encourage cooperation among peers. In [23] Zhan *et al*. propose a distributed incentive scheme called MARCH, which associates money and reputation parameters to each peer. In this scheme, peers can increase their reputation level by exchanging money for service. Since it is a scheme based on a business model, a central authority is used to settle disputes between peers when services offered by a peer are not satisfactory. Based on this rule peers can be classified as honest, selfish and malicious.

This paper presents an incentive model on P2P networks that isolates misbehaving peers. The goal is that by isolating these peers the system performance can be improved. Having peers with high reputation can cooperate to make an optimal search or a better content delivery. An incentive scheme can also help in collaboration and exchange of data between peers. Author evaluates proposed protocol in two different P2P infrastructures, which are Kademlia and BiTorrent. Kademlia [17] is a distributed hash table protocol designed for decentralized P2P networks, while BitTorrent is a hybrid P2P system based on super-peer.

4. System Overview

To introduce proposed incentives and reputation scheme, an overlay network is considered, which is formed by collaborating peers and free-riding peers. The proposed model introduces a special peer called manager peer, which manages the reputation of all peers in the system. Manager peer considers that each peer has reputation information of its neighbor-peers only. This reputation score is stored in a local table by each peer. Each peer exchanges its local table with any others peers located in its neighborhood, which is formed by 2 hops away. A peers establishes direct links with peers to be reached in one just hop. Contrary, indirect links are established if the number of hops between two peers is two. Figure 3 shows this scenario. The proposed model consists of two parts: reputation scheme and incentives scheme.

4.1. Reputation Scheme

Proposed reputation model considers that all peers contribute with their resources to the system. Two components to obtain the average reputation-score in each peer are used, which are its shared resources and its behavior in the system. Shared resources by a peer is used to define its reputation based on resources (RBR), while the behavior reputation is defined by the peer's behavior. In this work, the resources to be shared by each participating peer in the systems are: upload capacity, processing capacity, memory, storage capacity and number of shared files. Thus, reputation based on resources (RBR) can be computed as:

$$RBR = X_i(CPU) + X_i(HD) + X_i(BW) + X_i(SF)\ , \quad (1)$$

where: CPU = donated CPU capacity, HD = donated hard disk capacity, BW = donated bandwidth capacity, SF = shared files, and

$$\sum_{i=1}^{4}(X_i) = 1\,.$$

The weight X_i for each donated resources are fixed by the manager peers in the system. X_i can be variable and different for each resource.

The behavior reputation is the second component to be evaluated. This value is based on the peer's behavior in a cooperation environment, from a cheating level to a transient level. The cheating level is assigned when a peer supplies a wrong content or when its donated resources do not match to promised resources. The transient level in a peer is determined by the average time that this peer remains in the system (service-time) and by the average time that it takes to return to the system after it has left. Users are satisfied when they received content from peers with abundant resources and good behavior. In the other hand, users have a bad experience when the participating peers offer low bandwidth, high error-rates, limited processing resources, or frequent disconnections. Each component in the reputation scheme contributes with its weight in the final score.

Initially, the reputation score of a peer is based on its donated resources only. After several rounds, if this peer is still available, stable and it does not cheat, then the behavior reputation score is increased in this peer. Otherwise this score is decreased. Behavior reputation evaluation considers that a transaction realized by any peer can be either, performed correctly or not. Chosen behavior reputation scheme is inspired by a reputation scheme introduced in [30]. Peers interact using a reputation approach.

A complaint message is evaluated in every peer and in the manager peer. Using this information, each peer builds a reputation matrix with information from its neighborhood or from the network. Each peer computes the reputation of another peer by evaluating to its neighbors. In general, this reputation is usually based on an aggregate of the feedback ratings issued by the diverse peers [2]. When a peer interacts with another peer, it may rate the transaction as satisfactory (+1) or unsatisfactory (–1). A transaction is satisfactory if the retrieval process requested by a peer is realized successful. In contrast, a transaction is unsatisfactory if the requested file is not authentic, promised resources are false or the download process is often interrupted. In this case, a peer is cheating and its reputation score must be penalized. A peer records the reputation score in the reputation matrix, as a local table, while the manager peer records the reputation score in a global reputation table. This global table is consulted by a peer when it wants to know the reputation of peers outside its neighborhood (see Fig. 3). A peer can exchange its local reputation table within its neighborhood in order to insulate the cheating peers.

When a new peer joins to the system, this peer has all its entries as undefined, which are updated as the peer interacts with each other. Every peer updates the reputation of its local reputation matrix, while the reputation information from remote peers are obtained from the manager peers.

To compute the behavior reputation in the P2P system, proposed protocol periodically runs a process in order to update the network. A reputation agent updates the reputation score and the incentives of every peer based on its behavior (cheating level and transient level). Initially, the global reputation score is based on resources only, and the behavior reputation is initialized in 0. The behavior reputation score of a peer is increased if it maintains a good service or it does not cheat. On the other way, the behavior reputation score is decreased.

Each peer has the following statuses: Up, Down and Cheating. These status define several scenarios for each peer such as reputation update of a peer, expiration of a round or beginning the update process. Up status means that peer does not cheat, then a reputation agent computes the average number of round that a peer remains connected to the network and its behavior reputation score. Down status means that peer is disconnected. If the peer status is Down and it does not cheat, then, to determine how many rounds a peer is in this status is needed.

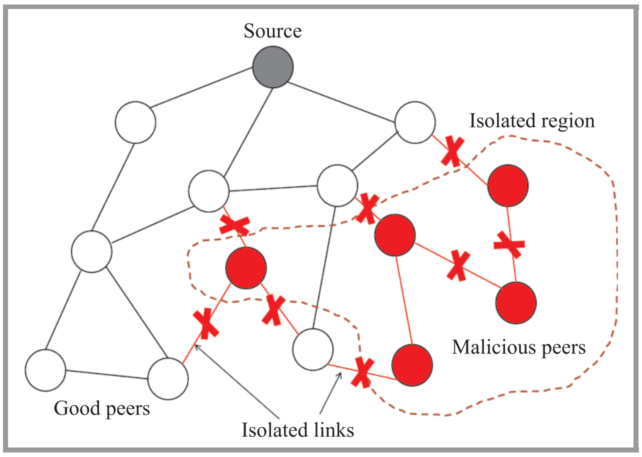

Fig. 4. Cheating peers are isolated by the system.

To calculate the behavior reputation, the author divides the simulation time in discrete rounds, where the network is updated. In this period the reputation values and incentives in each peer are revised and updated according with its behavior (e.g. connected time, stability, cheating, availability) and current resources. During the first round, the reputation score is zero for all peers. However, in the following rounds the reputation score of all peers is based on shared resources and their behavior. If the status of a peer is Cheating, then its reputation is decreased to 0 in all its neighbor peers and in the manager peers. All peers isolate the cheating peer, and they do not send, forward or receive any messages or data from it. An example of cheating peers isolated by the system is shown in Fig. 4.

