Slovak Electrical Society Section with the Department of Telecommunications
University of Žilina, Slovakia

Faculty of Electrical Engineering University of Žilina, Slovakia

Department of Telecommunications and Multimedia, Faculty of Electrical Engineering University of Žilina, Slovakia

Department of Circuit Theory, Czech Technical University in Prague, Czech Republic

Faculty of Management Science and Informatics, University of Žilina, Slovakia

Digital Technologies 2009

6th International Workshop

November 12-13, 2009

Žilina, Slovakia

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The Workshop is organised by:

Slovak Electrical Society Section with the Department of Telecommunications University of Žilina, Slovakia

Faculty of Electrical Engineering University of Žilina, Slovakia

Department of Telecommunications and Multimedia, Faculty of Electrical Engineering, University of Žilina, Slovakia

Faculty of Management Science and Informatics, University of Žilina, Slovakia

Department of Circuit Theory, Czech Technical University in Prague, Czech Republic

In cooperation with:

United Institute of Informatics Problems of the National Academy of Sciences of Belarus, Belarus

Belarusian State University, Belarus

Under the auspices of:

Association of Slovak Scientific and Technological Societies

Slovak Electrotechnical Society

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Workshop program

Place: **Stravovacie zariadenie Žilinskej univerzity v Žiline** (Canteen, University Campus)
Vysokoškolákov 26, 010 08 Žilina, SLOVAKIA

Room: Zasadačka vedeckej Rady (Room of the University Scientific Board)

Date: **November 12, 2009**
Schedule

08:00-08:45	**REGISTRATION IN ATRIUM**
08:45-10:15	**INFORMATION SYSTEMS AND KNOWLEDGE MANAGEMENT I.**
10:15-10:30	COFFEE BREAK
10:30-12:00	**SPEECH PROCESSING**
12:00-12:45	LUNCH
12:45-14:15	**COMMUNICATIONS NETWORKS I.**
14:15-14:30	COFFEE BREAK
14:30-15:40	**COMMUNICATIONS NETWORKS II.**
15:40-15:50	COFFEE BREAK
15:50-17:00	**COMMUNICATIONS NETWORKS III.**
17:00	SOCIAL EVENT

Date: **November 13, 2009**
Schedule

08:00-08:30	**REGISTRATION IN ATRIUM**
08:30-10:00	**IMAGE AND BIOSIGNALS PROCESSING**
10:00-10:15	COFFEE BREAK
10:15-11:50	**CIRCUIT TECHNOLOGIES**
11:50-13:15	LUNCH
13:15-15:00	**COMMUNICATIONS NETWORKS IV.**
15:00-15:15	COFFEE BREAK
15:15-17:00	**INFORMATION SYSTEMS AND KNOWLEDGE MANAGEMENT II.**
17:00	**CLOSING OF THE WORKSHOP**

TECHNICAL PROGRAM, DATE: November 12, 2009

08:45-10:15	**INFORMATION SYSTEMS AND KNOWLEDGE MANAGEMENT I.**
Novoselova, N.A.- Tom, I.E.:	METHOD OF EVOLUTIONARY FUZZY CLUSTERING
Chmulík, M.- Paleček, J.:	ON PARTICLE SWARM OPTIMIZATION AND ITS COMPARISON WITH GA
Šulhin, A.- Solenský, P.- Levashenko, V.:	ANALYSIS OF ALGORITHMS OF DECISION-TREES MAKING
Sebesta, J.:	CASCADE INTEGRATOR-COMB FILTER FREQUENCY RESPONSE OPTIMIZATION
Kútna, A.- Palásthy, H.:	PRINCIPLE OF ELECTRONIC STUDY MATERIALS CREATION
Nedvěd, J.:	LARGE MEASUREMENT DATA SET AUTOMATED PROCESSING

10:30-12:00	**SPEECH PROCESSING**
Hric, M.- Jarina, R.:	AUTOMATIC SPEAKER IDENTIFICATION USING GMM WITH LIMITED AMOUNT OF TRAINING DATA
Bartů, M.:	SPEECH PARAMETRIZATION FOR OPERATOR MAPS
Tučková, J.:	EMOTIONS AS A COMPONENT OF PROSODY
Zetocha, P.:	SELF-ORGANIZING MAP IN DEVELOPMENTAL DYSPHASIA ANALYSIS
Krejčí, R.- Hanžl, V.:	SPEECH PROCESSING ALGORITHMS ON HARDWARE PLATFORM TEXAS INSTRUMENTS OMAP-L137

TECHNICAL PROGRAM, DATE: November 12, 2009

12:45-14:15	**COMMUNICATIONS NETWORKS I.**
Serafin, J.- Brabec, Z.:	OPERATION OF IT/ICT PRACTICE FRAMEWORKS
Kocur, Z.- Pravda, M.- Vodrazka, J.:	ENHANCEMENT ETHERNET NETWORK SIMULATOR
Vanek, T.- Rohlik, M.:	SIMULATION OF THE SELECTED NETWORK ATTACKS ON THE TESLA AUTHENTICATION PROTOCOL
Machaj, J.:	SIMULATION MODEL FOR LOCALIZATION IN AD HOC NETWORKS
Paleček, J.- Vestenický, M.:	EMULATOR OF GPS RECEIVER
Jarinová, D.:	AUTOREGRESSIVE-BASED LONG TERM PREDICTION OF RAYLEIGH FADING CHANNEL
14:30-15:40	**COMMUNICATIONS NETWORKS II.**
Holub, J.- Veřmiřovský, R.:	NETWORK SIMULATOR FOR CONVERSATIONAL QUALITY MEASUREMENTS
Vodrážka, J.:	VOICE QUALITY ESTIMATION FOR NARROWBAND AND WIDEBAND PHONE COMMUNICATION
Schukat, M.- Connolly,C.:	A GSM-BASED INFRASTRUCTURE FOR LARGE GEOGRAPHICALLY DISTRIBUTED SPARSE SENSOR NETWORKS
Connolly,C.- Schukat, M.:	TRANSPARENT REMOTE WAVE DIVISION MULTIPLEXING PASSIVE OPTICAL ARCHITECTURE
15:50-17:00	**COMMUNICATIONS NETWORKS III.**
Shannon, J.- Melvin, H. :	SYNCHRONISATION CHALLENGES FOR WIRELESS SENSOR NETWORKS
Ó Flaithearta, P.- Melvin, H. :	NS-2 EXTENSION FOR DYNAMIC OPTIMISATION WITHIN MULTIPLE VoIP SESSIONS OVER WIRELESS NETWORKS
Yuste, L.B.- Melvin, H. :	INTER-MEDIA SYNCHRONISATION FOR IPTV: A CASE STUDY FOR VLC
Baroňák, I.- Janata, V.- Kovačik, M.- Janata, J.:	VIRTUAL NETWORK WITH MPLS SUPPORT

TECHNICAL PROGRAM, DATE: November 13, 2009

08:30-10:00	**IMAGE AND BIOSIGNALS PROCESSING**
Goncharov, D.- Nedzved, A.- Ablameyko, S.:	BUILDING VOLUMETRIC DISTANCE MAPS: ANALYSIS OF METHODS
Jelšovka, D.- Brezňan, M.:	3D MODEL RECONSTRUCTION OF THE 2D IMAGES BY VOXEL COLORING ALGORITHM
Kamencay, P.:	ALGORITHMS FOR RECONSTRUCTION OF 3-D IMAGE AND 3DTV TECHNOLOGIES
Bortel, R.- Sovka, P.- Chaloupka, Z.- Vondrášek, M.:	ROBUST HEART RATE ESTIMATION BASED ON IMPERFECT QRS DETECTION
Vondrášek, M.:	COMPARISON OF DIRECTIONAL HEARING FOR COCHLEAR IMPLANTS
Sprindzuk, M.V. et al.	COMPUTER-AIDED DIAGNOSIS OF THE PULMONARY NODULE
10:15-11:50	**CIRCUIT TECHNOLOGIES**
Valenta, M.- Martinek, P.:	IMPROVED CURRENT CONVEYOR OPTIMIZATION VIA DIFFERENTIAL EVOLUTION
Výrostko, M.:	METHODS OF KEY MANAGEMENT SYSTEMS USED IN SAFETY-RELATED APPLICATIONS
Žiška, P.:	ON THE DESIGN OF ESD PROTECTIONS
Vojtech, L.- Neruda, M.- Hajek, J.:	MEASUREMENT OF ELECTROMAGNETIC SHIELDING EFFICIENCY OF PLANAR TEXTILES
Šubrt, O.- Kubař, M.- Martinek, P.:	ON THE STATIC ADC NON-LINEARITY EVALUATED FOR RSD CYCLIC A/D CONVERTER CASE
Tichá, D.- Martinek, P.- Boreš, P.:	VIDEOFILTERS FOR DIGITAL TV
Hospodka, J.- Boreš, P.:	INTERACTIVE APPLICATION FOR ELECTRIC CIRCUIT ANALYSIS

TECHNICAL PROGRAM, DATE: November 13, 2009

13:15-15:00	**COMMUNICATIONS NETWORKS IV.**
Bešťák, R.- Hlaváček, J.:	NEW MODEL FOR THE SERVICE PROVIDERS: CAPABILITY MATURITY MODEL INTEGRATION FOR SERVICES
Benikovský, J.:	LOCALIZATION IN GSM NETWORKS USING RADIO FINGERPRINTING
Dolnák, I.- Kortiš, P.:	NETWORK ADDRESS TRANSLATION TRAVERSAL ISSUE IN VOIP NETWORK
Martyscak, V.:	DRAFT FOR METHODOLOGY FOR COMPARING THE STATE OF ICT INFRASTRUCTURE IN DEVELOPED WORLDWIDE COUNTRIES BASED ON KNOWLEDGE OF TREND
Markovič, M.- Dubovan,J.:	SIMULATION OF SEGMENT-BASED ROBUST FAST OPTICAL RESERVATION PROTOCOL
Frič, M.- Hudec, R.:	DETECTION OF SPECTRAL PROPERTIES OF THE SPECIAL FIBERS
15:15-17:00	**INFORMATION SYSTEMS AND KNOWLEDGE MANAGEMENT II.**
Ladovský, T.:	THE MATRIX PERMUTATION PROBLEM IN FUZZY ARITHMETIC
Adamko, D.- Orgoň, M.:	TOOLS FOR IMPLEMENTATION OF AUDIT SOFTWARE
Pottosina, S.A.- Kovalenko, I.V. :	SCORING SYSTEM FOR EXTRACTION KNOWLEDGE TO MAKE DECISIONS FOR GIVING CREDIT
Zhelezko, B.A.- Podgornaya, G.N.- Skrebneva, Y.V.:	PERFECTION OF INFORMATION INFRASTRUCTURE OF RETAIL TRADE NETWORK
Shved, O.- Zhivitskaya, H.:	THE EXPERT INFORMATION SYSTEM FOR THE THIRD PARTY LOGISTIC PROVIDER'S CHOICE
Zaitseva, E.- Kamensky, M.:	ESTIMATION OF MULTI-STATE SYSTEM RELIABILITY

Tools for Implementation of Audit Software

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Contribution entitled "Tools for the implementation of audit software" deals with the optimal procedure for software auditing and software management.

Under the phrase "software audit" means a release of software (inventory, licenses). It consists of several steps that are to be followed for the proper conduct software audits. The task of software auditing is to detect possession of legitimate software installed in companies. An integral part of the knowledge audit software portfolio. Based on the analysis of knowledge to propose a flowchart, which will provide optimal solutions software audit steps. The contribution is carefully analyzed in detail the various steps software audit. Flowchart consists of several steps that must be designed for different companies for different structures in society.

Obtaining information applied various steps software audit, whereby the achieved results which are compared with the facts of the licenses. To obtain correct results it is necessary to use audit software, which is subject to contribution. The paper also discussed the issue alignment state licenses. This step is an integral part of the software audit, whereby it is possible that "illegal" software to validate. The application of such action is necessary to know the licensing policies of software companies.

An integral part of the audit software is a software management. It serves us as a means to coordinate the software policy within the company. The work contained the basic steps of management software designed to improve the licensing of the facts in the society and also provides recommendations for the conservation of the execution audit software, which saves considerable funds for the purchase of new software.

Output Final is a software audit report, which consists of information received, incorporated therein results and recommendations of software management, evaluate the information acquired after the execution of a software audit and not least the final report provides the facts, the risks may be associated with retention of facts and recommendations to be followed to maintain the state after a software audit. The work included recommendations consist of formalization of software management procedures, elimination of interference in the workstation using the central management and control of the efficient use of software management.

In conclusion, the work is evaluated software audits carried out in Slovakia and the graphically illustrated results of these audits. Based on the analysis of the final audit reports have suggested some changes in the implementation of software audits and refinement of tools contained in the software management.

Virtual Network with MPLS Support

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Now, in age of real-time multimedia services and traffic, the interest is taken on questions around Quality of Service (QoS) and its implementation. In stage of Next Generation Network (NGN), Multiprotocol Label Swiching (MPLS) can be introduced as one of technologies used as a Transport Layer technology. With its ability to provide connection oriented transfer mode via non-connection oriented underlayerd technology and support of RSVP-TE and traffic engineering, MPLS is capable to satisfy QoS requirements of today multimedia services and providers. Diffserv-MPLS and Virtual Private Network VPN via MPLS are most known services that can be supported by MPLS networks.

In stage of studying transport layer technologies, simulation and emulation tools are mostly used, because of unavailability of these technologies from view of practical experience. This simulation tools are mostly developed with major interest taken into statistics results. For purpose of technology presentation, the virtual implementation of physical system can be the better approach. Of course some commercial examples, like CCNA Network Simulator and CCNP Network Simulator, from Cisco Inc. exists and are available yet. But commercial software like this is mostly not the cheapest solution. On the other side, there are some abilities in Open Source systems like Linux. These implementations are well documented and still under development.

This Article is focused on possible chances in virtualisation of components like Label Switching Routers (LSR) and Label Edge Routers (LER) that stands as parts of network with MPLS support. At the beginning of our article some basic information about MPLS and its support of Diffserv are introduced. In the next part the interest is taken into introduction and chose of virtualization tools like Virtualbos or Wmware. First of two mine article parts is dedicated to implementation and configuration feasibilities of routers based on OS Linux. In the next part examples and results of configured topologies and scenarios on virtual network was realised and presented.

This Project was realised as a team work on department of telecommunications and the result is the virtual platform that is convenient for presentation purposes of MPLS technology abilities. Conclusion of this work is oriented eventual opportunities in usability and potential upgrading of this project.

Speech Parametrization for Operator Maps

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The paper describes application of the artificial neural network with unsupervised learning, operator maps. They were developed to implement selective responses to dynamic phenomena. Main idea is that neuron weights poses parameters of a system. In the terms of operator maps, neurons are called operators. These operators represent a kind of filters for signal features. Adaptation of the filters is performed using unsupervised training algorithm similar to the algorithm utilized in Kohonen self-organizing maps.

For utilization of operator maps, the most important is to consider carefully proper signal parametrization. In the papers is described approached where speech signal is parametrized using filter bank based on MFCC and PLP coefficients. The results obtained are compared with the approach that described signal by Matching Pursuit algorithm. Utilization of Matching Pursuit algorithm could impose advantage over traditional parametrization of speech signal. The aim of described experiment was to find suitable speech parametrization that help operator maps find features in speech signal.

Whole network than represent transformation where filters for signal features are ordered in the two-dimensional output space. The ordering preserves the relations between features in signal. According to the origin of the Operators maps algorithm, the results could be shown using software developed for KSOMs. Moreover, all the methods how to visualize KSOM are applicable (e.g. U-matrices). Originally, the method works with various parametrizations of children speech and using KSOM to distinguish between ill and healthy children [2]. It is based on the fact that some disorders, for example Developmental Dysphasia, have direct impact on speech production. The method utilizes KSOMs trained on parametrized speech of healthy children to classify speech of speaker.

Acknowledgement

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Localization in GSM Networks using Radio Fingerprinting

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This paper attempts to present current state in the area of user localization in cellular networks and shows custom solution for positioning using pocket computer and fingerprint method also known as fingerprinting. It operates in Global System for Mobile communications (GSM) network; however fingerprinting is also applicable in other wireless networks, such as Universal Mobile Telecommunications System (UMTS), Bluetooth or 802.11. Implementation is explained and it is compared to existing solutions.

Entire system uses centralized database as well as central computation server. It is designed in service-oriented architecture, which is usable be service providers, such as GSM operators. The system structure allows easy implementation of changes, central maintenance and further optimization of accuracy and performance. It is usable by multiple users, which is one of the basic requirements for a network service.