4.2. Incentive Model

P2P systems are basically based on cooperation among interested users. However, many users of these systems have natural disincentives to cooperate because cooperation consumes their own resources and degrade their own performance [24], [31]. Consequently, each user attempts to maximize its own utility effectively lowers the overall utility of the system. Incentives have been used as a useful solution to motive the cooperation among peers in a P2P system. In this work, an approach based on the game theory is adopted. In particular, a choking algorithms model is used to capture the essential tension between individual and social utility, asymmetric payoff matrices to allow asymmetric transactions between peers, and a learning-based population dynamic model to specify the behavior of individual peers. Peers can continuously change its behavior.

Presented cooperation approach considers upload and download rates as a generosity factor, and translates the cooperation concept to earnings if peers cooperating or it loss if not. Generosity factor measures the benefit that an entity provides relative to the benefit it consumes. This is important because entities which consume more services than they provide, could cause the cooperation collapse. For some entity i, P_i and C_i are the services provided and consumed by i, respectively. Therefore, generosity of an entity i can be represented as:

$$Generosity(i) = P_i/C_i\,. \tag{2}$$

The generosity resumes the General Prisoner's Dilemma for an asymmetric payoff matrix [9], [32]. For our scheme, we define as a provided services unit to a packet transmitted successful. On the other hand, a consumed services unit is defined as a packet received successful. The range of the generosity factor comprises –1 to +1. The minimum value (–1) indicates a not cooperate behavior in which peer only receives packets and does not transmit any. The maximum value (+1) indicates an overall cooperate behavior in which a peer only transmits packets and does not receive any.

Each node builds a hierarchical structure of rings using its reputation table and the generosity factor. These reputation/generosity rings are used by a supplying peer to organize to the requesting peers, which wish to download any content from it. Each peer encloses each requesting peer in rings following a hierarchical organization. Thus, peers with highest reputation and generosity are allocated in the higher rings close to the source. This means that these peers receive more incentives than peers allocated in lowest rings. Figure 5 shows this scenario. In this case, eight requesting peers are distributed through three rings. The number of reputation rings and its reputation thresholds are values can be fixed by manager peers. In this work, if a requesting peer receives more incentives means that it receives more bandwidth to download contents. These reputation/generosity rings are used to implement proposed content distribution scheme. The incentives percentages can be fixed by each peer.

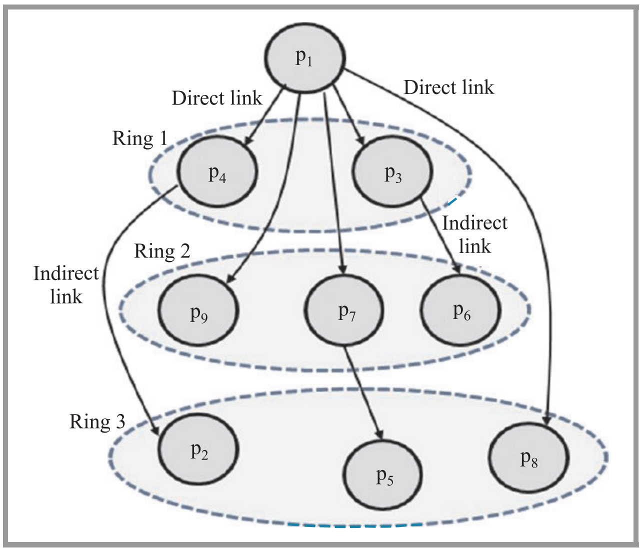

Fig. 5. Example of a distribution scenario with three reputation rings.

4.3. Operation Protocol

Presented protocol defines a set of communication messages to exchange data between peers. A set of rules are used to govern these communication messages. When new peer joins to a P2P network, it must contact with a manager peer and its neighbors. First, a new peer discovers its neighbors and builds its local table with its neighbors. Second, each new peer must deliver information about its local table to the manager peer. Third, the manager peer compares its global table with the local table received from the new peer in order to update the behavior of the peers in that neighborhood. Thereby, each peer has information about resources and reputation of its near neighbors (two hops). This information is stored and updated in a local table and in the general table of each manager peer. Once a peer is joined, it selects its first and second best neighboring peers, and calculate its best neighbor paths. The best neighbor paths are the best paths obtained from an average reputation of the neighbors (see Fig. 6), which have a direct connection between each other. Each peer arranges and stores these paths, which are used during the search stage.

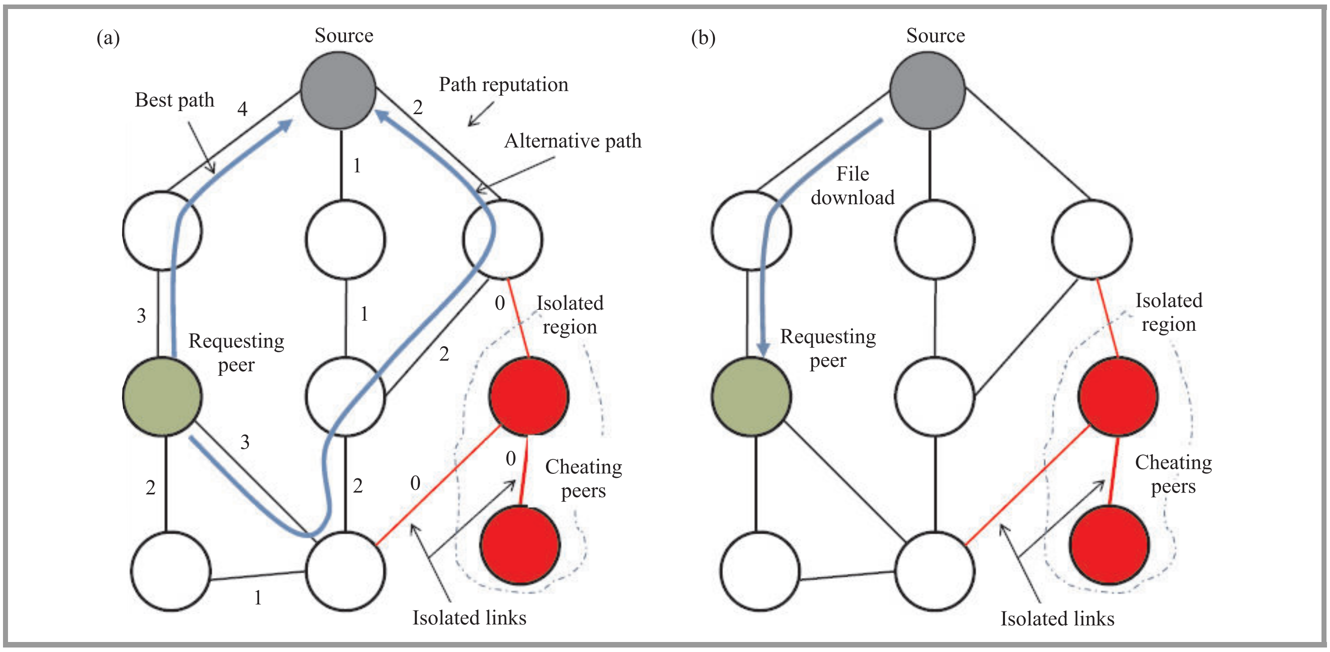

Fig. 6. Example of a content distribution scheme based on the best path strategy: (a) file search, (b) file download.

Content distribution scheme is constituted by two stages: file search and file download. In file search stage, a peer begins a file request by sending a "query" via its best neighbor. This query is forwarded by the neighbors to their best neighbor paths, and the responses are collected. Each peer also forwards to their best neighbor paths the queries realized by its neighbors until the requested file is found. For the file download stage, first, source peer checks all reputation scores of the requesting peer. Second, requesting peers are organized in rings. Following the incentives rules, source peer divides its resources between these requesting peers. If requesting peer is authorized to download a file from the source, then this file is downloaded via the best path. This scenario is shown in Fig. 6. In this case, best path has a reputation score of 3.5, while alternative path has a reputation score of 2.25. When content download is completed, requesting peers are removed from the reputation/generosity ring by the source peer.

5. Evaluation

In this section, the performance of proposed solution is evaluated. To this end, proposed model has been simulated in the Peersim simulator [33]. The author has used this network simulator because it supports dynamism and scalability. Peersim is written in Java language and it can be used to simulate small and large-scale P2P systems. This simulation tool also allows to measure the communication time between nodes. Peersim is composed of two simulation engines: the cycle-based model and a more traditional event-based model. The simulation uses the BitTorrent and Kademlia prototype [34] developed by the Trento University for the Peersim simulator. Communication protocols in both prototypes are developed using Java as programming language. Information about the resources donated by each peer such as CPU, bandwidth, memory and storage capacity is recorded for each peer in its local table. The initial reputation is based on resources only. Initially, there are not any relationship among peers in the system.

A BitTorrent network is formed by several actors and components such as peers, leechers, seeders, trackers and swarm. All users connected to the BitTorrent network are called peers. In this context there are two types of peers: seeders and leechers [35]. Seeders are users who have a file, while leechers are users that only download files. Kademlia is a distributed hash table protocol designed for decentralized P2P networks. Kademlia is deployed as a virtual network on an existing LAN/WAN network or Internet and its topology is based on the XOR metric. This metric is use to calculate distance between points in the key space [17]. Kademlia protocol consists of the following RPCs: Ping, Store, Find Node, and Find Value. These procedures allow to specify the network structure, regulates communication between nodes and exchange of information. The nodes communicate is realized via the UDP protocol. Reputation levels in BitTorent are based on shared pieces, while in Kademlia reputation levels are measured during a direct download from a node.

To simulate presented protocol on Kademlia and BitTorrent 300 nodes (peers) are used and the same file is downloaded from the source node. The author evaluates BitTorrent and Kademlia, without reputation and with reputation levels. The first experiment evaluates both P2P networks without reputation levels in the nodes. In this experiments, 15% of nodes are initialized as seeders nodes while 85% are leechers nodes. The size of the file to be download is

100 MB. Initial reputation is random. If node's reputation rate is 0, then it removed from the system. Results from first experiment are shown in Figs. 7 and 8. The number of leecher nodes that are downloading the same file from Kademlia network such as from BitTorrent is compared in Fig. 7. It shows how BitTorrent maintains a smaller number of leecher nodes compared to Kademlia. Figure 8 shows the number of seeder nodes from which some parts or full file can be download, and how in BitTorrent the number of seeder nodes is increased as time goes. Thus, BitTorrent has more available seeder nodes from which download the test file than Kademlia. This facts allows that BitTorrent presents a best performance to search and download a file compared to Kademlia.

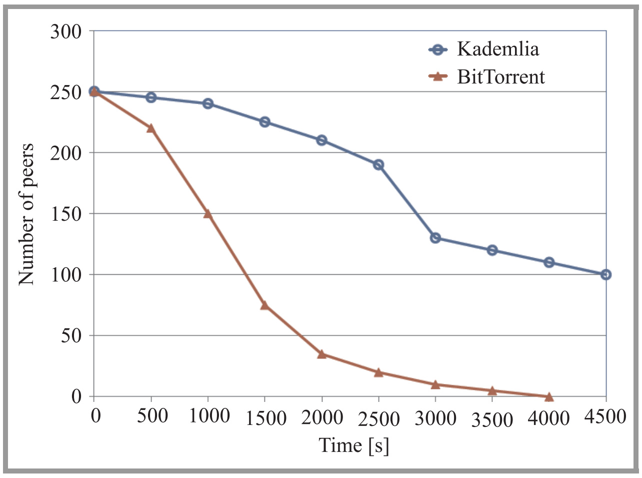

Fig. 7. Comparison of number of leecher nodes in both P2P infrastructures without reputation.

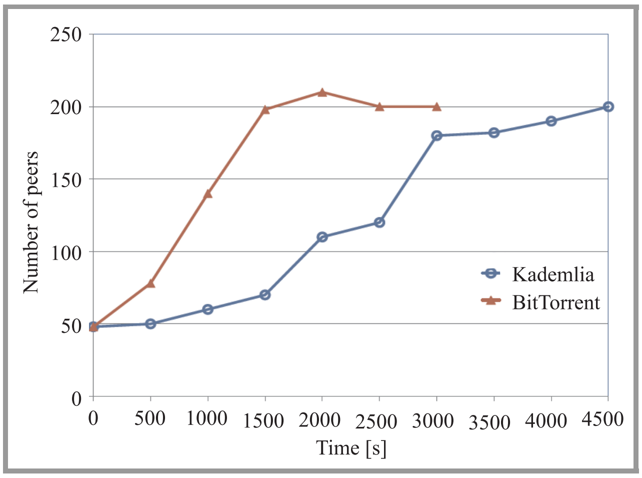

Fig. 8. Comparison of number of seeder nodes in both P2P infrastructures without reputation.

The second experiment evaluates both P2P infrastructures using reputation levels. As in previous experiment, 300 nodes and a file of 100 MB is considered. Initially, also 15% of nodes are seeder nodes and 85% are leercher nodes. Figures 9 and 10 show the results obtained from our second experiment. Figure 9 shows that BitTorrent still maintains a smaller number of leecher nodes than Kademlia. However, leecher nodes are dramatically reduced in the Kademlia network as time goes. Figure 10 compares the number of seeders nodes in both P2P networks with reputation. Initially, BitTorrent has more seeder nodes than Kademlia, but as time goes the number of seeder nodes in both P2P infrastructures is similar. Therefore reputation levels in Kademlia have served to reduce the number of leechers nodes and to increases the number of seeder nodes.