Main focus is aimed to GSM network, because it is currently most used standard in global mobile market with over 3450 millions of connection and it takes over 80% market share [1]. Motivation for localization ser-vices as it is one of the most developing markets is also discussed. In order to explain GSM network, cellular networks are briefly introduced. Particular components, such as Base Transceiver Station (BTS) and Mobile Station (MS), which are necessary for fingerprinting, are shortly clarified. Components, which are not necessary for understanding of fingerprinting in GSM network, such as Base Station Controller (BSC) or Network Subsystem (NSS) are not explained.

The paper then shortly discusses localization techniques used in GSM networks nowadays, such as Cell Identification (Cell ID), Angle of Arrival (AoA), Time of Arrival (ToA), Received Signal Strength (RSS) and more [2]. This section also explains basic properties of fingerprinting, online and offline phases, as well as its pros and cons. It is explained, what accuracy (or average localization error) is provided by current localization methods. The accuracy is later on compared to the one achieved with the solution utilizing pocket computer and fingerprint method.

Last chapters contain printed screens, which show possible visualizations of data collected with pocket computer while building up radio map. This data are basically various parameters of radio signals in GSM network. For instance coverage of specific base station or end-user localization utilizing Google MapsTM API is shown.

Conclusion shows usability of this solution, for instance the achieved results conform to Federal Communications Commission (FCC) Wireless 911 Requirements standard [3] which makes this solution applicable for emergency calls.

Acknowledgement

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New Model for the Service Providers: Capability Maturity Model Integration for Services

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Quality is one of the most important factors for service providers. Excellence of services cannot be achieved without effective service management. As in other sectors, the quality is achieved by application of standardized processes. Since February 2009 a new collection of standardized processes called Capability Maturity Model Integration for Services (CMMI-SVC) is available. This model is based on Capability Maturity Model Integration for Development (CMMI-DEV), which was modified and developed to answer service management needs. In this work we present the history, structure and evaluation practice of the CMMI model and particularly of the CMMI-SVC. CMMI-SVC is in competition with other models and standards like ISO 20000 or Information Technology Infrastructure Library. This model is interesting since it is not limited for information technologies services. CMMI is process oriented; it means that the focus is on the task performance rather than on the output of the activity. The Carnegie Mellon Software Engineering Institute (SEI) issued the first CMMI model in the beginning of 2002. The first model was oriented on the development, later it was extended by a model oriented on acquisition of products and services, service oriented part was added a few months ago. CMMI doesn't propose any certification, its evaluation method is called appraisal. Specific evaluation method called Standard CMMI Appraisal Method for Process Improvement (SCAMPI) was elaborated by the SEI and its description is available on the website of the SEI. The appraisal method specifies three classes of appraisal, two for internal use and an official one, which needs to be performed in order to get the SEI official rating. CMMI proposes two ways of maturity representation, continuous and staged representations. Staged representation evaluates a maturity of the whole organization. Continuous representation represents the maturity of each group of process defined by the CMMI model using capability levels. It gives a detailed view of the quality management in the organization and allows identification of poorly controlled processes. The goals of the CMMI model are completely processes controlled using quantitative techniques and are continuously improved. CMMI defines areas of processes with associated practices called process areas. These sets of practices have to be performed in order to achieve goals specified for that area. These goals are verified during appraisals. CMMI models are using a set of commons process areas called CMMI Model Foundation. Each model (CMMI for Development, CMMI for Acquisition and CMMI for Services) adds a set of its proper process areas. CMMI–SVC contains 24 process areas, 16 of them commons to all models and 8 CMMI-SVC specific process areas. Every process area defines followings components: required components, expected components and informative components. Required components describe objectives that are verified during the appraisal. Expected components are recommended in order to achieve required components, it's a kind of guide how to achieve required components. Informative components are examples and explications of required and expected components. This paper analyses the CMMI-SVC model, gives an overview of its structure and describes a method of appraisal. The model for services is very

recent; the SEI made some pilots with positive results. Stated benefits are improved quality of services, better view of processes, better communication between development and service staff etc. One of the main advantages of the CMMI model is the common base of the CMMI-DEV and CMMI-SVC and therefore a reduction of implementation costs. The CMMI-SVC model is the first model that covers all industry service needs. Other models are oriented to some specific kind of services, often information technologies.

Robust Heart Rate Estimation Based on Imperfect QRS Detection

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This paper presents an algorithm for the estimation of heart rate (HR) based on imperfect detection of QRS complexes. It aims to be applicable in real world HR measurement on individuals working in highly stressful and harsh environments (fireman, soldiers, etc...). Under these circumstances the electrocardiogram (ECG), from which the HR is estimated, can be often corrupted by the strong presence of various artifacts caused by electrode movement (or even skin contact loss), activity of skeleton muscles or external electromagnetic interference (primarily power noise). Additionally, the HR changes can be rapid and wide, ranging from the resting to the maximum HR. Under these conditions even the best of QRS complex detectors are not guaranteed to correctly identify all the heartbeats. Faulty detections can occur, and QRS complexes can be missed in time intervals stretching over several tens of seconds. It is therefore necessary to further process QRS complex detections, and devise a robust estimator that can reconstruct HR in a stable and reliable way.

So far only limited work has been done in the field of HR estimation from corrupted detection of QRS complexes. Most commonly, HR is estimated directly from R-R intervals, where the only correction is the exclusion of physiologically unfeasible data (e.g. [1]). A simple attempt to deal with noisy data was presented in [2], where HR was estimated based on RR interval most commonly occurring in a given time window; however, this method provides no confidence intervals, gives HR estimation only after it collects a certain number of QRS complex positions (thus introducing a time delay), and its averaging nature can cause trouble with rapidly changing HR.

In this paper we describe an algorithm that processes the data provided by QRS detector we previously presented in [3]. This algorithm is based on Bayesian framework and grid filters which utilize prior information in the form of a supergaussian probability distribution of HR changes. The algorithm was designed to follow even rapidly changing HR, whilst withstanding a series of missed QRS detections and false QRS detections. Also, the HR estimate is complete with confidence intervals to allow the identification of moments, where a precise HR estimation was not possible. Additionally, the computational complexity of this algorithm is acceptable for battery powered portable devices.

Upon testing on data obtained from subjects performing bauds of vigorous physical exercise, it was observed that the algorithm was able to cope with the aforementioned difficulties. When an interval of several missed QRS detection was encountered the HR continued to be estimated but the confidence interval started to widen. Faulty detections were rejected, and did not corrupt the HR estimate. The rapid HR changes were well followed.

Overall, the described algorithm proved to be a robust method for the estimation of HR based on intensely corrupt ECG that can be recorded on individuals performing dangerous, physically demanding and stressful tasks.

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Transparent Remote Wave Division Multiplexing Passive Optical Architecture

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The presented passive optical architecture is based on Wave Division Multiplexing (WDM) and provides a solution to problems associated with current Passive Optical Networks (PON). It has various advantages over existing PONs and AONs (Active Optical Networks) including enhanced security, increased bandwidth and greater achievable distance.

PONs and AONs are different FTTx (Fibre To The x) technologies. PONs are cheaper to deploy, they use less fibers, work on low power (cheaper) lasers and have much better economies of scale. AONs on the other hand have dedicated bandwidth per subscriber and can achieve better distances than PONs. AONs are still the most prevalent type of optical network. While both systems have their advantages there are disadvantages regarding their setups: PON splitters reduce the distance that can be achieved between active components, more users can be affected by a single point of failure and a high splitting ratio reduces the bandwidth per subscriber. AONs have a much higher deployment cost due to additional fibers. Active electronics in the field and high power lasers increase the maintenance of AONs compared with PONs. The maintenance costs of AONs are estimated to be somewhere between 20%-30% higher compared with PONs.

The combination of future data-rate projections and traffic patterns coupled with recent advances in WDM technology may result in WDM-PONs becoming the preferred solution for a future proof fiber-based access network [2]. While most of these WDMPON networks are being developed in the lab, Korea has built some real world tests using WDM-PON.

The proposed system can be used in backhaul systems. Internet Services Providers (ISP), businesses, government and scientific institutes would be the initial market for the proposed system. Incorporating the proposed passive network with AON technologies can provide greater bandwidth to the switch or router, hence delivering greater bandwidth to users.

Bandwidth requirements are growing rapidly in many areas of society. Governments are seeing the potential in using fiber optic communications to reduce costs and deliver better services. E-Government, E-Health and Tele-Education are some of the services that will benefit from a superior communications infrastructure. These services can only reliably come from fiber optic networks [3]. Among the various types of PON, WDMPON is considered to be the most future-proof technology, given that it can deliver a very large amount of dedicated bandwidth with protocol transparency [4].

This is supported by the fact that a WDM-PON network is protocol transparent. Therefore it is realistic to expect it integrated into current telecommunications systems to increase bandwidth and reduce bottlenecks.

This paper discusses the overall architecture of the proposed system, preliminary test results and an assessment/evaluation of its core features/benefits.

It also examines various modulation techniques and outlines the advantages of using Non Return-to-Zero (NRZ) over Return-to Zero (RZ) conversion (with NRZ modulation being extensively used in many data communications systems because of its relative ease of generation)[1].

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Network Address Translation Traversal Issue in VoIP Network

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Network Address Translation (NAT) is a technology commonly used by routers to allow multiple LAN network devices with private IP addresses to share one or multiple public IP addresses on the router WAN interface. In order to allow communication of LAN network devices to the Internet, router does translation between private a public IP addresses. This technique is commonly referred as Network Address Translation (NAT) or IP Masquerading. The passing of traffic through router doing NAT is called NAT Traversal.

The NAT principle is simple. When a network device on the LAN initiates connection to the Internet, this device sends all packets to router first. Then router replaces the source IP address (the private address) with its own public address before passing the traffic to its destination. When a response is received, the router searches its translation table to find the original source of communication in LAN, from which the communication originally started and then passes the response to that device.

The way in which VoIP is designed is creating a problem to some kind of traffic passing through a router doing NAT. Conventional VoIP protocols deal with call signalization of telephone connection. The audio traffic is handled by another protocol(s), and unfortunately, the port on which all the audio traffic is sent is random. The NAT router is able to handle call signalization, but it hasn't knowledge about audio traffic and it's relation with call signalization. As a result, audio traffic is not translated and router has no way of knowing that it should be passed to the same device the signaling traffic was passed to.

Voice over IP (VoIP) is being rapidly embraced as an alternative to the traditional public-switched telephone network (PSTN). The issue of NAT Traversal is a major problem for the widespread VoIP deployment. It is significant problem with non trivial solution. However, the easiest way is to avoid using NAT. Unfortunately, it's not possible in IPv4 environment today.

This article provides description of the NAT Traversal issue in VoIP network and put emphasis on explanation of call signalization and audio traffic flow between devices with public and private IP addresses.

Detection of Spectral Properties of the Special Fibers

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In present time, the claims for bandwidth due to increased volume of provided multimedia services based on TCP/IP protocol is still growing. It appears that these needs can be successfully satisfy by optical infrastructure. Moreover, the service providers evolve major effort to tighten the optical fiber to the customer's home. In a modern optical infrastructure not only the conventional fibers for data transmission application but the special optical fibers for wide application like the optic-fiber sensors are most popular. In these applications just the special fibers are better than conventional ones. In this paper, the family of special fibers, namely, the PCF (Photonic Crystal Fiber), TCF (Twin Core Fiber) and DC-TCF (Dual Core TCF) will by introduced.

In conventional fibers used in today's telecommunications infrastructure, the light propagation is caused by different refraction index between core – cladding. On the other hand, in PCFs no reflections on the two environments interface, but the interface of regularly arranged defects surrounding the fiber's core cause the light propagation. These defects are created by holes, which are imposed along the x-axis of optical fiber. There are two possible mechanisms of the light guiding inside the microstructure (photonic) fibers, namely, the index guiding PCF and bandgap guiding PCF. In the index guiding fibers, the light inside the fiber is guided by modified total internal reflection. On the other side, in the bandgap guiding fiber the light is guiding by fiber core with lower refractive index than the cladding.

In this paper, the detections of the spectral properties of the photonic twin-core fibers for optic sensor applica-tions are described. In experiments, the TCF fiber with total length of 8.5 cm was used. This fiber is characterized by one core, which is located in the fiber's axis. The experiments were realized in wide range of wave-lengths 1450 - 1700 nm, where the monochromator was set up at 0.04 mm. The experiment shows that the achieved signal power never reaches the maximum value of power and even dropped to zero. This can be associated to non-homogeneous material forming the fiber, eventually to a higher cores distance for better power spilling between them. On the other hand, it can be explained as the influence of coupling length dependent on the wavelength. Anyway, if the signal is growing, the signal want to achieved the maximum power to another core and then periodic change wavelength caused a change coupling length.

In second experiment, the DC-PCF fiber at length 28.51 mm with 90 µm in diameter was used. The measured optic fiber consists of two cores with 2 mm in diameter and with average holes separation at 3 mm (centre distance hole to hole) produced from the same material. The experiment realized on the special optical fibers detects that spilled optical power between individual cores depends on coupler length and it is for each wavelength different. Realized experiments approve that the behaviours of conventional dual core optic fibers and dual core microstructure fibers are very similar and also that microstructure dual core fibers have a higher frequency oscillations than the dual-core fiber. That is caused by differences in phase constants of symmetric and asymmetric modes.

Building Volumetric Distance Maps: Analysis of Methods

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The distance map is data representation where at each point of object we know distance to the nearest point of some area. The operation for distance map construction is named distance transform. In the image literature distance transforms (DT) have appeared in 1966 in work Rozenfeld and Pfaltz [1] where they have applied a chamfer distance transform to construction of a skeleton of object. But algorithms of distance map transform are improved every time. Also alternative decisions have been invented. DT have been studied for various representations of data: hexagonal grids [2], point lattices [3], elongated voxel grids [4]. DT are applied in many areas of image processing (morphology [5], [6], skeletonization and medial axes [7], [8], [9]); object decomposition [10], which greatly diminishes the complexity of a recognition task; visualization (ray-tracing, endoscopy), modeling (collision detection, morphing, animation, constructive solid geometry). Last time many methods of 3D-objects construction was created. In this connection, there was a necessity for fast and exact algorithms for construction of 3D distance fields. In this paper the most popular approaches are considered and compared with each other, as a result it is possible to answer on a question, in what case using different method of distance maps building is better.

The methods were coded in C++ by using open source library Insight ToolKit (ITK, www.itk.org). The code was released under the GPL free software license. Algorithms were compared on several tests in the sizes. The first test – a point in the area centre. The second – the image of a bone with a cavity. The third – "tree" with set of segments of a small thickness.

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Network Simulator for Conversational Quality Measurements

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End-to-end telecommunication network simulator design and its realization is described in the paper. The main applications of the simulator are conversational tests e.g. as per ITU-T P.800 and demonstrations. The simulator can be also used as a replacement of telecommunication chain part that is missing during end-to-end testing or measurements.

To run subjective conversational tests, the following is needed (except of test subjects):

- Defined test environment (two anechoic or semi-anechoic rooms with known reverberation times and background noise) as stated e.g. in ITU-T P.800

- Tested communication network or its real-time model (network simulator). The network simulator must replace tested network as accurately as possible in terms of subjectively perceptible impairments (signal level and its variations, signal drop-outs caused e.g. by packet loss, signal distortions caused e.g. by used voice coding or even multiple coding techniques, signal reflections (echo), etc.)

The deployment of network simulator is much more convenient than real network usage due to the need of laboratory test environment (anechoic rooms are typically not available at the telecom exchange premises).

The designed network simulator consists of the following components: User Terminals, DSP blocks for signal coding, frequency filtering, delay and echo simulation and IP simulator (NistNet) for jitter and packet loss simulation.

In the full text, examples of achieved subjective test results are presented, including their scoring obtained by PESQ.LQ (ITU-T P.862 and P.862.1) and 3SQM (ITU-T P.563).

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Interactive Application for Electric Circuit Analysis

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The paper presents interactive application for analysis of electric and electronic circuit through internet. The application is based on PHP scripts and use Spice and Maple with PraCAn [4] package for circuit simulation. Numeric results of analysis are produced when SpiceOPUS [3] is used, while symbolic or semisymbolic analysis is invoked in case of Maple and PraCAn package. Continuous-time linear and nonlinear circuits as well as periodically switched linear (PSL) circuits can be analyzed. Description of the circuit can be entered through graphical schematic editor as a Java applet. The whole system is developed at the Department of Circuit Theory, for research and teaching support.

The application is based on client-server conception [1] analogous to [2], where programs are installed and run on the server side. However the described system offers better user interface and greater scope of simulations including symbolic analysis and analysis of periodically switched linear (PSL) circuits, i.e., circuit with switched capacitors (SC) or switched currents (SI).