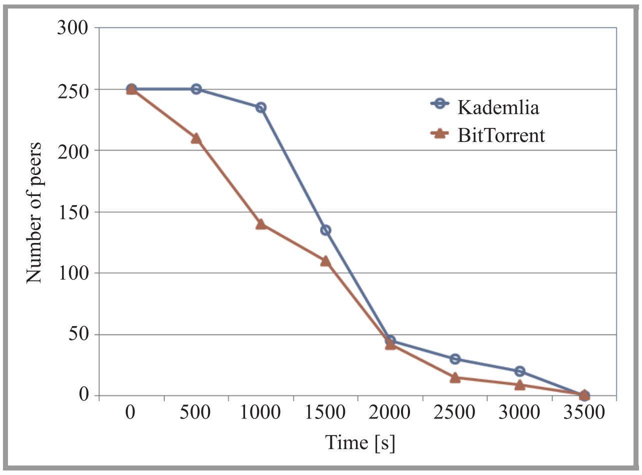

Fig. 9. Comparison of number of leecher nodes in both P2P infrastructures with reputation.

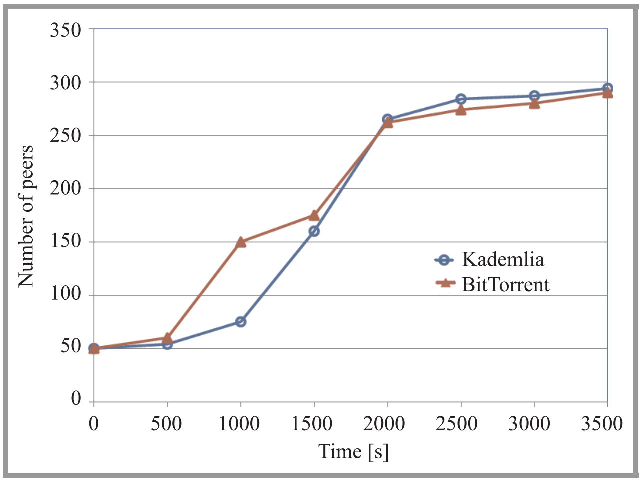

Fig. 10. Comparison of number of seeder nodes in both P2P infrastructures with reputation.

In a reputation system, good peers cannot send queries to the misbehaving peers or receive request from them, because they are isolated. Also, a system without reputation allows download content from the misbehaving peers increasing the probability of having a greater number of corrupted files. Contrary, cooperation between peers and reputation help to reject corrupt files in the systems. However, in the real electronic communities correcting the malicious peer's behavior is a hard task. Instead of correcting each such malicious peer, the author need to minimize its impact in the system performance.

6. Conclusions

In this work, a reputation scheme based on incentives for a P2P system was proposed and evaluated. Most of the reputation systems consider correction of malicious peers by giving incentives for positive feedbacks. However, in our proposed model isolates misbehaving peers from good peers, and incentives are used to motivate most cooperation among peers in the system. The model was evaluated for two different P2P infrastructures which are Superpeer (hybrid) and distributed. BitTorrent is used to simulate the super-peer infrastructure, while the Kademlia protocol is used to simulate the distributed infrastructure. The experiments show how reputation reduces the number of leechers peers and increases the number of seeder peers in both infrastructures. Although, Kademlia is the P2P infrastructure most benefited by incorporating reputation levels at nodes, BitTorrent still presents a little best overall performance, which is reflected by downloading the file most faster than Kademlia. An incentives scheme based on reputation can also reduce free riding, because the non-cooperating peers are isolated from the system. This fact has benefits both in faster downloading of files as non-receipt of corrupt files. Therefore, the reception of corrupted files from the misbehaving peers can be reduced or eliminated. Each peer uses its reputation rings and reputation score in order to distribute its upload capacity among good peers. As future work, the author plans to extend proposed scheme to social networks based on P2P infrastructure. Another possible extension for this work could be addressed to the large P2P streaming systems, and to P2P cloud systems.

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Paper

Two Semantics of Trust Management Language with Negation

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Abstract—The family of Role-based Trust management languages is used for representing security policies by defining a formalism, which uses credentials to handle trust in decentralized, distributed access control systems. A credential provides information about the privileges of users and the security policies issued by one or more trusted authorities. The main topic of this paper is $RT_{\ominus}$, a language which provides a carefully controlled form of non-monotonicity. The core part of the paper defines two different semantics of $RT_{\ominus}$ language – a relational, set-theoretic semantics for the language, and an inference system, which is a kind of operational semantics. The set-theoretic semantics maps roles to a set of entity names. In the operational semantics credentials can be derived from an initial set of credentials using a set of inference rules. The soundness and the completeness of the inference system with respect to the set-theoretic semantics of $RT_{\ominus}$ will be proven.

Keywords—access control, inference system, monotonicity, Role-based Trust management, set-theoretic semantics.

1. Introduction

Confidential data, whether in electronic, paper or other form must be properly protected. Guaranteeing that the data and services offered by a computer system are not made available to unauthorized users is an increasingly significant and challenging issue, which must be solved by reliable software technologies. This problem is usually solved by implementing access control techniques. In a typical access control scenario there are two entities, one is the requester that wants to access a protected resource, while the other is an entity that is the resource owner or provider. Usually these are the only entities involved in making the authorization decision. This approach fits well into closed, centralized environments, in which the identity of users is known in advance. Unfortunately, this simple scenario does not apply to highly distributed and decentralized networks. Quite different challenges arise in such decentralized and open systems, where the identity of users is not known in advance and the set of users can change. In decentralized environments, the resource owner and the requester often are unknown to one another, making access control based on identity ineffective. To be able to deal with different requesters coming from different security domains, we need a more flexible solution, named *trust management*.

Trust management is an approach to access control in decentralized distributed systems, where access control decisions are based on policy statements made by multiple principals. The potential and flexibility of trust management approach stems from the possibility of *delegation*: a principal may transfer limited authority over a resource to other principals. Such a delegation is implemented by means of an appropriate credential. This way, a set of credentials defines the access control strategy and allows deciding on who is authorized to access a resource, and who is not.

Despite the fact that the credentials-based models do, to a large degree, solve the access control problem in open systems, they still have some shortcomings. Most trust management languages are *monotonic*: adding new assertion to a query can never result in canceling an action, which was accepted before [1]. It is a problem, because each policy statement or credential added to the system can only increase the capabilities and privileges granted to others. The monotonicity property can simplify the design and analysis of complex network-based security protocols. It is a good property for researching, analyzing and proving, but causes limited usability, because revocation of privileges is not possible to assert. However, we can find in a literature various extensions of basic languages and models that create negation on different levels. And thus, in this way we achieve a non-monotonicity. One of the extensions is Role-based Trust management language with negation.

The rest of the paper is organized as follows. An overview of the work related to trust management systems and languages is given in Section 2. Section 3 shows the overview of the family of Role-based Trust management languages, Section 4 describes set-theoretic semantics of *RT* languages, including an example, and inference system over *RT* credentials with example is shown in Section 5. Final remarks are given in Conclusions.

2. Related Work

Traditional access control systems often rely on Role-Based Access Control (RBAC) model [2], which groups the access rights by the role name and limits the access to a resource to those users, who are assigned to a particular role. It is the most flexible type of access control policy.

The first trust management application described in the literature was PolicyMaker [3], which defined a special assertion language capable of expressing policy statements, which were locally trusted, and credentials, which had to be signed using a private key. The next generation of trust management languages were KeyNote [4], which was an enhanced version of PolicyMaker, SPKI/SDSI [5] and a few other languages [6]. All those languages allowed assigning privileges to entities and used credentials to delegate permissions from its issuer to its subject. What was missing in those languages was the possibility of delegation based on attributes of the entities and not on their identity. Responding to this need, a family of Role-based Trust management languages has been introduced in [7]–[9]. These languages have a well-defined syntax and semantics, which made them easy to extend in order to apply them to different needs.

A set-theoretic semantics, which defines the meaning of a set of credentials as a function from the set of roles into the power set of entities, has been defined for RT_0 [10], [9] and relational semantics, which apply to RT^T in [11].

One of the extensions of RT languages is the use of time validity constraints of the credentials, which made the languages of the RT family more realistic, because in the real world permissions are usually given just for a limited period of time. Time-dependant credentials were introduced in [10] (for RT_0) and in [12] (for RT^T). This type of time constraints can eliminate the need of non-monotonic system in some cases. Another approach to the monotonicity feature is an extension of RT_0 language, which was created to manage trust in P2P applications and access control in virtual communities described in [1].

3. Role-based Trust Management Languages

The term of *trust management* was introduced in year 1996 by Blaze *et al.* in [13], who defined it as a unified approach to specify and interpret security policies, credentials and trust relationships. In a trust management system an entity's privilege is based on its attributes instead of its identities. An entity's attributes are demonstrated through digitally signed credentials issued by multiple principals. A *credential* is an attestation of qualification, competence or authority issued to an individual by a third party. Examples of credentials in real life include identification documents, driver's licenses, membership cards, keys, etc. A credential in a computer system can be a digitally signed document.

RT is a family of Role-based Trust management languages, which combines trust management and RBAC features. To define a trust management system, a language is needed for describing entities (principals and requesters), credentials and roles, which the entities play in the system. $\mathbf{RT_0}$ is a simple yet powerful trust management language. It is the core language of RT family, described in detail in [9]. It allows describing localized authorities for roles, role hierarchies, delegation of authority over roles and role intersections. All the subsequent languages add new features to RT_0, they are progressively increasing in expressive power and complexity. $\mathbf{RT_1}$ introduces *parameterized roles*, which can represent relationships between entities. $\mathbf{RT_2}$ extends RT_1 with *logical objects*, which can be used to represent permissions given to entities with respect to a group of logically related objects (resources). Those extensions can help in keeping the notation concise, but do not increase the expressive power of the language, because each combination of parameters in RT_1 and each permission to a logical object in RT_2 can be defined alternatively as a separate role in RT_0. $\mathbf{RT^T}$ language has been introduced to support threshold and separation of duties policies. Similar to a role, which defines a set of principals, a manifold role defines a set of principal sets, each of which is a set of principals whose cooperation satisfies the manifold role. $\mathbf{RT^D}$ provides mechanism to describe delegation of rights and role activations, which can express selective use of capacities and delegation of these capacities, which are useful when one wants to delegate authority temporarily. In many scenarios, an entity prefers not to use all his privileges, all the more, to delegate all his rights. $\mathbf{RT_\ominus}$ provides a carefully controlled form of non-monotonicity. The members of the RT family presented so far are monotonic: adding a credential to the system can only result in granting additional privileges, it cannot result in canceling an action, which was accepted before. It is a problem, because each policy statement or credential added to the system can only increase the capabilities and privileges granted to others. In [1], Czenko *et al.* argue that many access control decisions in complex distributed systems, like virtual communities, are hard to model in a purely monotonic language. They propose $RT_\ominus$, which adds to RT a restricted form of negation called *negation in context*. $RT_\ominus$ introduces a new operator $\ominus$ and the so called *exclusion* credential. It was created to manage trust in P2P applications and access control in virtual communities. The features of RT^T and RT^D can be combined together with the features of RT_0, RT_1 or RT_2. A more detailed treatment of RT family can be found in [8].

3.1. The Syntax of RT *Family Languages*

Basic elements of RT languages are *entities*, *role names*, *roles* and *credentials*. Entities represent principals that can define roles and issue credentials, and requesters that can make requests to access resources. An entity can, e.g., be a person or program identified by a user account in a computer system or a public key. We denote an entity by a name starting with an uppercase letter (or just an uppercase letter), e.g., A, $Alice$, $University$. Role names represent permissions that can be issued by entities to other entities or groups of entities. A role name is denoted by a string starting with a lowercase letter (or just a lowercase letter), like r or $student$. Roles denote sets of entities that have particular permissions granted according to the access control policy. Roles have the form of entity followed by a role name, separated by a dot, like $A.r$ or $University.student$. Credentials define roles by appointing a new member of the role or by delegating authority to the members of other roles. There are four types of credentials in RT_0, which are interpreted in the following way:

- $A.r \leftarrow B$ – *simple membership*: entity B is a member of role $A.r$;
- $A.r \leftarrow B.s$ – *simple inclusion*: role $A.r$ includes (all members of) role $B.s$. This is a delegation of authority over r from A to B;
- $A.r \leftarrow B.s.t$ – *linking inclusion*: role $A.r$ includes role $C.t$ for each C, which is a member of role $B.s$. This is a delegation of authority from A to all the members of the role $B.s$;
- $A.r \leftarrow B.s \cap C.t$ – *intersection inclusion*: role $A.r$ includes all the entities who are members of both roles $B.s$ and $C.t$. This is a partial delegation from A to B and C;
- $A.r \leftarrow B.s \ominus C.t$ – *exclusion*: all members of $B.s$ which are not members of $C.t$ are members of $A.r$.

A policy is a finite set of credentials.

The languages discussed in this paper can be used, in general, in very complex systems. Therefore, we present here only a simplified example, with the intention to illustrate the basic notions and the notation, with a focus on $RT_\ominus$ credentials.

Example 1. Suppose that John wants to share his pictures and movies using file sharing system. John restricts the access to his pictures to those of his friends, who are a members of Picture Club and he gave similar restriction to his movie, but he requires that his friends should be members of Movie Club instead of Picture Club.
John can also create a list of people who are forbidden to see the gallery of his private pictures, which means that people, who are on the list can see the general gallery of his pictures but not the private one.
The entire policy can be expressed as follows:

$$John.accessPic \leftarrow John.friend \cap John.pictureClub \quad (1)$$

$$John.accessMov \leftarrow John.friend \cap John.movieClub \quad (2)$$

$$John.privatePic \leftarrow John.accessPic \ominus John.blackList \quad (3)$$

Now, assume that the following credentials have been added:

$$John.friend \leftarrow Bob \quad (4)$$

$$John.friend \leftarrow Lily \quad (5)$$

$$John.friend \leftarrow Maria \quad (6)$$

$$John.friend \leftarrow Sofia \quad (7)$$

$$John.pictureClub \leftarrow Bob \quad (8)$$

$$John.pictureClub \leftarrow Etan \quad (9)$$

$$John.pictureClub \leftarrow Lily \quad (10)$$

$$John.movieClub \leftarrow Alice \quad (11)$$

$$John.movieClub \leftarrow Maria \quad (12)$$

$$John.movieClub \leftarrow Sofia \quad (13)$$

$$John.blackList \leftarrow Bob \quad (14)$$

Then one can conclude that, according to the policy, people who have access to John's pictures are *Bob* and *Lily*, but only *Lily* has access to his private gallery, and *Maria* and *Sofia* have access to John's movies.