It runs on using batch-processing which is necessary for utilization in the interface where the programs are called by the PHP scripts [1]. The programs solve the tasks of simulations, create graphs and formulas with graphic representation, ... According to client requests the results are presented by the dynamically created www pages. These pages created also by PHP scripts are provided to the client by means of HTTP server Apache.

Circuit is represented on the basis of text description using so called netlist. The netlist can be created directly by user or by schematic editor where the user creates graphic representation of circuit. The schematic editor is designed as a Java applet. The application supports plugins, which are loaded dynamically on the application start. Plugins can receive notifications in case of scheme modification, so they can update properly.

The application was created especially to teaching support on Faculty of Electrical Engineering, Czech Technical University (CTU) in Prague. It should help the students to make analysis of electric and electronic circuits easy without any program installation and without learning of any command syntax.

Operating of the interface is very easy. Circuit description can be entered using graphical interface of schematic editor. All pages of application are supplemented by interactive help. User can use the interface without any manual or study of syntax. The analysis can be very easy created and modified.

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Automatic Speaker Identification using GMM with Limited Amount of Training Data

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The paper presents Automatic Speaker's Identification (ASRI) system based on Gaussian Mixture Model (GMM), which is used as a statistical representation of speaker's patterns. The paper deals with an influence of statistical model parameters and feature vector dimension to overall accuracy of the system. ASRI allows to recognize „who is talking„ from speech signal. This technique assigns verification of speaker identity based on biological individuality similarly like the fingerprints or palm prints. One of the advantage of ASRI is that it is not dependent on ID cards, keys, passwords, etc., which might be easily lost or forgotten. Among various application, we can mention convenient voice-based access to secure areas or implementation to systems of automatic retrieval of audio documents. The recognition process can by divided into three basic steps: 1) Feature extraction where original speech signal is replaced by much more effective compressed parametric representation; 2) Estimation of GMM based statistical model for each speakers in the database; 3) Decision if the input voice belongs to the known speaker, whose voice model is in the set of known speakers.

The features used in the experiment are Mel Frequency Cepstral Coefficients (MFCC) together with their first and second time derivates, which serve as descriptors of speech spectral dynamics. The distribution of feature vectors extracted from speech is modelled by a mixture of Gaussian density functions. Complete GMM is defined by mean vector, covariance matrix and mixture weights. Here, diagonal covariance matrix is used, which usually gives better results in recognition compared to full covariance matrix. The best results in parameter estimation are achieved by using the iterative Expectation Maximization (EM) algorithm..

One of the practical problems in many ASRI applications is a lack of appropriate amount of speech training data. Usually the speech of a person lasts only a few seconds which may not be sufficient to estimate speaker's model. The developed ASRI system was evaluated on such constrained training data to better understand its limit in the sense of a minimal sufficient length of training speech sequence.

The CSRI system was evaluated on the subset of the TIMIT speech database that consisted of 630 speakers where only 12 seconds of speech for each speaker was available. Half of the speech signal of each speaker was used for training and the rest for testing. The input utterance was divided into frames of 100 ms duration and windowed by Hamming window. From all of these frames, MFCC coefficients of dimension 12 and 20 complemented by its time derivates were extracted. Also various number of Gaussians in GMM were evaluated. During recognition, a decision was performed on 1 second speech segments.

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On Particle Swarm Optimization and its Comparison with GA

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In this article, a relatively new approach to the optimization problems called Particle Swarm Optimization (PSO) designed by Eberhart and Kennedy in 1995, is presented and compared with more common Genetic Algorithm (GA) approach. PSO is inspired by social behaviour of animal communities, particularly birds. For instance, while searching for food the most of the individuals tend to follow the one that has found the "best" food – this principle is integrated into PSO. It has been recently used in the area of neural networks training, image classification, digital filter design or speaker segmentation.

PSO works as follows. Every individual from the community represents one single solution in the search area. Each of these individuals has its location in the search area, velocity directing its "flight" and remembers its own best location. The system is initialized by the population of random solutions (called particles) and tries to find optimal solution creating new generations (populations) with better solutions. The fitness is counted for each randomly located particle. The particle with the best fitness stores its location into global "memory" (gBest). Except this memory shared by all particles, a personal memory of each particle also exists, in which the best individual solution is stored (pBest). Every particle in the population compares the present fitness value with its personal best and global best values, and if appropriate, replaces the values in the memory (personal or global) by the best one. When every particle knows gBest and pBest, new velocity and position is counted. Particles move to the new locations with new velocities. New fitness of each particle is counted again and the whole cycle is repeated. PSO uses only primitive mathematical operations therefore it is computational inexpensive. It achieves a fast convergence however it may tend to the premature convergence on a complex problem with many optimums.

GAs are other very often and successfully used evolutional optimization methods. Its origin reaches to the middle of seventies when James Holland applied Darwin's evolutional principles working in the nature These very well known techniques use evolutionary operators - selection, crossover and mutation - that affect the convergence speed of GA. But GAs often suffer from slow convergence due to requirements of a large number of function evaluation.

We have performed comparison of PSO and GA in the meaning of speed and accuracy of their optimal solution. Four test functions were chosen for PSO and GA to find its global minimum – Ackley's, Rastrigin's, Griewangk's and Schwefel's functions. All these functions have many local minimums and only one global minimum. The search algorithms are potentially prone to convergence in a wrong direction. Very remarkable function is the Schwefel's one. It is deceptive in the way that the global minimum is geometrically distant from the next best local minima in the parameter space. The test, consisted of 20 runs for both algorithms, was performed in the Matlab program. The results show that PSO is noticeably faster and reaches higher speed of convergence. For all the test functions PSO has found a better solution than GA, though the mean value of Ackley's and Griewangk's functions was worse for PSO. PSO also reached, in spite of Schwefel's function problem expectation, markedly better solution in comparison with GA.

Autoregressive-Based Long Term Prediction of Rayleigh Fading Channel

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One of the fundamental limitations of mobile radio communication is time-varying fading nature of radio channel. In adaptive transmission systems, the channel state information (CSI) needs to be predicted ahead for rapidly time variant fading channels [1]. This paper addresses the utilization of fading amplitude prediction and thus more efficient use of CSI to improve reliable communication. To enable adaptive transmission, a fading prediction algorithm must predict the fading signal at least for the upcoming transmission frame. Such predicted information about channel state in near future is fed back (uplink) to the transmitter (base station). Prediction ranges in adaptive transmission applications are typically much larger than in channel estimation, signal detection, decoding and other receiver-based methods. The desired prediction range can vary from a millisecond to many milliseconds, depending on the application and transmission system specification (CDMA, OFDMA, MIMO systems, etc.)

Linear prediction algorithms are one of the most common and favourite because of their good tractability, very well developed theory and simple implementation [1]. Works, recently published, have shown that the prediction accuracy of time sequences of the fading channel amplitudes is comparable with or even outperforms other more complex prediction methods. Various aspects of linear prediction of fading channel gain by autoregressive (AR) modelling [2] are studied in the paper.

Tests are performed on Rayeigh flat fading channel simulated by generally well accepted Clarke-Jakes model. The prediction accuracy is evaluated for various values of AR model order, prediction horizon, and adaptation rates of AR parameters together with different fading scenarios, such as change of fading rates, number of paths and scatters for signal propagation and/or change of sampling periods for observations of CSI.

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3D model Reconstruction of the 2D Images by Voxel Coloring Algorithm

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This paper presents a efficient method for volumetric scene reconstruction from multiple views. Voxel coloring can reconstruct a 3D scene from multiple photographs with known camera location (intrinsic and extrinsic parameters). The algorithm identifies a special set of invariant voxels which together form a spatial and photometric reconstruction of the scene, fully consistent with the input images. The principle for understanding this algorithm is epipolar geometry, which expresses attitude between two images. It is used to calculate 3D coordinates and calibrate matrix.

Epipolar geometry is used for scene geometry reconstruction and is used reconstruction coordinates of images on principle two or more images too. Epipolar geometry expresses attitude between two images. Epipolar geometry is derived only from intrinsic camera parameters and the mutual positions of camera. Fundamental matrix F describe attitude between three-dimensional coordinates of object and two-dimensional coordinates. Fundamental matrix is a 3x3 matrix. In this way, fundamental matrix can restrict the searching area of a matching point in a line in each image, because the correct matching point must exist in the corresponding epipolar line in each image. Camera calibration is the process after during is detected intrinsic and extrinsic camera parameters. Intrinsic parameters are used to expresses as is the point projection into the eventual image. We know the four intrinsic camera parameters: focal length, principal point, skew coefficient, distortions. Extrinsic camera parameters : are two rotations, translations.

Voxel coloring is an algorithm to reconstruct the 3D shape from a number of input photographs with known camera locations. The reconstruction consists of a regular grid of cube-shaped voxels. Voxels are 'volume elements' in analogy with pixels. Like a pixel, every voxel is associated with a color [2]. Voxel coloring is the assignment of colors to points in a 3D volume as to be consistent with the input photographs. We used calibration Toolbox for Matlab to detected intrinsic and extrinsic camera parameters. And we used these parameters to correct distortion of images. We had images object on chessboard. We used thirty views spanning 360° object rotation. Entire calibration is time-consuming because we have to each of all images manually to calibrate [1]. When we have done calibration with all of images and we have calibration matrix K, than we start main 3-D reconstruction. We used Voxel coloring framework. Finally reconstruction is time-consuming and it depends on achievement of PC. The voxel coloring algorithm is relatively involved on time a comprehensive because it uses complicated mathematical form but is reliable and has no problem with absorbing colors.

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Algorithms for Reconstruction of 3-D Image and 3DTV Technologies

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Processing, reconstruction and the subsequent creation of 3-D image plays a very important role especially in the development of modern three-dimensional television and still newer imaging technologies, displays. The display is the last component in a chain of activity from image acquisition, compression, coding transmission and reproduction of 3-D images. 3-DTV is one of areas which recently achieved a huge expansion. However, certainly 3DTV is still waiting for an intense progress in nearest future.

We live in a three-dimensional dynamic world and possess stereo vision thanks to the pair of our eyes. The fundamental problem of computer vision is to obtain 3D geometric information of the scene from captured planar images. This process is traditionally termed 3D reconstruction.

The display is the last, but not the least, significant aspect in the development of 3DTV and vision. As has been outlined in other papers in this issue, there is a long chain of activity from image acquisition, compression, coding transmission and reproduction of 3-D images.

There are more attitudes in development of 3DTV, they deal with reconstruction of 3d picture, namely stereoscopy, auto-stereoscopy and holography. For reconstruction of 3D scene is necessary to scan it with several video cameras, because reconstruction from more video records leads to simpler and more accurately techniques. Stereoscopic method belongs to the oldest and simplest methods. A Stereoscopy method is based on natural perceiving of the world with two eyes whereby each eye perceive slightly shifted picture of scene, what creates the illusion of depth of field. Unfortunately for watching of this kind of picture is necessary to use special equipment (glasses), which is uncomfortable. This small disadvantage is eliminated by auto-stereoscopy. Holography is based on principles of real light duplication, which is giving us complete information about 3D space.

3DTV display technologies:

- Stereoscopic
- Autostereoscopic
- Holographic
- Volumetric

Holography is a newer technology compared to stereoscopy, and there are indicators that satisfactory holographic 3DTV. The term “autostereoscopic”, strictly speaking, describes all those displays which create a stereoscopic image without any special glasses [1].

EXPERIMENTS WITH SELECTED ALGORITMS

The reconstructed 3-D image is closely related to the following experiments - calibration, detection outstanding corners in the image and find the corresponding pairs, calculate the fundamental matrix, rectification. These experiments are carried out in MATLAB.

CALIBRATION

Recovering 3D structure from images becomes a simpler problem when the images are taken with calibrated cameras. For our purposes, a camera is calibrated if the mapping between image coordinates and directions relative to the camera center are known. However, the position of the camera in space (i.e. its translation and rotation with respect to world coordinates) is not necessarily known.

Calibration is a matter of translating what the camera sees in sensor co-ordinates (pixels) to world coordinates in e.g. millimetres For achieving of exact data is necessary to calibrate both cameras. With calibration are defined inside camera parameters (calibration matrix, coefficients of distortion) and outside camera parameters (relative position and orientation of cameras).

Extrinsic Parameters

- R, 3x3 rotation matrix
- T, 3-D translation vector

Intrinsic Parameters

- the x-coordinate of the the center of projection, in pixels
- the y-coordinate of the the center of projection, in pixels
- effective focal length in pixel
- centre of projection

EPIPOLAR GEOMETRY AND FUNDAMENTAL MATRIX

Epipolar geometry refers to the geometry of stereo vision. Typical use case for epipolar geometry - two cameras take a picture of the same scene from different points of view. The epipolar geometry then describes the relation between the two resulting views [2].

The epipole is the point of intersection of the baseline, with the image plane.

The epipolar plane is the plane defined by a 3D point M and the optical centres C and C'.

The epipolar line is the straight line of intersection of the epipolar plane with the image plane.

RECTIFICATION

Image rectification is a transformation process used to project multiple images onto a common image surface. It is used to correct a distorted image into a standard coordinate system. The process of rectification makes it very simple to impose this constraint by making all matching epipolar lines coincident and parallel with an image axis. Many stereo algorithms assume this simplified form because subsequent processing becomes much easier if differences between matched points will be in one direction only [3], [4].

I mentioned some manners allowing a 3dimensional reconstruction of picture or object in this article. These methods of 3D picture processing finding great application in sphere of medicine, for example detection and identification of tumor in brain. However, it is progressively finding application also in other branches as physics, astronomy, biology or geography.

A Matlab environment has been used for final realization of experiment.

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Enhancement Ethernet Network Simulator

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The Next Generation Networks are built mainly on the Internet Protocol (IP) stack and the Ethernet. The simulator of the Ethernet and the IP network can be used for testing of new network elements and services. The Ethernet Network Simulator "EtherShaper" was developed on the Telecommunication Technology Department of the Faculty of Electrical Engineering of the Czech Technical University in Prague. The feature and capability of the EtherShaper include following:

- Two RJ-45 ports for Ethernet testing (port A and port B).
- Supported Ethernet versions: 10Base-T, 100Base-Tx
- Duplex mode: Half, Full, auto-negotiation
- Speed: 10 Mbit/s, 100 Mbit/s, auto-negotiation
- Minimal Ethernet frame size: 50 bytes
- Maximal Ethernet frame size: 1500 bytes
- Independent adjustable true linear delay in both directions in range from 0.5 ms to 30 s
- Independent adjustable "RAMP" delay in both directions. RAMP is defined with start delay, end delay and incremented in s per ms.
- Independent adjustable loss in both directions in range from 0 % to 100 %
- Load/Save user configuration

The Ethernet Network Simulator is based on a PC platform with double LAN card and OS Linux. The user interface of the EtherShaper is designed to be intuitive and easy to use. Using the monitor and keyboard, you can set up the Ethernet frames processing. The navigation through the program menu is carry out by the cursor keys (Left, Right, Up, Down or Tab key) while the selection must be confirmed by Enter key.

The EtherShaper can be connected between the Ethernet Generator and Analyzer anywhere in a broadcast domain. In this case generated Ethernet traffic is modified in the EtherShaper and sent to the Analyzer. The ethernet traffic can be modified only in one direction. Both - the EtherShaper Unit and the measuring equipment must be placed in the same broadcast domain.

The Ethernet Network Simulator EtherShaper Unit can be connected between the Ethernet Generator/Analyzer and the loopback units placed anywhere in a broadcast domain. In this case, the generated Ethernet traffic is modified in the EtherShaper and sent afterwards to the loopback device. The Loopback device loops the traffic back to the analyzer through the EtherShaper. The Ethernet traffic can be modified on both directions. The EtherShaper and measuring equipment must be placed in the same broadcast domain.

The Ethernet Network Simulator EtherShaper Unit can be utilized in networks that are compatible with the Ethernet standard. Placing the EtherShaper Unit into the same broadcast domain (as the devices whose traffic is shaped) is necessary condition for a good function.

The paper will be represented the integration of the EtherShaper Unit within a common wired and wireless networks based on the Ethernet. Adding software modules for other types of variable delay will be developing during this year. The extension for simulation network with E1 interfaces and TDM over IP transmission will develop in next phase. The standard serial produced converters which change E1 to Ethernet and can be used for connecting to the EtherShaper Unit. The program hast to calculate the adding process delay for TDM to the packet conversion.

Acknowledgment

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Speech Processing Algorithms on Hardware Platform Texas Instruments OMAP-L137

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There are types of real-time speech recognition tasks where it is not possible to use "normal" PC platform with the x86 processor (for example there are requirements for low power, small size, portability of equipment or cost minimizing). Therefore it is necessary to find another hardware platform that would meet these requirements, despite the more limited system resources, such as lower processor clock speed, less RAM, etc. The relatively lower performance of the platform should be replaced by more efficient performance of algorithms, but often at the cost of reducing accuracy of calculations.

As one of the suitable platforms for speech recognition, we have selected the dual-core processor from Texas Instruments OMAP-L137 and we performed basic experiments to test some algorithms, which are part of the speech recognizer. OMAP-L137 is a hybrid processor, which is composed of one core of digital signal processor "TMS320C647x" and one core of general purpose processor "ARM926EJ-S". Both cores are currently well-known industry standard, thus a large amount of documentation and software is available.

TMS320C647x core, as is known, computes with the fixed and floating-point format of numbers and its VLIW (Very Long Instruction Word) architecture would allow to carry out operations in eight parallel units, but for example only two Multiply-and-Accumulate instructions simultaneously. ARM926EJ-S core also includes an extended instruction set for digital signal processing.

A part of the speech recognition process is to calculate a probability b() function. This is a very computationally intensive operation. This function evaluates the similarity of a parametrised segment of the signal with before trained data of acoustic model of trifons. The core of the function consists of several elementary operations (difference, multiplication, square, accumulation), followed by other operations such as logarithm of sums of exponential functions approximation. The results of our experiments show that the greatest contribution to accelerating of calculations has the optimization of the innermost core of the b() function.

We have performed a number of optimizations, which resulted in a multiple increase of the speed of calculations compared with the original function taken from the source code for the PC platform. Good results can be achieved by just turning on automatic optimization of the C language compiler (the speed increased by about 67%). On the TMS320C6000 platform can be achieved an optimal balance between speed and precision of multiplication, if the input data is expressed in 16-bit format and the result is in 32-bit format in fixed point. Further acceleration of calculations can be achieved by appropriate locating of frequently used data into a faster access memory (memory of "near" type).

By rearranging the source code (written in C programming language) can be achieved a better continuity of instructions without waiting for reading data from memory (this is so-called the "Code scheduling" method). Significant further acceleration can be achieved using so-called "intrinsic functions" that are almost direct access to special instructions of the specific processor architecture in C language programming environment, which would be otherwise probably unused by the compiler. The possibilities of the architecture can be fully used with these functions, such as vectorization (calculating with multiple operands in one instruction, for example $c1 = a1 * b1$, $c2 = a2 * b2$) and macro-operations (calculation of multiple operations in one instruction, such as $c = a1 * b1 + a2 * b2$).

TMS320C647x architecture also provides the floating-point arithmetic, although the possibility of optimization is limited compared with calculations in fixed-point format of numbers.

The combination of digital signal processor (DSP) and general purpose processor (GPP) is proving to be very suitable for embedded systems for speech recognition. GPP, which operates on an operating system (for example Linux) here makes overall system function, interface for communication with the user, input-output processes, network connections, etc., while the DSP is repeatedly called to calculate the time-consuming operations.

Principle of Electronic Study Materials Creation

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With the development of new information and communication technology, there is the creation of study materials in electronic form. Their importance is often overrated. Some authors consider that the electronic study materials are usable virtually everywhere and always, consider them almost as self-redemptive. There are study materials of different qualities, which may be of practical use questionable.

In creating the educational materials should be always well-considered goals that we want to achieve and by teaching it to develop an appropriate study material for the target user group and the form of teaching. Creation of an overhaul of electronic learning materials is not easy, and it is very time-consuming and financially demanding. It is always necessary to choose the reasonable option..

Under the term electronic learning we understand, that student gets the main part of necessary knowledge and skills on topic, which they study by e-learning. The creation of electronic studying materials is not only technical question but also social question. The creation requires new approach of education and learning. It's impossible to manage this difficult task only as second task of education, because we don't have any material, which can clearly explain methodology like this.

The basis of e-learning courses is to prepare an electronic material. LMS (Learning Management Systems) offers information, content, communication and testing tools. Total material carries general information about the course (e.g., title, teachers, objective ...) and then is divided into logical parts - chapters, which can be studied separately; each chapter has its own single structure (e.g., interpretation, its application, discussion, tests ...). Electronic material should be similarly structured as classically written book: title, introduction: input assumptions determination (year, specialization), the objective content - generated automatically, schedule and support for the main components - modules, dates of tests, transmission projects, requirements, contact the teacher, needed software support, a glossary, a list of abbreviations and signs, a summary of the module guide - link chapters, feedback - satisfaction surveys before and after the course, self-testing - "blind trial" before stabbing the verification of knowledge. Some of these parts take itself LMS system; educators should focus in particular on the content page.

Feedback is part of e-learning, and each electronic textbook plays an important role. The student needs to process the subject matter, to verify to what extent he really understood and determine whether it satisfies specified objectives. Information on the results of the study is an important component of quality electronic study material and expressive factor motivating the students for their further studies. The role of tutors is to ensure the activity of students using different types of questions, exercises, tasks and tests. Each activity must be clearly explained to the student understand what to do; otherwise it may happen that a student misses activity.

Part of each test is the test results. Individual tasks didactic test are not grade, but assign numbers. Scoring of each task provides students with information about the outcome. Total of all points gained in the development of a student, called the score. Sometimes we meet with the notion of individual scores, as it applies to a particular student. Some tutors to create assessment tests used express the result as a percentage It is appropriate to add for total score verbal rating scale in one sentence, that briefly describes the level of student knowledge. Control all outputs, results of operations is based on student comparison.

They compare the goals of education to what students actually achieve. Tutor studying the test results as a testimony, examine their accuracy and precision. It follows the process solutions, compares the results of peer students in different roles, as well as throughout the test. Analysis tasks, tutor found the most common mistakes students who did not solve the task. It is an overview of the errors committed by the students in the testing of knowledge is valuable information for a mentor for the improvement of test tasks and plan for changes in teaching.

The Matrix Permutation Problem in Fuzzy Arithmetic

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Article deals with the problem of permuting the elements within columns of the fuzzy matrix. The goal is to minimize a fuzzy-valued function (irregularity measure) of the row sums. Article is motivated by practical need of the regularity in the job scheduling where inputs are vague or unprecisely given. We are primary concerned with the two columns case application. Solution of the problem is based on a systematic extension of the traditional formulation of the LP problem, in this case linear assignment problem.

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Simulation Model for Localization in Ad Hoc Networks

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Ad hoc networks are one of the most perspective areas of mobile wireless networks. The idea of Ad hoc wireless networks has been under development since 70s and 80s of 19th century [1]. In the half of 90s new standards for wireless communication has come and research community found in ad hoc networks new possibilities of radio networks evolution.

In many ad hoc networks applications is often important to know position nodes, that can be achieved by implementation of GPS receivers into nodes, but this is not effective from economic point of view, because it is expensive to have GPS receiver in each node. In case that we use GPS receivers in every node is problem with battery consumption, on which depends whole operating life of network. These were reasons for developing numbers of localizations algorithms that can estimate position of nodes in network.

Most of localization algorithms use data from two main sources: information about Angle of Arrival (AoA) or information about distance. Information about distance can be achieved in two ways. First way is to measure Received Signal Strength (RSS), the second is measure Time of Arrival (ToA), respectively time difference of arrival (TDoA). When RSS is used, we need to know signal strength of transmitter, if we use time measurement is important accurate synchronization between transmitter and receiver [2].

Many research teams still works on new, more accurate localization algorithms, or on improvement of existing algorithms. We created complex simulation model for comparing of localization algorithms and their improvements. It will be described in this paper.

This is the second generation of simulation model, its vision and basic parameters can be found in [4]. Our simulation model for localization in ad hoc wireless networks was created in programming environment of Matlab. In designing we put emphasis on robustness and extremely easy expandability of new localization algorithms and their parts. It is open system which can be enlargement of new localization and optimization algorithms in future.

Simulation model allows comparing performance of localization algorithms and evaluating the impact of individual parameters of the network. Simulation model can be divided into three main parts – input part, localization part and part of processing of results [3].

In localization part is placed section for mathematical methods of position calculation and base of localization algorithms that contains all implemented localization algorithms. Actually there is nine algorithms based on AoA, RSS, ToA and on hop count too. Radio channel can be simulated as simple fading channel with additive white Gaussian noise (AWGN), channel with Rayleigh fading or channel with Rician fading.

In part of results processing, function for achieving data from simulation results is placed. The performance of the localization algorithms can be easily evaluated by this data [3].

Proposed simulation model for localization in mobile ad hoc networks allows us to compare performance of different localization algorithms, based on localization accuracy. Biggest advantage of this simulation model is fact that it is open system that can be anytime renewed with new functions, whether localization, optimization algorithms, or mathematical method of position calculating.

Acknowledgments

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Simulation of Segment-based Robust Fast Optical Reservation Protocol

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Every year we could see increase demand on transfer speed optical networks. We also need create robust and sophisticated nods. On the present we give accent to transfer speed and quality, but not to effects and simplicity. In robust wide area network with very fast transfers we must create enough efficient reservation signalling protocols that are able make prevention before loss big amount of dates.

On the present are various reservation signalling protocols, such as Resource Reservation Protocol - Traffic Engineering (RSVP-TE), Intermediate-node Initiated Reservation (IIR), Robust Fast Optical Reservation Protocol (RFORP) and Segment-based Robust Fast Optical Reservation Protocol (S-RFORP). In paper we show only from best one S-RFORP, which taking advantage of the parallel segment-based discovery/reservation.

Each reservation signalling protocol must be able solve various problems such as minimize discovery, reservation and blocking recovery delays, and avoids costly path-based recovery processing.

As we wrote before, S-RFORP has only two phases. The first is path discovery and the second is reservation.

Discovery phase starts in source node of the first segment. Source node sent parallel discovery initialization message to the all head nodes in all segments. After received this message each segment is in discovery phase, i.e. each node in segment must insert information about his free wavelengths (e.g. λ_1, ..., λ_4) into the discovery message. When discovery message has arrived into the tail node in the segment, tail node must evaluate information about wavelengths and determine optimal wavelength in a given segment.

When discovery phase is finish, next phase is reservation. In this phase could be two cases. The first, reservation in the segment when pass correctly, i.e. tail node in the segment sent serial reserve message with choice wavelength (e.g. λ_1) to each node until arrive to the head node in segment. Head node sent a reservation results to the source and destination node.

In the second case, if reservation will fail in one node of segment, because this node hasn't free resources (wavelength choice). After detection, node without free resources sent a fail message to the tail node and tail nod must activate blocking recovery phase. In this case, tail node use a backup wavelength (λ_4) and sent new reserve message along with release message, because nodes, which had reserved first wavelength (λ_1) must release it and reserve a new wavelength (λ_4). When new reserve message arrive to the head node, head node sent a reservation results to the source and destination node. If is backup wavelength in segment not free, head node sent fail reservation results to the source and destination node.

Draft for Methodology for Comparing the State of ICT Infrastructure in Developed Worldwide Countries Based on Knowledge of Trend

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Rapid development of information and communication technologies supported by research in optics has brought sufficient bandwidth for new broadband services and applications. New opportunities became evident not only in technological area of information and communication infrastructure, but also in expected increase in number of broadband access. Broadband access and services in nowadays, are seen as crucial for business development and overall country prosperity. Thanks to that, the Internet is not only a service, but it is a platform, which has a significant impact on all areas of society.

Implementation of research results and deployment of broadband accesses in OECD countries is different despite expectations. The major factors influencing broadband services usage are following: broadband penetration, utilization of broadband services in the households, the geographic relief of the country, offers of services/access speeds and cost of access/services. In generally, we speak about the groups of technological, financial and social criteria.

Analyzing the implementation steps of broadband access in the countries with the highest penetration of broadband accesses gives an opportunity to the other countries to utilize appropriate know-how for their own benefit and growth with respect to their own specific market environment. This should be also supported by this new methodology of ICT infrastructure evaluation, which will be based on comparative analysis of chose key criteria.

An important part of presented methodology is the choice of evaluation criteria. According to abovementioned groups of successful growth of broadband accesses, I rate the group of selected countries based on 8 criteria. These criteria include: increase in broadband penetration per year, the number of broadband users per 100 inhabitants, the number of Internet customers per 100 inhabitants, the growth of internet penetration, price for 1 kbit/s calculated in U.S. dollars converted to purchasing power parity (PPP), investments in telecommunications technologies, the average gross domestic product per working hour, economic unit e-readiness. To each criterion was set its weight according to its importance. Criterion weight = Weighting its importance during the process is determined by a factor of 1, 1.5 or 2. The most important technological criterion is the number of broadband users per 100 inhabitants and it has a weighting factor 2. However, 'the value of gross domestic product per working hour' criterion, which depends on several factors has, the weighting coefficient 1. Other criteria with the same importance for the growth in broadband accesses are rated with coefficient of 1.5. Sum of all criteria multiplied by the rating factor gives us the overall country score. The country with the score higher than an average and concurrently with high growth in broadband and Internet penetration is a country with good processes for the growth of broadband access. Calculation of best practice (BP) boundaries identifier is defined as the sum of average score for all countries and weighting coefficients. The country scores higher or equal than BP is the country with the best practices for implementing broadband into life.

Conclusion of the methodology is dedicated to the description of activities and processes, which significantly support the growth of in broadband accesses. These activities include: improving user skills in ICT, increase internet penetration rates, tax relief, coverage of broadband access in remote areas, development of broadband services, changing regulating/regulation frameworks, digital content development, synergies promote between the private and public sector, improvement of broadband access / services and funding of research projects focused on ICT.

Large Measurement Data Set Automated Processing

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This article will discuss the methods and tools used to process and analyze the data acquired during automated laboratory measurement. It emphasizes the advantages of the usage of SQL database computer language for managing the measurement data set with large amount of members and, secondly, it highlights the use of Gnuplot command-line program that generates overview plots of data selected by user for comprehensive results presentation.

The measured object used in this article as the data source example is the function of the radio- frequency receiver data demodulator. The measurement laboratory bench provides theautomated method of data acquisition as well as automated external parameters settings, such as received field level, modulation index, data bit rate, temperature etc. The functionality of the demodulator under this different parameters setup is then expressed as measured values of data delay, data pulse width etc. as well as the oscilloscope screen- shot that holds the measurement time instant screen situation for possible off-line user inspection.

There are good reasons why store the measurement data in the database and not keep them using other conventional storage methods like office spreadsheets. Firstly, the amount of the measured data may become too large for the spreadsheet to swallow. Secondly, while the number of measured members continuously grows as the measurement experiment goes on, the user already wants to run the analysis on the data set that has been already measured. This is quite an easy task for database queries and SQL standard language for accessing and manipulating databases is on hand to be used.

The SQLite program has been selected for managing the measurement results database due to its simplicity of operation, possible embedding into other programs and, moreover, the program is preferred due to the availability in the public domain. Nowadays, the SQLite is most widely deployed SQL database engine.

The graphs are usually the best way to understand the data. They give a comprehensive presentation of the analyzed results and may give a hint on existing dependencies within the measurement data members. In this article the Gnuplot tool capabilities will be used to explore data dependencies. It provides a wide variety two- and three- dimensional plot types like line, scatter, bar, histogram, contour, surface or schmoo plots. It also provides flexible text labels to be sticked around the plotted data. Again, last but not least reason why Gnuplot program got user preference is because it comes with open source license.

The topic discussed in this article will, as the author believes, give positive a feedback on using the cited tools in scientific or engineering practice.

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Method of Evolutionary Fuzzy Clustering

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Cluster methods are widely used as a pattern recognition tool for the analysis of multivariate data in medical, industrial, financial and other applications. In these applications it is necessary to reveal the groups of objects with similar features and behavior and on the basis of defined groups to make a decision about new data object by estimating its similarity to one or another previously defined group. One of the major problems in clustering is the determination of the number of clusters in data. That cluster number specifies the further course of clustering algorithm and as a rule is unknown in advance. A widely known and simple approach to circumvent this drawback consists of repeatedly executing the clustering algorithm multiple times for different number of clusters and then selecting the particular number of clusters that provide the best result according to a specific cluster validity criterion [1]. Such a sequential procedure with increasing number of clusters may be efficient if the optimal number of clusters is "small", what is not be always fulfilled in practice. Therefore the development of methods for automatic determination of the cluster number during the clustering process is of particular importance.