4. The Set-Theoretic Semantics of *RT* Languages

A set-theoretic semantics of RT_0, which defines the meaning of a set of credentials as (monotone) function from the set of roles into the power set of entities, has been originally defined in [9]. A slightly different approach, closely related to the semantics of RT_0 language, was shown in [14], where various semantic readings of Simple Distributed Security Infrastructure (SDSI) were provided. A definition quoted in this section is a modified version of the same semantics, which has been introduced in [10], with addition of $\ominus$ operator.

Definition 1. The semantics of a set $\mathscr{P}$ of *RT* credentials, denoted by $\mathscr{S}_{\mathscr{P}}$, is the smallest relation S_i, such that:

1. $\mathscr{S}_0 = \emptyset$
2. $\mathscr{S}_{i+1} = \bigcup_{c \in \mathscr{P}} f(\mathscr{S}_i, c)$ for $i = 0, 1, \ldots$

which is closed with respect to function f, which describes the meaning of credentials in the following way (A, B, C, X, Y are entities):

$$
\begin{aligned}
& f(\mathscr{S}_i, A.r \leftarrow X) = \{(A, r, X)\} \\
& f(\mathscr{S}_i, A.r \leftarrow B.s) = \{(A, r, X) : (B, s, X) \in \mathscr{S}_i\} \\
& f(\mathscr{S}_i, A.r \leftarrow B.s.t) = \textstyle\bigcup_{C:(B,s,C) \in \mathscr{S}_i} \{(A, r, X) : (C, t, X) \in \mathscr{S}_i\} \\
& f(\mathscr{S}_i, A.r \leftarrow B.s \cap C.t) = \{(A, r, X) : (B, s, X) \in \mathscr{S}_i \\
& \qquad \wedge (C, t, X) \in \mathscr{S}_i\} \\
& f(\mathscr{S}_i, A.r \leftarrow B.s \ominus C.t) = \{(A, r, X \setminus Y) : (B, s, X) \in \mathscr{S}_i \\
& \qquad \wedge (C, t, Y) \in \mathscr{S}_i\}
\end{aligned}
$$

Example 2. Set-theoretic semantics for Example 1

We use Example 1 from Section 3 to illustrate the definition of RT semantics.
The sequence of steps to compute consecutive relations $\mathscr{S}_i$ is shown in Table 1. Consecutive sections of the table describe relations $\mathscr{S}_0$ through $\mathscr{S}_3$. Each section of Table 1 has exactly one row, which corresponds to the issuer of the role, *John*. The columns of the table correspond to role names. This way, a cell of the table shows the set of entities, which are members of the respective role issued by *John*.
The starting relation $\mathscr{S}_0$ is, by definition, empty. According to Definition 1, only credentials (4)–(14) are

Table 1
The relations $\mathscr{S}_0$ through $\mathscr{S}_4$

John	friend	pictureClub	movieClub	blackList	accessPic	accessMov	privatePic
$\mathscr{S}_0$	ϕ	ϕ	ϕ	ϕ	ϕ	ϕ	ϕ
$\mathscr{S}_1$	*Bob* *Lily* *Maria* *Sofia*	*Bob* *Etan* *Lily*	*Alice* *Maria* *Sofia*	*Bob*	ϕ	ϕ	ϕ
$\mathscr{S}_2$	*Bob* *Lily* *Maria* *Sofia*	*Bob* *Etan* *Lily*	*Alice* *Maria* *Sofia*	*Bob*	*Bob* *Lily*	*Maria* *Sofia*	ϕ
$\mathscr{S}_3$	*Bob* *Lily* *Maria* *Sofia*	*Bob* *Etan* *Lily*	*Alice* *Maria* *Sofia*	*Bob*	*Bob* *Lily*	*Maria* *Sofia*	*Lily*

mapped in $\mathscr{S}_0$ into nonempty sets by function f. These sets are shown in relation $\mathscr{S}_1$ in Table 1. In $\mathscr{S}_1$, credentials (1) and (2) are mapped into instances $(John, accessPic, Bob)$, $(John, accessPic, Lily)$, $(John, accessMov, Maria)$, and $(John, accessMov, Sofia)$ of relation $\mathscr{S}_2$, and in $\mathscr{S}_2$, credential (3) is mapped into instances $(John, privatePic, Lily)$. The resulting relation $\mathscr{S}_3$ cannot be changed using the given set of credentials, hence

$$\mathscr{S}_{\mathscr{P}} = \mathscr{S}_3$$

and it is the end of the set-theoretic semantics for Example 1.

5. The Inference System over *RT* Credentials

The member sets of roles can also be calculated in a more convenient way (than set-theoretic semantics) using an inference system, which defines an operational semantics of *RT* languages. An inference system consists of an initial set of formulae that are considered to be true, and a set of inference rules, that can be used to derive new formulae from the known ones.

Let $\mathscr{P}$ be a given set of *RT* credentials. The application of inference rules of the inference system will create new credentials, derived from credentials of the set $\mathscr{P}$. A derived credential c will be denoted using a formula $\mathscr{P} \succ c$, which should be read: credential c can be derived from a set of credentials $\mathscr{P}$.

Definition 2. The initial set of formulae of an inference system over a set $\mathscr{P}$ of *RT* credentials are all the formulae

$$c \in \mathscr{P},$$

for each credential c in $\mathscr{P}$.

The inference rules of the system are the following:

$$\frac{c \in \mathscr{P}}{\mathscr{P} \succ c} \qquad (W_1)$$

$$\frac{\mathscr{P} \succ A.r \leftarrow B.s \qquad \mathscr{P} \succ B.s \leftarrow X}{\mathscr{P} \succ A.r \leftarrow X} \qquad (W_2)$$

$$\frac{\mathscr{P} \succ A.r \leftarrow B.s.t \qquad \mathscr{P} \succ B.s \leftarrow C \qquad \mathscr{P} \succ C.t \leftarrow X}{\mathscr{P} \succ A.r \leftarrow X} \qquad (W_3)$$

$$\frac{\mathscr{P} \succ A.r \leftarrow B.s \cap C.t \qquad \mathscr{P} \succ B.s \leftarrow X \qquad \mathscr{P} \succ C.t \leftarrow X}{\mathscr{P} \succ A.r \leftarrow X} \qquad (W_4)$$

$$\frac{\mathscr{P} \succ A.r \leftarrow B.s \ominus C.t \qquad \mathscr{P} \succ B.s \leftarrow X \qquad \mathscr{P} \succ C.t \leftarrow Y}{\mathscr{P} \succ A.r \leftarrow X \setminus Y} \qquad (W_5)$$

The five kinds of credentials described in Section 4 are handled by the rules above. The rules should be self-explanatory.