In this paper the original method of evolutionary fuzzy clustering using genetic algorithm (GA) is proposed. The method enables in one GA run to find the near optimal data partition into clusters and automatically determine the cluster number during the clustering process. GA encodes the solution of clustering task, using the variable-length chromosomes, which provides the extended search for the coordinates of cluster centers. The number of clusters together with the corresponding cluster coordinates simultaneously evolve in the process of GA execution. As the chromosome length is variable, a single population can contain the individuals, encoding the different number of cluster centers. The value of fitness function indicates the degree of goodness of the GA individual and the efficiency of the solution of optimization task. In proposed method the inverse value to the Xie-Beni [2] index is used as the GA fitness function $f = 1/XB(U,V,X)$. The optimal fuzzy partition corresponds to the maximal value of function f.

The main distinctive features of the proposed method are the following:

- Specially developed coding scheme of GA chromosomes.
- Specially developed crossover operation, which takes into account the constraints on minimum number of cluster centers.
- Using the Xie-Beni validity index as a fitness function.

The results of clustering were used as input to the procedure of classification rules' construction, which includes the determination of the antecedent and consequent membership functions for each rule. Each obtained cluster determines a single fuzzy rule and the new data object classification is performed using all the set of rules. The efficiency verification of classification rules, constructed on the basis of proposed fuzzy clustering method in comparison to standard repetitive *FCM*-clustering was performed on the artificially generated dataset *set_all* and on the dataset *Iris* from the international data archive on machine learning. The results have indicated the possibility of the new fuzzy clustering method not only to define automatically the number of clusters and to perform the data clustering with the better value of Xie-Beni index in comparison with clustering method *FCM*, but also to construct the fuzzy classification rules, that more accurately classify the data objects.

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NS-2 Extension for Dynamic Optimisation within Multiple VoIP Sessions over Wireless Networks

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Introduction. The Network Simulator NS-2 is a discrete event simulator targeted at networking research. It provides support for simulation of many protocols over wired and wireless networks. To investigate network performance, researchers can use an easy-to-use scripting language to design and configure a network, or a series of network tests, and then observe results generated by NS-2.

There is an urgent need for supporting Quality of Service (QoS) in 802.11-based WLANs. IEEE 802.11e goes some way towards meeting this need, however, severe congestion leading to unacceptable delays and packet loss can still occur.

In a broader context, our research aims to investigate how synchronized time implemented in end terminals and/or access points can aid in dynamically optimizing 802.11e parameters to improve QoS for VoIP. In particular this paper will look at the Network Simulator (NS-2), and outline how we plan to build an extension to the open source simulator which maps to our research. As well as running simulations, our broader research will involve constructing a wireless test-bed to back up our research.

NS-2. The Network Simulator version 2 [1] is an open source discrete event simulator used in networking research. NS-2 provides substantial support for simulation of TCP, UDP, 802.11 and other routing protocols over wired and wireless networks. NS-2 is written in C++ and has an OTcl interpreter as a frontend. OTcl should be used for configuration, setup and "one-time" actions, and in situations where existing C++ objects can be manipulated. Coding in C++ is required where processing of each packet in a flow is required, or if the behaviour of an existing C++ class needs to be changed in ways that weren't anticipated.

The basis of our research is to quantify the extent to which synchronized time can improve the QoS of VoIP over wireless networks. With precise knowledge of each way M2E (Mouth to Ear) delays and the facility to assign priority (tuning EDCA parameters) to VoIP sessions that have large one-way packet delays [3] (due to long geographical distances, e.g; a voice call between Ireland and the US), relative to VoIP sessions with lower delays, we can equalize overall M2E delay (wireless to AP and beyond) for all VoIP sessions in an 802.11 BSS. We will quantify any improvements using the ITU-T E-Model [4]. We have run preliminary NS-2 simulations for multiple simultaneous voice calls, which have shown clear improvements in delay values for VoIP sessions that have preset network delays, where we assigned priority to those sessions at the Wireless MAC layer.

Current Status. We are currently finalising the design of the extension to NS-2. On the initiation of a simulation with multiple VoIP sessions the patch will work as follows;

• Multiple VoIP sessions begin

• Calculate one-way delays for each individual session

- Two scenarios exist: Either uplink + downlink or just uplink delays will be considered *(downlink traffic would require multiple virtual queues within the AP)*. This obviously has implications for real test-bed implementation.

• An algorithm will prioritize VoIP sessions, based on total one-way delay values

- Tune EDCA parameters to optimize QoS (based on our NS-2 simulation results) so that total M2E delays are equalised.

• Continue to monitor session data (Delay, Loss Rate etc..), and maintain an equalisation of delay values among sessions.

This functionality can be expanded to further break down the one-way delay into ([Wireless MAC] + [non-Wireless MAC]) delay, in order to calculate delays that are due to varying geographical distances. This patch will be developed in C++, and will build on the NS-2 802.11e extension [2] on which we have run our initial VoIP simulations.

Future Work. Along with the NS-2 extension we are constructing a wireless test-bed to re-produce results found in NS-2. Most of the hardware for the test-bed has been acquired and some preliminary wired to wireless VoIP tests have been carried out on the NUIG LAN with NTP time synchronization implemented at each end.

We plan to have our test-bed fully operational by early 2010, in order to re-create results obtained in NS-2. We expect our NS-2 extension to be complete by summer 2010.

This research is done within the Performance Engineering Laboratory (PEL) group at NUI Galway. PEL is a widely distributed research group based principally at University College Dublin and Dublin City University with links to other research institutes including NUI Galway. See http://pel.it.nuigalway.ie.

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Emulator of GPS Receiver

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This work deals with realization of software for GPS receiver emulation in accordance with documentation of navigation appliance GPS 18 made by Garmin company. Function of software is based on transmitting of standardized sentences via serial interface in accordance with NMEA0183 protocol.

Determination of geographic position was important for people in the world for long time ago. It was most interesting for sailors in the past, now is it used in all branches of traffic and a lot of another places. Thanks to the development of electronics, satellite systems and in effect of ambition people to travel and orient in the unknown places, there was a place for nascency of navigation systems. Most used became American navigation system GPS (global positioning system) thanks to his global accessability and relative high accuracy. With development of this navigation system many companies start producing various navigation devices, which can inform user about his current geographic position in graphical form through integrated display or in sentences form which are transmitted by its communication interface. Navigation devices are in present unthinkable components in our lives, therefore is advisable to learn how they are working. On this purpose can be used software for emulation of GPS receiver.

The main aim of this work was to create software application which can be used for definition of main transmitted sentences, which are used in communication between navigation device and computer. Software was created in accordance with manual for navigation device GP18 from Garmin company, which deals with this problem. Communication between devices is described in NMEA 0183 standard. The National Marine Electronics Association (NMEA) is a non-profit association of manufacturers, distributors, dealers and others interested in peripheral marine electronics occupations. The NMEA 0183 standard defines an electrical interface and data protocol for communications between marine instrumentation through three types of standardized sentences. The correct function of emulator of GPS receiver, is to composite data from individual edit boxes into a standardized sentence, and transmit this sentence via required communication interface. As a communication interface can be used serial link RS232 or Ethernet (sentences are accessible through UDP server). The sentences are composited in accordance to NMEA0183 standard, and they include data about geographical coordinates, speed, altitude, dilution of precision, information about satellites and about signals from them received, UTC time, date and another.

Basic transmitted sentences:

GPGGA GPS Fix Data. Time, Position and fix related data for a GPS receiver

GPGLL Geographic Position – latitude/longitude

GPGSA GPS dilution of precision and active satellites

GPGSV satellites in view

GPRMC recommended minimum navigation information

GPVTG Track Made Good and ground speed

PGRME Estimated Position Error

PGRMF Position Fix Sentence

Received sentences:

Program can receive sentence via serial link too, and basically this sentence, it can change its behavior.

PGRMI - Sensor Initialisation Information – this received sentence can change values of geographical coordinates and time

PGRMO - Output Sentence Enable/Disable.

For the possibility changing of geographical coordinates program includes function simulation of motion, where is change values of latitude and longitude based on start location, destination location and current speed. User can change simulation of motion in three different ways, scilicet vertical, horizontal and simulation of motion in both ways. For simplification work with program can user save data written in to edit boxes to the text file, and later can be data loaded from this file again. It´s a great opportunity to fill edit boxes with data from file at each new run of program. Because program may be used as education instrument, there was a big attention devoted to handling values of edit boxes. Handling is based on checking of minimal and maximal value and number of decimal places for number written in edit box. If is value incorrect, edit box changed its color to red, also is disabled transmitting of sentences. Function "handling of format" adjusts number with missing zeros to set correct format of number. Functionality of program was tested in three different ways, as transmitting sentences to serial port, communication via serial link and communication via Ethernet. Developed software application was realized in "Delphi" program environment.

Scoring System for Extraction Knowledge to Make Decisions for Giving Credit

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On the base of the credit histories of the bank, the scoring system starting from the social demographic characteristics of the potential borrower estimates the probability of his default. The system is destined to accumulate and analyze unregulated information on the base of input and output forms for various types of data. The system automates all the sequence of operations – from receiving a request for a credit in a distant shop up to making decision for giving the credit and forming the document package.

The software consists of several parts:

Back- and front-office of distant workplaces;

Scheme of documents circulation (sequence of passing questionnaires through the bank services);

Data base containing the information on borrowers and the history of making decisions on them;

System of support of making decisions on credit giving and analytic accounts.

In the process of automating, all links are enabled – shop operator, security service, credit bank inspector, scoring model to support making decisions on giving credit and to study the decision sensitivity to initial data change.

To prove a decision on credit giving in the scoring system, the model of classification tree and the model of logistic regression were used. We investigated these models statistically based on the same credit history of the borrower. A sample was considered that consisted of 1000 observations and 20 variables with information about the clients of one of the German banks. A client of the bank was presented as a collection of characteristics that he indicated in the request- questionnaire for obtaining credit. To continue the work, we analyzed these data preliminarily. The correlation matrices of predictors and the significance matrices of those variables according the solvency (Y) were obtained. After analysis of main statistics, the following variables were chosen from 20 variables: “cur_balance” (account current balance), “moral” (outgoing of previous credits), “savings” (sum of savings), “age” (age), “assets” (assets), “installment” (payment in percentage of income), “married” (social status / sex), “other_credit” (other acting credits), “duration” (credit duration).

The process of decision making includes constructively the following stages: (1) formalization of the problem and object description; (2) identification of alternatives; (3) choice of the best alternative; (4) analysis of sensibility of alternatives to changing initial values. To describe discrete processes being in this system and implemented by components with several stable states, linguistic variables are used. These variables take values from a finite set of size *m*. When these values are interpreted by positive integers, the transition to *m*-valued data {0, 1, ..., m–1} occurs. The multi-valued logic is the mathematical instrument for processing these data.

Table 1 shows a fragment of interpretation of the resulting variables (*Y*) and the predicative ones (*X*)

Table 1. Interpretation of attributes of thy structural function (fragment)

Variable	Description	Categories	Value
y	Solvency: 1 – solvent 0 – insolvent		
x1	cur_balance	No balance or debit	2
		0<=…<500$	3
		…>=500$ or account was checked in the last half a year	4
		No account current	1
x2	duration	Short-term	1
		Medium-term	2
		Long-term	3

Cascade Integrator-Comb Filter Frequency Response Optimization

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As data converters and digital circuits in software defined radio (SDR) become faster and faster, the application of narrowband extraction from wideband spectrum is becoming more important. Cascade integrator- comb (CIC) filters are the most frequently structure used in high-speed multi-rate digital receivers as a decimator or interpolator because their implementation in FPGA or DSP is relatively simple even if the sample rate change is very large. The CIC implementation of a decimation filter is the cascade of an integrator stage (otherwise accumulator), a comb stage, and a decimation process (decreasing of a sampling frequency by integer rate R) inserted between the accumulator and the comb stage. An interpolation is the inverse operation of decimation, it means that the structure of interpolator consist of a cascade: comb stage – increasing of sampling frequency – accumulator. The CIC filter can be implemented as the 1st order architecture (one accumulator and one comb stage) or as the N-th order system (N accumulators and N comb stages). Consider a decimation process in SDR. The transfer function of the CIC has to be derived with respect to the high sample rate (sample rate of accumulators) and than it has the form

$$H(z) = \left(\frac{1-z^{-RM}}{1-z^{-1}}\right)^N = \left(\sum_{m=0}^{RM-1} z^{-m}\right)^N , \tag{1}$$

where M is a design parameter called as the differential delay (usually set to 1 or 2). The frequency response with respect to the higher input sample is

$$H(f) = \left[\frac{\sin(f\cdot\pi\cdot RM)}{\sin(f\cdot\pi)}\right]^N . \tag{2}$$

This filter is a cascade of N copies of an RM-th order FIR filter whose coefficients specify a rectangular time-domain window with a linear phase.

The prepared paper will be focused on the analysis of CIC filters and their implementations. The decimation process with the output sample rate corresponding to the symbol rate (usually twice of symbol rate due to method of symbol timing synchronization) is the typical application of CIC filter in SDR for data demodulation. In this case, the bandwidth of the desirable incoming signal is the same as the bandwidth of the first zone after decimation. However, the magnitude of frequency response in the first zone is not flat (it is obvious from equation (2)) and the desired signal can be involved with these non-flatness of CIC response. Therefore, the main task of this contribution is the analysis and method of a compensation filter design which can be inserted behind comb stage. Ideally, the adjustment of the CIC frequency response requires the compensation filter with an infinite gain at the halfsampling frequency (respecting to the output sample rate of CIC), nevertheless such compensation filter can not be realized. In the paper, the compensation filters with approaching to ideal response, which can be practically implemented, will be shown. The compensation filter design is focused on simple implementation and responses with linear phase. In the paper will be analyzed a system stability too and discussed a problem of an aliasing.

Operation of IT/ICT Practice Frameworks

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The IT outsourcing service provider bears a great responsibility for a client's business success. Without a doubt, IT infrastructure is the heart of every business. The effectiveness of the IT is the effectiveness of the business. Therefore, the best known best practice of IT service management ITIL (*Information Technology Infrastructure Library*) often becomes one of the major acceptance criteria when companies hand over their business critical IT infrastructure to their outsourcing service providers. Typical examples are telecommunication companies in the world which are migrating to a NGN (*Next Generation Network*) based on IP technology. The IP-base technology is changing especially quickly, with new services appearing all the time while the life cycles of individual services tend to become shorter and shorter. With CSPs (*Communicating Sequential Processes*), the telecommunication network operation has usually been organizationally separate from the IT network operation. With convergence of telecommunication and IT technologies, the skills required become more flexible.

The TeleManagement Forum *(TM Forum)* has developed and published the eTOM (*enhanced Telecoms Operations Map*) which is business process framework that aims to describe all enterprise processes used by a service provider. A comprehensive reference point for process engineering this framework enjoys almost universal acceptance in the telecommunications world. TM Forum as well defined NGOSS (*New Generation Operations Systems and Software*) and along with eTom is extremely rich on detail and complicated beyond the highest level of aggregation. In addition, it is acknowledged that many practical problems must be addressed for successful implementation. Given the investments required to change organizations, software systems and the skills of people, many of the benefit derived from compliance to NGOSS ale likely to be experienced only in tactical or even strategic time frame. In a parallel effort, the IT community has developed the ITIL (*IT Infrastructure Library*) a widely accepted approach to IT service management. ITIL was developed by the Central Computer and Telecommunications Agency of the British government and adopted by many companies as their internal IT best practice. However, for many of the ICT (*Information, Communication and Technology*) service providers, especially in the Telecom industry, eTOM is more widely applied and deployed for end to end service delivery and support. This paper provides an overview of the similarities and the differences between ITIL and eTOM and how service providers can control the strength of both to deliver high quality services.

Synchronisation Challenges for Wireless Sensor Networks

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Introduction. The concept of time and the units used to measure it provide a means to sequence and measure the duration of events. Over the last number of centuries the granularity to which time can be recorded has increased from tens of minutes to nanoseconds, and with the growth of and reliance on Real-time computer systems and high-speed communication technologies, such granularity is increasingly important. The duration of a significant event or the interval between significant events in the life of a (perhaps distributed) computer system may be in the order of microseconds or less. It is this fact that drives us to find more accurate and reliable means of measuring and recording time.

In order to keep some order in the world, systems and people most conform to some time standard. The official time standard is Coordinated Universal Time (UTC) time which is based on International Atomic Time (TAI) time with the addition of leap seconds to account for the Earth's slowing rotation. Most computer systems rely on quartz crystal oscillators for time keeping. A quartz clock's performance is dictated by the manufacturing techniques used to create it and the physical environment it operates in. As such, these clocks are subject to both permanent and temporary frequency offsets which lead to subsequent drift from true time.