5.1. Soundness and Completeness of Inference System over RT^T Credentials

There could be a number of inference systems defined over a given language. To be useful for practical purposes an inference system must exhibit two properties. First, it should be sound, which means that the inference rules could derive only formulae that are valid with respect to the semantics of the language. Second, it should be complete, which means that each formula, which is valid according to the semantics, should be derivable in the system.

Due to space constraints, we only present sketches of proofs and proofs for (W_5) formula (introduced in the $RT_{\ominus}$ language), full proofs for the rest formulae can be found in [15]. The semantics of a set $\mathscr{P}$ of *RT* credentials, defined by the inference system, is given by a set of all the formulae of the type: $\mathscr{P} \succ A.r \leftarrow X$.

To prove the soundness of such a formula, one must prove that the triple (A,r,X) belongs to the semantics $\mathscr{S}_{\mathscr{P}}$ of the set of credentials $\mathscr{P}$. Let us first note that all the formulae $\mathscr{P} \succ A.r \leftarrow X$, such that $A.r \leftarrow X \in \mathscr{P}$ are sound. This is proved in Lemma 1.

Lemma 1. If $A.r \leftarrow X \in \mathscr{P}$ then $(A,r,X) \in \mathscr{S}_{\mathscr{P}}$.

Proof. The relation $\mathscr{S}_{\mathscr{P}}$, which defines the semantics of $\mathscr{P}$, is a limit of a monotonically increasing sequence of sets $\mathscr{S}_0, \mathscr{S}_1, \ldots$, such that $\mathscr{S}_0 = \emptyset$. According to Definition 1: $f(\mathscr{S}_0, A.r \leftarrow X) = (A,r,X)$. Hence,$(A,r,X) \in \mathscr{S}_1$ and because $\mathscr{S}_1 \subseteq \mathscr{S}_{\mathscr{P}}$ then $(A,r,X) \in \mathscr{S}_{\mathscr{P}}$.

■

To prove the soundness of the inference system over $\mathscr{P}$, we must prove the soundness of each formula $\mathscr{P} \succ A.r \leftarrow X$, which can be derived from the set $\mathscr{P}$. This is proven in Theorem 1.

Theorem 1. If $\mathscr{P} \succ A.r \leftarrow X$ then $(A,r,X) \in \mathscr{S}_{\mathscr{P}}$.

Proof. By induction with respect to the number n of inference steps, which are needed to derive a formula $\mathscr{P} \succ A.r \leftarrow X$. If $n = 1$ then the formula $\mathscr{P} \succ A.r \leftarrow X$ could be derived only using rule (W_1), because the premises of only this rule are axioms. Hence, the thesis is true according to Lemma 1. For the inductive step, assume that the thesis is true if the number of inference steps was not greater than n. Then, it is possible to show that it is true also in a case when the number of inference steps equals $n+1$. Since any one of the rules (W_2) through (W_5) could be used in the last $(n+1)$ step of inference, all those four cases should be discussed separately, analyzing the premises and using Definition 1 to show that the thesis holds. As it was mentioned before, a proof for the rule (W_5) is provided, see [15] for rules (W_2)–(W_4).

$(\mathbf{W_5})$ The fist premise of (W_5) cannot be derived otherwise than using (W_1). Hence, $A.r \leftarrow B.s \ominus C.t \in \mathscr{P}$. The second premise of (W_5): $\mathscr{P} \succ B.s \leftarrow X$ was derived from $\mathscr{P}$ using at most n steps of inference, hence, $(B,s,X) \in \mathscr{S}_{\mathscr{P}}$ according to the inductive hypothesis. By Definition 1, there exists such $\mathscr{S}_i$ that $(B,s,X) \in \mathscr{S}_i$. Similarly, in the case of the third premise of (W_5): $\mathscr{P} \succ C.t \leftarrow Y$, there exists such $\mathscr{S}_j$ that $(C,t,Y) \in \mathscr{S}_j$. Let k be the maximum of (i,j). Then $(B,s,X) \in \mathscr{S}_k$, $(C,t,Y) \in \mathscr{S}_k$ and $(A,r,X \setminus Y) \in f(\mathscr{S}_k, A.r \leftarrow B.s \ominus C.t)$ according to $f(\mathscr{S}_i, A.r \leftarrow B.s \ominus C.t) = \{(A,r,X \setminus Y) : (B,s,X) \in \mathscr{S}_i \wedge (C,t,Y) \in \mathscr{S}_i\}$. Because $f(\mathscr{S}_k, A.r \leftarrow B.s \ominus C.t) \subseteq \mathscr{S}_{k+1} \subseteq \mathscr{S}_{\mathscr{P}}$ then $(A,r,X \setminus Y) \in \mathscr{S}_{\mathscr{P}}$.

■

To prove the completeness of the inference system over a set $\mathscr{P}$ of RT credentials, one must prove that a formula $\mathscr{P} \succ A.r \leftarrow X$ can be derived using inference rules for each element $(A,r,X) \in \mathscr{S}_{\mathscr{P}}$. This is proven in Theorem 2.

Theorem 2. If $(A,r,X) \in \mathscr{S}_{\mathscr{P}}$ then $\mathscr{P} \succ A.r \leftarrow X$.

Proof. Assume $(A,r,X) \in \mathscr{S}_{\mathscr{P}}$. By Definition 1, there exists such $i \geq 0$ and such $c \in \mathscr{P}$ that $(A,r,X) \in f(\mathscr{S}_i, c)$. The proof of the thesis is by induction with respect to the value of index i. If $i = 0$ then credential c must take the form of $A.r \leftarrow X$. This is because $\mathscr{S}_0 = \emptyset$ and $f(\mathscr{S}_0, d) = \emptyset$ for each credential d other than $A.r \leftarrow X$. Hence, $A.r \leftarrow X \in \mathscr{P}$ and the formula $\mathscr{P} \succ A.r \leftarrow X$ can be derived using rule (W_1). For the inductive step, assume that the thesis is true, if the value of index i in the expression $(A,r,X) \in f(\mathscr{S}_i, c)$ was not greater than n. Then it suffices to show that it is true also in a case when the value of index i equals $(n+1)$. Assume $(A,r,X) \in \mathscr{S}_{\mathscr{P}}$ and $(A,r,X) \in f(\mathscr{S}_{n+1}, c)$ for a certain $c \in \mathscr{P}$. The credential c can take one of the five forms allowed in RT_0 and $RT_{\ominus}$. Each of these types of credentials should be discussed separately, showing that it can be derived using one of the rules (W_1)–(W_5). Definition 1 is used in all cases except $\mathbf{c = A.r \leftarrow B}$, which trivially results from (W_1). As it was mentioned before, a proof for $c = A.r \leftarrow B.s \ominus C.t$ is provided, see [15] for rules (W_2)–(W_4).