In a distributed computer environment, the challenge is even greater and conformance to a time standard such as UTC requires the use of a time protocol to keep the computer clocks synchronised. It is not enough to synchronise computer clocks once and allow them to free run because typical computer systems utilise inexpensive quartz crystals.

Time Protocols. Over the last number of decades a few notable time protocols have been created, each one suited to a different environment. The complexity of the operating environment dictates the protocol complexity and accuracy achievable by a time protocol and as such each protocol is tailored for a specific setting. The two most distinguished protocols are the Network Time Protocol (NTP) [1] and the Precision Time Protocol (PTP) [2]. The NTP protocol is designed to function in variable latency and dynamic packet switched networks. Its incorporated algorithms utilise proven statistical methods to determine the most accurate time reference from an array of time sources. The PTP protocol, in contrast, is designed for local managed networks that require very high accuracies. Rather than utilise sophisticated algorithms and numerous time references, PTP takes advantage of the fact that the network in which it operates is controlled. In an optimum PTP setup, a network's composite nodes are PTP-aware and incorporate dedicated hardware to allow for maximum accuracy.

While NTP and PTP differ to a high degree in their method of operation, they are related by the fact that they assume symmetrical path delays from the source to the destination and vice versa. It is this fact that often leads to a significant degradation of each protocol's performance when forced to operate over wireless networks. Wireless protocols such as

802.11/802.15.4 operate in a limited frequency band and as such dictate that hosts share this band in some manner. In a situation where a large numbers of hosts are competing for the same wireless medium, there can be significant and asymmetric network delays due to Medium Access Control (MAC) procedures. Experiments performed by the authors over 802.11 confirm that these delays have a significant impact on time protocols. NS2 simulations [3] of NTP clients competing with web clients over 802.11 networks indicate time errors of over a hundred milliseconds, well beyond the requirements of many time sensitive applications.

Synchronisation over Wireless Network. An area in which time synchronisation over wireless networks is of critical importance is Wireless Sensor Networks (WSNs). A basic WSN consists of a group of wirelessly interconnected autonomous sensing devices. These devices are dispersed over a particular environment of interest and programmed to acquire and distribute specific physical data. In many WSNs the time at which a piece of data is acquired is crucial for allowing meaningful analysis of the data. A time protocol designed for a WSN is subject to a great deal of constraints since the nodes it must run on are severely limited in terms of memory and processing capabilities. In addition the protocol must achieve a certain degree of accuracy over a communication medium which can be subject to high contention delays. Current time synchronisation protocols for sensor networks include Lightweight Time Synchronisation (LTS) [4], Reference-Broadcast Synchronisation (RBS) [5] and Timing-sync Protocol for Sensor Networks (TPSN) [6]. At their core, each protocol uses a very different technique to achieve synchronisation but all are related by the fact that they require a time reference that conforms to an external timescale such as UTC.

In a widely distributed WSN composed of multiple nodes, the task of synchronising to an external time reference is bestowed on a few superior nodes with enhanced capabilities. These nodes typically synchronise to a GNSS (Global Navigation Satellite System) such as Global Positioning System (GPS) via an integrated GPS receiver. This method provides the best accuracy but is limited by the fact that receivers require a clear view of the sky. This reality and the (relatively) high cost of GPS hardware make it an unsuitable technology in many settings and as such an alternative approach is preferred. The research proposed in this paper aims to identify and remedy the shortfalls of distinguished time protocols, such as NTP and PTP, so that they operate effectively over widely distributed wireless networks. The enhancement of these protocols would allow them to recognise and adjust to varying traffic conditions and provide the accuracy that they currently attain over wired networks. Subsequently, these protocols would be used as a feasible alternative for synchronising WSNs to an external timescale.

This research is done within the Performance Engineering Laboratory (PEL) group at NUI Galway. PEL is a widely distributed research group based principally at University College Dublin and Dublin City University with links to other research institutes including NUI Galway. See http://pel.it.nuigalway.ie.

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The Expert Information System for the Third Party Logistic Provider's Choice

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Direction "outsourcing of logistical services (Third Party Logistics Services-3PL/Logistics Outsourcing) " is defined as transfer of a part (all) of logistical functions to the external service logistical organizations and a way of optimization of enterprise's activity due to transfer of not profile functions (partial or full) and corporate roles to the external specialized companies (3PL-providers). Abroad outsourcing is practiced for a long time and more often in sphere of selling and distribution of finished goods.

The urgency of outsourcing of logistical services and question of a choice of the logistical provider consists that at transition from the market of the seller to the market of the buyer use of logistical intermediaries allows to reduce investments into auxiliary processes, to reduce their cost price, there is a flexibility of reaction to changes inside of the company and outside of it, financial parameters improve. It is obvious, that the companies aspire various ways to lower all logistical costs, and outsourcing is one of effective ways to make it. Packages of given services of logistical providers can include transportation, warehousing, integration and disaggregation cargoes, readdressing, storage, recommendations at the choice of rational logistical decisions, consulting, engineering, marketing, information services.

The market of logistical outsourcing is forming in Belarus and questions of its development are very actual. This process is influenced with following factors:

- Globalization of supplying and marketing networks and globalization of trade;
- Management of supply chains;
- Pressure of consumers;
- Application of outsourcing as business-model of the organization.

The developed software product has allowed to simplify decision-making at the choice of 3PL- provider for employees of a company and to optimize firm activity in organization and account of cargo transportations and logistics in whole. The given software is developed by a principle of dialogue with user, does not demand special training of the personnel to work with it. The simple user interface, convenient arrangement of action buttons, opportunity of fast access to any kind of information do the given product an integral part of enterprise success achievement.

Use of software product reduces expenses of logistics department work - instead of daily viewing papers and monthly forecasting of demand workers trace the data brought in the program, choose and analyze the necessary information on providers, shipments, transport charges, payments, analyze seasonal prevalence of sales under the constructed schedules. So, the developed program provides reduction of expenses of personnel work - management gets higher level, the firm can devote more time to work with clients, to studying of the markets, advertising activity; also program provides reception of the proved rational decisions at the choice of 3PL- provider. It is important for a firm as for today the economy of means (decrease in monetary, time charges) is simply necessary for developing companies in conditions of the socially-focused economy. A firm which wants to estimate and choose the partner for business can use developed application, besides investors who wish to enclose money in 3PL-providers development can use software product and solve the same question at the choice of the best logistic provider.

A GSM-Based Infrastructure for Large Geographically Distributed Sparse Sensor Networks

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This paper describes the concept of a large geographically distributed sparse sensor network. It consists of a 3-tier architecture, whereby individual wireless sensor nodes (WSNs) and monitoring stations (MSs) communicate via a device authentication and management station (DAMS). The sensor nodes are optimised for battery efficiency. The DAMS allows the simple management and configuration of sensor networks, while providing a centralised database that is used for the storing, dispatching and processing of sensor data.

Overview. Large geographically distributed sparse sensor networks are typically deployed in applications like vehicle tracking systems and habitat monitoring of environmentally sensitive areas [1]. Due to the low sensor density of such networks conventional short range wireless network technologies like IEEE 802.11 or ZigBee are not feasible. Instead they are usually implemented on top of existing GSM networks. The obvious advantage of such a network is, apart from its broad coverage, the reliability and build-in routing capabilities that are managed by the service provider.

On the other hand GSM-based networks are shared between numerous users and are open networks. This makes security and authentication a critical factor, particularly when such mobile phone networks become the backbone for a sensor network [2, 3, 4].

This paper describes a sensor network architecture, which takes advantage of the services provided by GSM networks, while adding a scalable device management, data encryption and source authentication component.

The proposed system is an infrastructure-based network. It is based on a 3-tier architecture that consists of

- a number of wireless sensor nodes (WSNs) that form one or more logical networks (LNs),
- a central device authentication and management station (DAMS), and
- one or more monitoring stations (MSs).

Data exchange between these components is done via encoded SMSs. This messaging service has been chosen over GPRS due to the poor coverage of the latter in some rural parts of Europe. Conceptually data is transmitted bi-directionally between the WSNs and the monitoring stations via the DAMS as the gateway in the middle-tier.

1. This paper will address the following issues:
2. The overall system architecture

3. Hardware and software elements of the WSNs, the DAMS and the MSs [5]

4. Security and device authentication throughout the network

5. Sensor network services [6, 7]

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Computer-Aided Diagnosis of the Pulmonary Nodule

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Lung cancer is the most common cause of cancer death in both men and women in the world. Lung cancer is the most commonly diagnosed cancer worldwide, accounting for 1.2 million new cases annually. Lung cancer is one of the most prevalent cancers, with an estimated 173,770 new cases and 160,440 deaths attributed to the disease in 2004 in the United States. In 2006, the LC caused over 158,000 deaths—more than colorectal, breast, and prostate cancers combined.

The advantages of imaging techniques for the pulmonary nodule detection and differential diagnosis are the noninvasive nature of these manipulations, which causes no discomfort and pain to patient, the possibility to easily collect and share the images for diagnostic and research purposes. These methods are an addition to the existing methods, which provide the biochemical picture of the disease or serve as the ultimate expert diagnosis (pathohistology). Adequately applying the pulmonary nodule imaging diagnostic methods, it is possible to reduce the number of biopsies and other painful medical procedures which patients are inevitably subjected to.

The analysis of the performed research shows the evidence that various *computer-aided diagnosis* (CAD) systems can be successfully applied for chest radiographs, *computed tomography* (CT), *magnetic resonance imaging* (MRI), *positron emission tomography* (PET). These modalities can serve as a useful tool for a practicing medical professional facing the burden of routine diagnostic job.

A CT scan is commonly used to evaluate whether lung cancer is present in the hilar and mediastinal lymph nodes, liver, and adrenal glands, but its accuracy in identifying mediastinal lymph node involvement is suboptimal (sensitivity of 40 to 65% and specificity of 45 to 90% versus either a PET scan or mediastinoscopy). Most importantly, CT as a rule misses small metastatic foci that do not result in mediastinal lymph node enlargement. Mediastinal lymph nodes that are normal in size have an 8 to 15% probability of having metastatic disease, whereas mediastinal lymph nodes that are 1 to 1.5 cm, 1.5 to 2 cm, and greater than 2 cm in size will contain metastases 15 to 30%, approximately 50%, and about 90% of the time, respectively. PET which uses 2-[18F] fluoro-2-deoxy-D-glucose to identify areas of increased glucose metabolism in lung tumors, is more sensitive than CT in staging lung cancer it has a sensitivity of 83%, specificity of 96%, and negative predictive value of 96%. However, increased glucose metabolism also occurs with inflammatory processes. Obtaining both a PET and CT scan can enhance the accuracy in the staging of lung cancer. PET also enhances detection of bone, liver, and adrenal metastases. However, if treatment decisions are to be based on PET scan results, positive PET scan findings require pathologic or other radiologic confirmation. MRI is another option for detection of malignant pulmonary nodules and in particular based on the image analysis. Recent experience with MRI points to its potential for detection and characterization of pulmonary nodules while avoiding ionizing radiation.

In a brief review article we have described the concepts, mehods and some definitions of the modern CAD and thoracic radiology and the facts derived from the available literature on the topic of pulmonary nodule differential CAD published during 2004-2009. The review of the literature shows that a significant experience of the CAD clinical application has been gained. The unresolved problems of CAD are the ways to improve the techniques of the image analysis to increase the sensitivity of diagnostic strategies, to broaden the spectrum of the differential diagnosis. Perhaps the implementation of samples of miscellaneous verified pathologic entities to cover the entire pulmonary nodule differential diagnosis extent may be the rational effective strategy to decrease the rate of false-positive diagnostic results. Apparently, this amount of work requires an extremely large database of the correctly detected and verified images. Another challenge is the performance of the large-scale evidence-based trials in order to discover the advantages and disadvantages of the computer-assisted approaches to diagnosis of malignant pulmonary nodules.

On the Static ADC Non-Linearity Evaluated for RSD Cyclic A/D Converter Case

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One of the recent approaches to test A/D converter performance is the so-called Servo-Loop Method. This method is aimed at the non-linearity extraction of static ADC transfer curve. In this paper, we prove an advanced Servo-Loop version focusing on behavioral and transistor-level example of the Residual Signed Digit (RSD) cyclic A/D converter design. The background of the considered Servo-Loop version was proposed in [1]. In this paper, we establish a Virtual Testing Environment (VTE) built on Verilog-A implementation of the Servo-Loop unit fully integrated into Cadence design environment.

The Servo-Loop method [2], [3] is widely used for direct A/D converter test and measurement. However, to implement this method in a simulation way into advanced IC design tools, improvements of the standard implementation are strongly advised as to increase the performance and efficiency of the search algorithm. We successfully made such improvements in [1], significantly accelerating the Servo-Loop convergence. The subsequent task being the main topic of this paper is to develop a Virtual Testing Environment (VTE). The VTE proposed incorporates the Servo-Loop core and creates a convenient user interface for IC design engineers involved in the A/D converter design and verification. Particularly, the VTE is dedicated to solve the following tasks. First, it is capable to simulate a behavioral ADC model annotated by a priori known circuit error sources and to extract INL and DNL performances invoked by these errors. The individual error source contributions to the total ADC performance can be then evaluated. In such a way, the most crucial parameters for a given ADC circuit implementation are pinpointed, enabling a system-level design optimization. Secondly, the VTE enables to run full transistor-level simulation necessary for the verifications carried out in the IC design practice. At this point, the accelerated Servo-Loop convergence offers a significant advantage during the IC verification process.

Our paper is expected to be partitioned as follows. In Section 2, the main advantages of the considered powerful Servo-Loop implementation are summarized. In Section 3, Virtual Testing Environment idea is presented, describing the Verilog-A system background for use with Cadence design environment, particularly the Spectre circuit simulator. In Section 4, we demonstrate the system architecture and behavioral model of the RSD A/D converter used as a Device Under Test (DUT) for the proposed VTE. Section 5 deals with the most important verification results of the ADC design prototype, discussing various models from fully behavioral to advanced circuit-similar level. As a preliminary conclusion (Section 6), we can state that the presented paper brings the most significant results of the ADC simulation procedure, including the description and setup of the testing environment. The VTE proposed is dedicated for the semiconductor industry practice and it already has been found as a useful tool for IC design engineers.

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Analysis of algorithms of decision-trees making

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An important part of the development of the information society is the need to work with the growing amount of data. Companies, financial and scientific institutions store these data in databases in the hope that they are hiding useful information. Therefore it is a natural consequence that scientists from different disciplines focus on finding methods that would be able to discover knowledge in large volumes of data.

The data in the database originate from the real world. We collect them by identifying of objects. This identification is based on their measurable characteristics. These characteristics can be described by attributes. There are independent and dependent attributes. Dependent attributes are function of independent attributes. We call this function classification model [2, 8].

One form of representation of the classification model are decision trees. You can imagine the decision tree how a simple data structure. This structure contains nodes, branches and leaves. Nodes represent independent at-tributes. Branches are values of the independent attributes. Leaves represent the outcome of the dependent at-tribute. One path coming from the tree root to any leaf represents the classification rule: IF condition THEN conclusion. The condition is represented by the conjunctions of constraints on the independent attribute values. Conclusion of the rule is dependent attribute value.

Classification rules have some advantages. These rules are easily understandable to people and quickly processed by the machine learning. They are optimal for implementation in expert systems [4]. Therefore, the classification rules are often used for classification [3].

The main objectives of this work were to make a software package designed to create classification models and analyze their effectiveness. In this package are implemented several methods of knowledge discovery (methods of decision trees induction [7] and statistical classification [6]). Advantage of the software is better classification accuracy, because it realizes the choice of best methods for each specific database.

The publication includes also the results of experiments on several databases from Repository [1]. The classification models with the regard to their accuracy are rated on the basis of the training databases. From the measured results we can make recommendations, which model is suitable for which type of database on the basis of their characteristics. Experimental results show that our software provides better-summarized classification accuracy for training databases.

The project is divided as follows: The first section is referred to the issue of knowledge discovery and role of the decision trees and classification rules as the solution of the classification problem. The second section de-scribes our software package. In the last section are explained the results of the experiments.

Acknowledgment

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Videofilters for Digital TV

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The transition from analogue to digital television has brought significant changes and an application of new building blocks in the signal chain. Here the fast A/D and D/A converters and anti-aliasing and reconstruction video-filters play very important role regarding to the keeping quality of the processed video-signal. In the following we discuss properties of the video filters and possibilities of their design and circuit implementations. The main attention is devoted to the approximation problem and a realization using voltage-mode and current-mode building blocks.