$\mathbf{c = A.r \leftarrow B.s \ominus C.t}$: If $(A,r,X) \in f(\mathscr{S}_{n+1}, A.r \leftarrow B.s \ominus C.t)$, then according to Definition 1, $f(\mathscr{S}_i, A.r \leftarrow B.s \ominus C.t) = \{(A,r,X \setminus Y) : (B,s,X) \in \mathscr{S}_i \wedge (C,t,Y) \in \mathscr{S}_i\}$, there exist two sets of entities Z, Y such that $Z \setminus Y = X$ and $(B,s,Z) \in \mathscr{S}_{n+1}$ and $(C,t,Y) \in \mathscr{S}_{n+1}$. Hence, there exist credentials c_1, c_2 such that $(B,s,Z) \in f(\mathscr{S}_n, c_1)$ and $(C,t,Y) \in f(\mathscr{S}_n, c_2)$. This implies that $(B,s,Z) \in \mathscr{S}_{\mathscr{P}}$ and $(C,t,Y) \in \mathscr{S}_{\mathscr{P}}$, hence, $\mathscr{P} \succ B.s \leftarrow Z$ and $\mathscr{P} \succ C.t \leftarrow Y$ according to the inductive hypothesis. $\mathscr{P} \succ A.r \leftarrow X$ is a conclusion of rule (W_5).

■

A conclusion from Theorem 1 and Theorem 2 is such that the inference system of Definition 1 is sound and complete with respect to the semantics of RT credentials. This way, the inference system gives an operational definition of RT semantics (for RT_0 and $RT_{\ominus}$) and it proves that the inference system provides an alternative way of presenting the semantics of RT.

Example 3. (Inference system for Example 1):

We use the inference system to formally derive entities which can have access to $John's$ galleries. Using credentials (1)–(14) according to rule (W_1) it can infer:

$$\frac{John.accessPic \leftarrow John.friend \cap John.picClub \in \mathscr{P}}{\mathscr{P} \succ John.accessPic \leftarrow John.friend \cap John.picClub}$$

$$\frac{John.accessMov \leftarrow John.friend \cap John.movieClub \in \mathscr{P}}{\mathscr{P} \succ John.accessMov \leftarrow John.friend \cap John.movieClub}$$

$$\frac{John.privatePic \leftarrow John.accessPic \ominus John.blackList \in \mathscr{P}}{\mathscr{P} \succ John.privatePic \leftarrow John.accessPic \ominus John.blackList}$$

$$\frac{John.friend \leftarrow Bob \in \mathscr{P}}{\mathscr{P} \succ John.friend \leftarrow Bob}$$

$$\frac{John.friend \leftarrow Lily \in \mathscr{P}}{\mathscr{P} \succ John.friend \leftarrow Lily}$$

$$\frac{John.friend \leftarrow Maria \in \mathscr{P}}{\mathscr{P} \succ John.friend \leftarrow Maria}$$

$$\frac{John.friend \leftarrow Sofia \in \mathscr{P}}{\mathscr{P} \succ John.friend \leftarrow Sofia}$$

$$\frac{John.pictureClub \leftarrow Bob \in \mathscr{P}}{\mathscr{P} \succ John.pictureClub \leftarrow Bob}$$

$$\frac{John.pictureClub \leftarrow Etan \in \mathscr{P}}{\mathscr{P} \succ John.pictureClub \leftarrow Etan}$$

$$\frac{John.pictureClub \leftarrow Lily \in \mathscr{P}}{\mathscr{P} \succ John.pictureClub \leftarrow Lily}$$

$$\frac{John.movieClub \leftarrow Alice \in \mathscr{P}}{\mathscr{P} \succ John.movieClub \leftarrow Alice}$$

$$\frac{John.movieClub \leftarrow Maria \in \mathscr{P}}{\mathscr{P} \succ John.movieClub \leftarrow Maria}$$

$$\frac{John.movieClub \leftarrow Sofia \in \mathscr{P}}{\mathscr{P} \succ John.movieClub \leftarrow Sofia}$$

$$\frac{John.blackList \leftarrow Bob \in \mathscr{P}}{\mathscr{P} \succ John.blackList \leftarrow Bob}$$

Then, using credentials (1), (4), (5), (8), (10), and rule (W_4) we infer:

$$\frac{\begin{array}{c}\mathscr{P} \succ John.accessPic \leftarrow John.friend \cap John.pictureClub \\ \mathscr{P} \succ John.friend \leftarrow Bob \qquad \mathscr{P} \succ John.pictureClub \leftarrow Bob\end{array}}{\mathscr{P} \succ \mathbf{John.accessPic} \leftarrow \mathbf{Bob}}$$

$$\frac{\begin{array}{c}\mathscr{P} \succ John.accessPic \leftarrow John.friend \cap John.pictureClub \\ \mathscr{P} \succ John.friend \leftarrow Lily \qquad \mathscr{P} \succ John.pictureClub \leftarrow Lily\end{array}}{\mathscr{P} \succ \mathbf{John.accessPic} \leftarrow \mathbf{Lily}}$$

showing that the group of people who can see John's pictures are Bob and $Lily$.

In the next step the newly inferred credentials and additionally credentials (3) and (14) with the rule (W_5) is used:

$$\frac{\begin{array}{c}\mathscr{P} \succ John.privatePic \leftarrow John.accessPic \ominus John.blackList \\ \mathscr{P} \succ John.accessPic \leftarrow Bob \qquad \mathscr{P} \succ John.accessPic \leftarrow Lily \\ John.blackList \leftarrow Bob\end{array}}{\mathscr{P} \succ \mathbf{John.privatePic} \leftarrow \mathbf{Lily}}$$

showing that only $Lily$ can see John's private collection.

Additionally, if we want to find a group of people, who can see John's movies, we can do this using credentials (2), (6), (7), (12), (13), and rule (W_4). We infer:

$$\frac{\begin{array}{c}\mathscr{P} \succ John.accessMov \leftarrow John.friend \cap John.movieClub \\ \mathscr{P} \succ John.friend \leftarrow Maria \qquad \mathscr{P} \succ John.movieClub \leftarrow Maria\end{array}}{\mathscr{P} \succ \mathbf{John.accessMov} \leftarrow \mathbf{Maria}}$$

$$\frac{\begin{array}{c}\mathscr{P} \succ John.accessMovie \leftarrow John.friend \cap John.movieClub \\ \mathscr{P} \succ John.friend \leftarrow Sofia \qquad \mathscr{P} \succ John.movieClub \leftarrow Sofia\end{array}}{\mathscr{P} \succ \mathbf{John.accessMov} \leftarrow \mathbf{Sofia}}$$

showing that the group of people who can see John's movies are $Maria$ and $Sofia$.

6. Conclusions

This paper deals with modeling of trust management systems in decentralized and distributed environments. The modeling framework is a family of Role-based Trust management languages, especially RT_0 and $RT_\ominus$ languages. Two types of semantics for RT credentials have been introduced in the paper.

A set-theoretic semantics of RT languages is defined as a relation over a set of roles and a set of entities.

An operational semantics of RT languages is defined as an inference system, in which credentials can be derived from an initial set of credentials using a set of inference rules. The semantics is given by the set of resulting credentials of the type $A.r \leftarrow X$, which explicitly show a mapping between roles and sets of entities.

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