Requirements for video-filters depend on their purpose and application areas. As mentioned, there are two main groups - anti-aliasing and reconstruction filters, which are further differ with respect to the format of video-signal. The available video-formats are RGB, component video, composite video and RGB PC graphics. The filter requirements include not only conditions for an attenuation frequency response, but also for group delay frequency response, and are more precisely specified under recommendations of ITU, NTSC, PAL/DVB, SMPTE and VISA, see e.g. [1], [2].

Successful video-filter design requires efficient solution of two key tasks - first, find the appropriate transfer function approximation, and then choosing the appropriate implementation of the filter with respect to its working frequency range, which is approximately 75MHz for HDTV applications.

The first task is usually solved using standard approximations, mostly Butterworth type - see e.g.[3]. Such a solution is not optimal, as it does not respect the requirements to the group delay. Our paper provides a different approach, based on non-standard approximation obtained using an algorithm that simultaneously accepts both attenuation and group-delay requirements. The algorithm and its implementation in MAPLE program has been presented in [4]. A result is a transfer function of a lower order than the Butterworth one. In addition, it shows better group-delay frequency response and a sufficient attenuation margin.

A solution of the second task is strongly influenced by requirements to the simplest integrable realization, less sensitive to the component value tolerances and minimized power consumption. Presented implementations are mostly based on cascade realization using known single amplifier biquadratic sections (SAB). The bottleneck of this approach is the high requirements to the used amplifiers, especially their real behavior, that negatively affect the signal suppression in filter stop-band. The article further discusses the filter implementation using the SAB based on CCII current conveyor. Such a solution we consider more appropriate, both for the implementation of voltage- and current-mode. In the following, a design based on LC prototype simulation using lossy FDNR is discussed and compared to the cascade realization. All these considerations are demonstrated on the practical example of SDTV video-filter design.

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Emotions as a Component of Prosody

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Prosody is very important for any kind of synthetic speech. As a matter of fact, improper prosody is one of the factors which make natural and synthetic speech sound differently. Apart from frequently used prosodic parameters, such as fundamental frequency, speech unit duration and intensity, other parameters related to prosody, for example voice timbre, are also studied. Timbre is related to the expression of emotions in speech, but can be impacted by pathological changes as well. The timbre, characteristic of each individual, is based on higher formant frequencies, which are a result of signal modification in vocal organs, reverberation in cranial cavity, but also lip and tongue position variations. For these reasons, speech analysis can be a supporting tool for the diagnosis of some neurological diseases and one of the means for evaluating the therapeutic process. A direct expression of emotions is problematic in Text-to-Speech (TTS) synthesis. Speech corpora used in synthesis have been based on neutral speech so far. Emotions are coded in speech signal only, not in text. They can serve as one of the input parameters for prosody modelling by artificial neural networks (ANN) in TTS synthesis. The prosody will not reflect any specific emotion, but will increase the naturalness of speech. Emotions can be expressed in larger speech units only, such as phrases and sentences. In both of these tasks, however, it is necessary to find a method of describing emotions so as to implement them in computer applications. A generalization of the way in which people perceive sound is one of the possibilities to model prosodic parameters. This approach could be a basis for emotional speech modelling as well. I will proceed from the way in which frequency changes are perceived in speech signal. Difference tones (or combination tones), which characterize tone proximity, play an important role in the perception of emotions in speech. We base our reflection on the analogy with music theory, assuming that the most important factor in human perception is the perception of harmonic series. Harmonic components of speech signal represent a significant part of the formants, and their changes will rank among ANN input parameters. Another important parameter of emotional speech are temporal changes of speech segments, i.e. tempo changes (variations in the length of individual segments within the utterance). The change of tempo at the end of an utterance is conditioned physiologically. Furthermore, intensity as the parameter which is considered as the least important in prosody, has more importance in emotional speech than in neutral speech. Intensity helps distinguish passive and active emotions (unpleasant and pleasant). Sadness or boredom rank among passive emotions, pleasure or anger among active emotions. Describing emotions mathematically is difficult. However, we can assume that the use of artificial neural networks is helpful in analysing and processing them. In our first experiments, we have used four basic emotions: pleasure, sadness, fear and anger. To obtain an accurate classification of emotions, we shall apply supervised self-organizing maps in our research.

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Improved Current Conveyor Optimization Via Differential Evolution

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This article refers recent results of author's work in the field of optimization of analog electronic circuits. The main task of this work is to find acceptable parameters of known circuit topology such as geometric parameters, supply voltage and bias voltage values. In other words it means to find a minimum of a function which is multicriterial multimodal, non-continuous and hard to evaluate.

To reach such a solution we have used optimization system, developped at our department, which links the differential evolution algorithm with a spice-like simulation system Ngspice. It helps us to evaluate the fitness function, which is computed directly within Ngspice. This solution makes us possible to work with recent libraries and models at a transistor level, such as BSIM3,4, EKV etc. The input and output of circuit block which is to be optimized is also spice-like textfile and could be used for other or further optimization and simulation process.

The optimization part is based on differential evolution algorithm, which is recently well proved to find a satisfactory solution in a variety of technical optimization problems. Our system allows us to select one of the schemes of differential evolution algorithms. After several computations, we chose the typical „DE/current-to-rand/1/bin „ scheme. Although the computational time of this scheme is longer, it is able to reach a satisfactory result compared to faster variants such as „DE/current-to-best/1/bin". The last-mentioned was mostly very fast, but in many cases it was „caught" into the local extreme. Nevertheless in several cases it found desired solution in distinctively shorter time.

The main deal of the article is however to show that a design using proposed results of optimization is comparable to conventional designs. Generally it imposes requirements especially on the domain of definition of the objective function. To demonstrate them we now show the requirements for known topology of a current conveyor (CCII-). Basically, we need to know the approximate operation domain of such a circuit topology and main performance characteristics. We can divide the requested features into the basic and the extended ones. The basic features of CCII- were current and voltage transfer between corresponding clamps, input and output resistively of these clamps, transfer functions in voltage and current mode and the cut off frequency of the block. The extended characteristics were sizing and the rate of geomatrical proportions of transistors, saturation region of transistors, bias voltages in biased casode output and the bias current. There are also some requirements, which are not included yet. They are the stepping (discrete) size of geometric values and current flowing each branch of CCII, which can optimize total power requirements.

At the end of optimization process, we use professional Mentor Graphics design tool to verify proposed design and manually trim its sizing to meet selected requirements of the design kit.

Solutions found within this optimization flow are comparable to conventional design. This optimizational and design flow is able to find suitable values of unknown parameters to reach desired operation of given topology depending on our requirements. The recent work heads for the implementation of the differential algorithm into the Mentor Graphics design tool.

Simulation of the Selected Network Attacks on the TESLA Authentication Protocol

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This paper deals with the task of simulation the broadcast authentication protocols using Colored Petri Nets. CPN is a special instance of orientated graph that enables to describe data flows and information dependencies inside of modeled systems. Protocol TESLA (Time Efficient Stream Loss-tolerant Authentication) was taken as an example of broadcast authenticating protocol to show how Color Petri Nets can be used to create a functional model of the protocol. Broadcast authentication protocols can be used in many situations where is one transmitter and multiple recipients such as message exchange in sensor networks routing protocols, or the process of leader election in sensors networks.

Keywords-component: broadcast authentication; simulation; protocol; sensor networks; TESLA; Colored Petri Nets

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Voice Quality Estimation for Narrowband and Wideband Phone Communication

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Voice quality estimation for narrowband and wideband phone communication is very important for Next Generation Networks (NGN) and for mobile networks UMTS. The Next Generation Networks are built mainly on Internet Protocol (IP) stack. A major challenge for emerging wire-line and wireless IP-based networks is to provide adequate Quality of Service (QoS) for different services. To do this requires a detailed knowledge of the performance requirements for particular services and applications. The starting point for deriving these performance requirements must be the user. The paper is oriented primary to end-to-end voice and web-services. The parameter Mean Opinion Score (MOS) is possible for subjective valuation of all service types (voice, video, web-browsing).

The guidelines and planning examples of Recommendation ITU-T G.108 are based on the utilization of the E-model as described in Recommendation ITU-T G.107. The intent of this Recommendation is to demonstrate how the E-model can be used in end-to-end transmission planning for a wide range of local, national, multinational and transcontinental networks. For wideband codec the R factor is higher than 100. Empirical formulas were defined for conversion of R-factor to MOS and reversely in connection with implementation of E-model. These references are valid for interval of R-factor from 0 to 100. Wideband speech transmission in frequency band from 50 Hz to 7 kHz is now very perspective way for upgrade classical telephony service. This band can be used for common audio signal for example music and high quality voice. There is required some extrapolation smoothing for higher R-factor by utilization of codec working with band up to 7 kHz. The R-factor in scale to 100 or 130 can be used for other cases.

It stands to reason, that enlargement of frequency band at transfer doesn't bring real growth in articulation, because it will be already close to 100%. However total subjective impress will be better, because timbre of speaker's voice will be transferred better. But there is a question, if the functional implementation of codecs with frequency band 7 kHz doesn't shift sense of quality in the direction of higher requirements to telephone communication and when classic codec will be subsequently subjectively evaluated worse than till this time by keeping five-grade scale of MOS. The subject testing was realized for standard and wideband PCM codec in IP network for different values of loss packet rate. The added value of wideband codec for increase of voice quality was not demonstrated.

The paper will present new on-line program for calculation R-factor of narrowband, wideband and mixed networks and program for conversion MOS parameter to R-factor in scale to 100 or 130. The programs will publish on The Matlab Server at address http://matlab.feld.cvut.cz. The corrected formula for author original approximation of transform curve from 100 point scale R-factor to MOS parameter from 1 to 4.75 was used for programs. New values of equipment impairment factor for narrowband and wideband codec from new version of Recommendation ITU-T G.113 were used.

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Measurement of Electromagnetic Shielding Efficiency of Planar Textiles

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A great development of electronic, especially microprocessor and high frequency telecommunication engineering in last decades requires human protection and protection of sensitive electronic devices against electromagnetic interference and electric charge. A necessary condition of correct operation of all electric devices and systems is an ability of operation in environment with different electromagnetic sources. In addition, these devices must not influence its surroundings by its operation, i.e. they must not generate electromagnetic and electrostatic fields with disturbing intensities for other devices. This rule is also valid for dangerous intensities of electromagnetic and electrostatic fields from the electric devices staff point of view. Demands on hygiene and safety at work are standardized by appropriate national standards and public notice, which are harmonized with EU standards [1]. A fulfilment of these sophisticated demands is not so easy.

One of the most important possibilities of protection against electrostatic and electromagnetic fields is a variety called shielding. The shielding is a constructive instrument for attenuation of disturbing signal field in limited space. Technical instruments (construction) are called viewing hood or shielding. Shielding is nowadays used for protection of particular components, functional blocks and whole electronic devices, which can generate or receive electromagnetic signals. Shielding principles also protect human himself against dangerous fields or devices, which ensure health protection and health functions (e.g. cardiac stimulator). Shielding is one of the most effective ways of electromagnetic protection against power impulse or continual disturbance.

Protective clothing of staff of power high frequency devices and systems (e.g. antenna engineer) is used like additional instrument.

New types of textiles were developed in the project Be-Tex, which is focused on human and technology protection against high frequency radiation. These textiles can reduce an effect of intensities of electromagnetic fields.

This article is focused on realization measuring experiments of electromagnetic shielding efficiency of planar textiles in frequency range 30 MHz - 1,5 GHz. This measurement uses ASTM D4935 standard [2]. The goal of the measuring experiments is to realize reproducable measurement based on comparison of reference sample and measured sample.

The size of measuring experiments was minimized by using capacity coupling systems. It also supports minimization of environment influence. This is very useful feature, which does not require measuring in anechoic chamber. This is a great advantage in comparison with e.g. electromagnetic compatibility measuring.

A construction of measuring experiments allows measuring 133 mm sample mean and higher. It makes possible to measure special laboratory prepared samples of different technical textiles. These samples are prepared in laboratories of project partners VUB a.s. Ústí nad Orlicí and Nyklíček a spol. s.r.o. Nové Město nad Metují.

Samples of planar textiles are formed by technical textiles with addition of silver nanoparticles in the yarn. These samples are not crash-resistant. Therefore the measured samples of these planar technical textiles have to be prepared by new developed procedure. This procedure uses two designed preparations, which are employed in dependence on technical textile construction. Thermal fixation of textiles structure is used in the case of higher capacity of synthetic fiber with the aid of thermostatic radial preparation construction [3]. The second sample, which is especially designed for surface stable knitting, forms radial organized die. The measurements can be reproducable with measurement error under 0.3 dB due to designed and constructed preparations. This measurement error is sufficient in electromagnetic shielding efficiency measuring of technical textiles.

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Comparison of Directional Hearing for Cochlear Implants

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The cochlear implant is electronic device that bypasses a non-functional inner ear by stimulating hearing nerve by patterns of electrical pulses, so that sounds and speech can be medicated to profound hearing loss patients. All over the world, the bilateral implantation is up to date. Till this time, only one cochlear implant for one patient was used, so the patient had only limited two-dimensional orientation. This paper provides comparison of two-dimensional orientation ability for normal hearing and hearing with cochlear implant using simulations.

1. Introduction. The cochlear implant system is composed of two parts - implantable (receiver/stimulator) and external (speech processor). Sound is picked up by a microphone, digitalized and divided in segments. Frequency analysis is applied on each segment and band selection is undertaken. The signal energy in each band is calculated and the most important information is chosen, depending on the used coding strategy and patient's setting. The information is coded and transmitted into the implant via radio frequency. The information is decoded in the implant and the nerve fiber stimulation is carried out.

2 Ace Strategy. The Advanced Combination Encoder (ACE) strategy is the newest strategy used in Nucleus® 24 implant systems. The input of the ACE strategy is digitalized speech with sampling rate 1600 Hz. The output is information about current pulses and appropriate electrode which will be used for stimulation and defined stimulating rate.

2.1. Speech Reconstuction. Backward speech reconstruction is a process which translates the stimulation frames normally used for stimulation to the speech. The reconstructed speech could be used for simulations with hearing volunteers. One possibility how to reconstruct speech signal from current pulses is synthesis using superimposed sinusoidal signals. The signal reconstruction using the pure tone generator is based on the assumption that every audio signal can be approximately reconstructed using a finite number of pure tones. Because ACE give no information about phase of processed signal (only absolute value of spectrum is used), the reconstruction using superimposed sinusoidal signals uses no phase information, too. The reconstructed signal s(n) is given by the formula:

$$s(n) = \sum_{k=1}^{N} A_k(n) \cdot \sin(2pi \cdot f_k \cdot n) \qquad (1)$$

The frequency value f_k is given by the central frequency of k-th band of currently used speech strategy. The number of bands N is 22 same as in ACE strategy. The $A_k(n)$ represents the amplitudes of the pure tones at a discrete time sample n = 1, 2…L, where L is length of signal to be created. The length of time sample corresponds to stimulating rate. For each time sample n the $A_k(n)$ is:

a) equal to amplitude of current pulse stimulated in electrode k (if the k-th band is selected as maximum),

b) equal to zero.

3. Directional Hearing

For the localization of sound, which came to normal hearing people is mostly important time delay shift, as it is depictured on Fig. 1. The time delay is given by path difference d, distance of left and right ear L and by angle α. If we count with L = 25 cm and sampling rate fs = 16000 Hz, the time delay between left and right channel is -8 samples for $\alpha = 0$, and 8 samples for 〈 = 180 degrees.

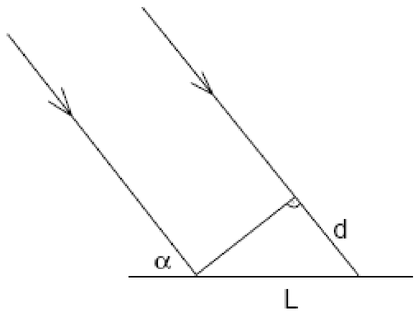

Figure 1: Time delay between left and right ear.

4. Directional Hearing Verification. For the directional hearing comparison for normal hearing and for simulation of hearing with cochlear implants, part of “Czech speech audiometric database” was used. 30 three syllable words recorded as monophonic wave files were selected. Firstly for each file, stereophonic signals with time delay -8, -6, -4, -2, 0, 2, 4, 6, and 8 samples were created. The left and right channels in these signals were exactly same except time delay between. Secondly the 30 original three-syllable words were processed using ACE strategy and reconstructed back to speech using algorithm described in section 2.1. Also for these reconstructed speech records, the stereophonic signals with 9 different time delay between left and right channel were generated. All 540 signals (270 for normal speech and 270 for ACE strategy) were randomly presented to 10 hearing volunteers using tool created in JAWA programming language and headphones. The volunteers were asked for direction of the presented signals and their answers were compared with real direction of speech.

Methods of Key Management Systems Used in Safety-Related Applications

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Cryptography is used to achieve confidentiality of information (protection against unauthorized sharing), to ensure the integrity of information (protection against unauthorized data modification, whether the protection against deployment viruses in the program) for authentication (proof of identity of the entity - a user, a process), to manage access to object with the proof of report origin (incontestability responsibilities) [1]. Cryptosystem security is given by confidential key, not by confidential algorithm.

The objective of cryptanalysis is to reveal the clear text from the cipher without any knowledge of the key. With successful cryptanalysis we could obtain an open text, or key. We may detect cryptosystem's weak point, which eventually will get an open text. Efforts to perform cryptanalysis is known as attacking.

There are four basic ways of cryptanalytic atack: Ciphertext-only attack; Know-plaintext attack; Chosen-plaintext attack; Adaptive-chosen-plaintext attack.

Attack with knowledge of the open text and attack familiar with the chosen texts are the most frequently used methods.

Current cryptography focuses on systems that are computationally unbreakable for which the algorithm is considered computationally secure or strong. The complexity of the attack is defined as: data complexity - amount of data required as input for attack, computational complexity - the time needed for attack, memory requirements - the extent of memory needed to attack.

Currently the most widely used cryptographic systems include a system of secret and public key. Both need key to realize algorithm (change element). The key is needed in calculating hash code in the key hash functions, such as MAC (Message Authentication Code). In cryptographic practice is currently used many methods of generation, transmission, archiving key from manual to automatic. Key management deals with this issue.

The article deals with the requirements for key management in commercial cryptography. Wing section is devoted to the description of methods for generating key mainly focusing on the RSA algorithm and the method of exponential key exchange, Diffie-Hellman's method. My article also devote to implementation, distribution and managing key in CrypTool software.

Inter-media Synchronisation for IPTV: A Case Study for VLC”

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Introduction. The tight integration and synchronisation of different multimedia streams presents opportunities for IP network delivery TV platforms, such as IPTV. This paper presents a case study whereby RSS (Really Simple Syndication) Feeds and Video are tightly synchronised using the media player VLC (VideoLAN Client). In particular it uses the embedded subtitles mechanism to facilitate this and examines two implementation scenarios: synchronisation at client-side or server-side. VLC is used both as a media streamer and as a media player in the client-side.

Common inter-media synchronisation challenges include synchronising video with audio or subtitles. Acceptable synch threshold between video and audio (also known as lip-synch) is a rich research area with differing viewpoints but typical values are between 15 and 45 milliseconds [1] with viewers being much more perceptive to audio leading than audio lagging. Inter-media synchronisation between video and subtitles is not as tight, mainly due to the nature of the subtitles as the viewer listens to the audio but has to read the subtitles. Subtitles display in text format the audio information, either in the same language or translated, and the viewer reads the subtitles at his/her own speed. Within VLC mechanism, video and subtitles synchronise to the millisecond relative to start of the video and indicates how long the subtitles should remain on the screen.

Implementation. In our VLC IPTV testbed, two approaches have successfully been implemented and are currently been analysed. Firstly, where RSS Feeds are streamed separately and embedded within the VLC client (Set Top Box). Secondly, where the TV and RSS Feeds are merged in the server and then streamed to the client.

```
1
00:00:06,000-->00:00:12,000
<p><font color="#00ffff">Fri, 2 Oct 2009 15:39:32 UTC</font></p>
<p>Pakistan Routs 'Cancer' Of Taliban Fighters</p>

2
00:00:16,000-->00:00:22,000
<p><font color="#00ffff">Fri, 2 Oct 2009 15:39:42 UTC</font></p>
<p>Rogue Bull Gores Runner To Death In Spain</p>
```

Fig 1: Example SubRip subtitles

MPEG-2 video format was chosen, being the main format for IPTV video streaming, although the latest platforms also use MPEG-4 [3]. VLC recognises different subtitle formats including SubRip. This format was selected for having 1 millisecond’s accuracy and some html text format [3]. An example of SubRip subtitles format can be found in Fig.1.

For VLC to display a subtitle file related to a video, both stored locally, all that is needed is for both files to be located in the same folder and to share the same name with the corresponding extension, .mpeg for video and .srt for subtitles file.

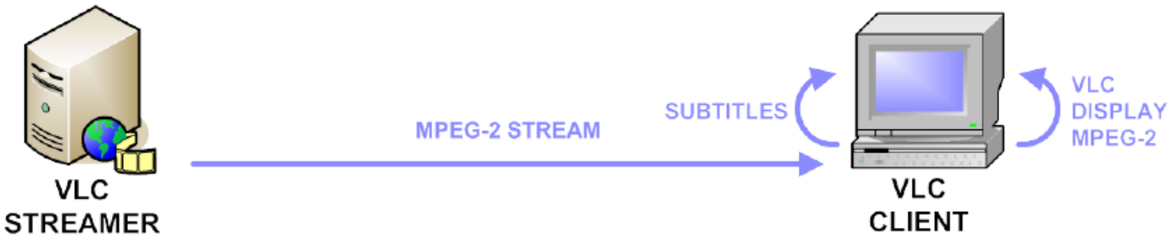

Fig. 2: VLC client-side alignment

VLC in client-side alignment works as follows: the server(s) stream(s) the TV file encoded with MPEG-2 and RSS Feeds independently. RSS is converted to subtitle format. On client subtitles are embedded within MPEG-2. See Fig.2.

Fig 3: VLC server-side alignment

VLC in the server-side alignment works as follows: RSS Feeds converted to subtitle format, the server transcodes the MPEG-2 video with the subtitles file, stored both in the server, and then streams it to the client. See Fig.3.

Results & Future Work. Initial results suggest that the way RSS/subtitles is handled (Client v Server) affects the end quality of the video. This is presumably due to the processing steps which are quite different i.e. streaming the video and then aligning it with the RSS (via subtitles) (client implementation) or embedding the RSS/subtitles within the video in the server and then streaming it to the client. This is currently been investigated further.

Other major issues to be considered and investigated in future work include the broader impacts of differing client/server implementation. These include server processing requirement, bandwidth requirement, potential for multicasting and STB functionality/cost.

In summary, this paper describes how the VLC media player is used to showcase inter-media synchronisation of logically and temporally related RSS and Video Feeds. Details are provided about VLC streaming combined with video transcoding and inter-media synchronisation between video and subtitles and how this affects the TV/RSS synchronisation project.

This research is done within the Performance Engineering Laboratory (PEL) group at NUI Galway. PEL is a widely distributed research group based principally at

University College Dublin and Dublin City University with links to other research institutes including NUI Galway.

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Estimation of Multi-State System Reliability

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Multi-State System (MSS) is used in reliability analysis for systems that have many (more than only two) states of system functioning [1, 2]. In a MSS, both the system and its components may experience more than two states, for example, completely failed, partially functioning and perfect functioning.

There are some principal problems in MSS reliability analysis. The first of them is analysing the probability of the system being in each state [1, 2]. Next the principal problem in MSS reliability analysis is to measure the effects of state changes of each individual component upon the system reliability [1, 2]. These measures are named as importance measures of MSS and they are used to evaluate the impact of a system component state change on MSS reliability [3, 4]. There are different tools for MSS reliability estimation. For example, Markov processes are used to analyze the system state transition process [2] or the structure function approach is used to investigate the system topology [1, 3]. We have been developed structure function tool for computation system reliability measures.

Nonbinary state property of the MSS and its component causes the difficulty in analysis of the system state probability and its change depending on component state changes. One of the decisions here is MSS transformation to a set of *Binary-State System* (BSS) [5]. In this transformation method, each component state is assigned a Boolean variable indicating whether or not the component is in that particular state. The major problems of this method are that many Boolean variables must be dealt with and the dependence among variables representing different states of the same component must be addressed [6].

In this paper two approaches for MSS reliability analysis are considered. One of them is based on the MSS reliability estimation by the methods that include transformation of the initial system to a set of BSS [5, 6]. Other method analyses MSS without additional transformation. In this case the following assumptions are made about the structure function: (a) it is the Multiple-Valued Logic (MVL) function [7, 8]; (b) the structure function is monotone and $\phi(s) = s$ ($s = s1\ s2 \ldots si$ and $si = sj$; $s, si \in \{0, \ldots, m-1\}$) [1]; (c) all components are s-independent and are relevant to the system [1].

Proposed method a MSS estimation is independent of initial type system and can be applied for reliability analysis different system that primary interpreted as discrete MSS. In this paper some examples of this method application are considered.

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Self-Organizing Map in Developmental Dysphasia Analysis

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A topic of this contribution is one of application of Self-Organizing Maps (SOM) in the field of medical research. It is a part of project finding a connectivity between children's neurological disorders called developmental dysphasia and the assessment of the degree of perception and impairment of speech.

We have tested the presumption that the developmental dysphasia can influence a shift of formant frequencies in spectral characteristics compared with the formant frequencies of healthy children. For that reason, our hypothesis says that the improvement in correctly classification means the improvement in health of the patient (i.e. the reduction of the developmental dysphasia).

We present results of analysis of vowels in 12 children patients with developmental dysphasia. In our tests we used vowels of 72 healthy children in age 4 -10 years and 12 ill children in age 4-7 years. Our aim was to find a shift in correct classifications of vowels in special type of Self-Organizing Map. This neural network is called Supervised Self-Organizing Map. The shift in correct classifications can support our hypothesis mentioned above.

We have three comparable records for each patient. Every record was made in three-month interval. A medical therapy was applied only for 6 patients. The special speech corpus was recorded directly at hospital. The comparative corpus, which includes isolated vowels, monosyllables and polysyllables, was compiled by neurological specialists as related to a medical therapy. Children were also assessed by clinical psychologist, using Gesell developmental diagnosis, Standford-Binet Intelligence Scale-IV revision and an additional test standardized for Czech language consisting of sound differentiation test, world differentiation test, auditory analysis and synthesis test. Our technical team from CTU recorded patient's speech [4]. Results of clinical psychologist and our results were compared.

In summary, the Supervised SOM should prove the ability to describe a trend of the developmental dysphasia.

We have taken advantage of software developed by the Laboratory of Information and Computer Science (CIS) in the Helsinki University of Technology. This software "SOM Toolbox" [3] is library for Matlab implementing the Self-Organizing Map (SOM) algorithms.

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Perfection of Information Infrastructure of Retail Trade Network

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In the given work there was made an estimation of influence of introduction of a system of intellectual analysis of data on financial and economic activity of the enterprise.

Optimization of activities of any structure of retail trade is possible not only by means of classical methods of business conduct, but also through perfection of information infrastructure of an organization.

The systems of intellectual analysis of data allow companies to get competitive advantages expressed in constant adaptation of manufacture and sale of production of the enterprises to changes in internal and foreign markets, achievement of high competitiveness of production and services, increase in sales in the existing markets and gain new markets.

The stored by this time experience of application of Decision Support System in processes of making various economic decisions has shown that common approach in general is not ideal. Therefore it is necessary to develop an approach to analytical researches that will allow to raise validity of decisions, in conditions of constant increase in data level in a company, in management of financial and economic activities of an organization.

The purpose of the given work is generalization and adaptation of experience of carrying out an intellectual analysis of data with use of an integrated and modified ABC–XYZ method, to improve the information infrastructure and increase on this basis the efficiency of activities of the network of retail shops.

Introduction of system of Business Intelligence (BI), realizing ideas of modified ABC–XYZ method in the existing corporate information systems (CIS) can solve these problems.

Similar systems provide business by essential advantages: possibility of forecasting of development of a company; deep analysis of markets and requirements of clients; freedom of giving accounts, their visualization; possibility of on-time reaction to changes of market conditions; rendering of the personalized services to clients.

In the given work an application of the ABC - XYZ analysis of the network of retail shops of Co Ltd "NTS" was considered. In carrying out of the intellectual analysis knowledge bases are used. The rules presented in the knowledge base of distribution of goods into groups allow to allocate priorities in management of product categories. On the basis of the rules the analysis of the product category of coffee has been carried out.

In the research it has been defined that the goods which fall into the group CcZz are characterized by permanently low importance and constantly low stability that demonstrates an unbalanced assortment. The goods which fall into this group make a small contribution to goods turnover and have considerable fluctuations of sales. Consequently, it is necessary to analyze these goods additionally and thoroughly before taking a decision about their exclusion from the assortment.

Everything aforesaid allows to arrive at a conclusion that the usage of the given integrated modified ABC-XYZ approach and its deepening with use of methods of classification of different kinds of production Data Mining will allow to identify priorities in their management on the basis of usage of the developed matrix of management. This in turn will allow to work out assortment matrixes which will fully meet the requirements of consumers.

The carried out calculations allow to arrive at a conclusion that investments in introduction of system of intellectual analysis of data at Co Ltd "NTS" will be compensated in 7.5 months and the enterprise will start to get profits. Introduction of the system will allow to reduce expenses of execution of laborious analytical works and creation of consolidated reporting and will also allow to optimize account payable, this will increase terms of use of cash assets. The increase of sales income will be executed due to decrease of the rests of production at the warehouses, exclusion of the unprofitable goods, application of effective marketing campaigns and calculation rules. The managerial staff will be able to accept competent managerial decisions based on extensive statistical data, with Data Mining methods they will get notable advantages in competitive struggle.

On the Design of ESD Protections

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Electrostatic discharge (ESD) can occur everywhere we can look at. Walking across carpet can charge our bodies up to thousands of kilovolts [1]. ESD has impact on semiconductors industry by increasing their cost. Wafer fabrication is not the only place where the semiconductor device can be attacked by ESD shock. A lot of operation steps have to performed between fabricated wafer and device in package placed into final modern electronics. The device can meet ESD event at all these process steps. So ESD measures have to implemented inside integrated circuit (on-chip ESD protection) and assembly environment as well.

Increase in performance and density of integrated circuits (IC) come from making the transistors smaller. Smaller devices are heated to higher temperature at given amount of ESD energy. Similarly, thinner gate oxides cannot sustain higher voltages generated by ESD events. Progress has been made in understanding the different types of ESD events damaging integrated circuits. This fact led to developing test methods which allow to characterize and to compare ESD robustness at the same conditions.

This paper focuses on design of on-chip ESD protections for radio-frequency (RF) chips. A lot of papers and patents related to design of ESD protection were published. One of the most often used ESD protection is so-called gate-grounded NMOS (ggNMOS). This popular ESD device can be easily used for protection of digital output buffers, gate inputs, output of regulators, power supply pads. This device allows to reach 2kV ESD level mostly requested on IC by industry customers simply.

Unfortunately, it is not so easy to use this ESD element for protection of RF input/output pads operating at frequencies in order of GHz. The reason is that a several improvements have to be implemented on layout design of ggNMOS due to progress in advanced submicron CMOS technologies. The main process step leading to degradation of ESD protection was introducing silicide-clad drain and source diffusions of the NMOS transistor to reduce the resistance and to increase speed of the transistor [2]. From ESD design point of view, this step is equivalent to bringing drain contacts closer to gate and diffusion edge. This fact results in reducing ballasting resistance. The consequence is degradation of current uniformity during discharge of ESD energy. Thus, as soon as a hot-spot is localized at drain diffusion edge, there is a small resistance to prevent current localization via the hot-spot [2]. Subsequently, this effect leads to silicon melting, thus short is created finally. This way, the ESD device is not able to reach requested ESD level, even though size of the ESD device is enough to sustain the ESD shock.

To overcome such issue, it is necessary to increase ballast resistance by layout modifications of ggNMOS in such way that the drain side is extended and silicide process step is blocked on the drain diffusion area. This solution improves ESD performance perfectly, but affects electrical behaviour. It increases series drain resistance and parasitic capacitance. Hence, it has critical impact on RF signal if such ggNMOS is applied to protect IO circuits of the RF pad against ESD events.

Therefore, the ggNMOS is not optimal ESD protection for RF pads and has to be substituted by alternative low-parasitic ESD protections. A good candidate can be diode or thyristor. The diode is low parasitic, but is not ESD robust if working in reverse-biased mode. Thyristor satisfies both conditions required: is low parasitic and ESD robust. Standard CMOS thyristor (SCR) has too high break-down voltage which is impossible to protect thin submicron gate oxides. Several variants of thyristor were developed and presented [2] which authors tried to decrease break-down voltage in. One of them is called low voltage trigger silicon controlled rectifier (LVTSCR), it combines SCR with NMOS transistor which triggers SCR during ESD event occurred. From author's of the paper experience, the break-down voltage of LVTSCR can be still higher than break-down voltage of the protected device.

This paper describes a possible way how to decrease break-down voltage of LVTSCR by a kind of trigger circuit driving LVTSCR. It was tested on silicon successfully.

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