Professor Marinčić's Contributions to the Development of the Department for Communication and Signal Processing

[1]V. Milošević, [2]D. Bajić, [3]V. Šenk, [4]M. Despotović, [5]V. Delić

I. Introduction

It was in bygone year of 1977 when the cooperation with Professor Aleksandar Marinčić began, with the opening the post-graduate studies of telecommunication orientation at Faculty of Technical Sciences of University of Novi Sad. The first post-graduated students were the first assistants and future professors of Department of Communication: Prof. M. Temerinac and prof. V. Milošević. Continuing his regular course of Fundaments of Telecommunication at under-graduate level, Prof. Marinčić starts to teach Digital Telecommunications at post-graduate level (1978).

By supervision of two master's thesis with Prof. Marinčić as a mentor the cooperation continues gaining new subject matters. Digital telecommunication become fundamental orientation of the Department

A number of PhD. thesis supervisions and/or evaluations followed: M. Temerinac (ETF Belgrade 1982), V.Miloševića (ETF Belgrade 1984), Zdenka Živković-Džunja (ETF Novi Sad), Mirko Smiljanić (FTN Novi Sad 1992), Aleksandra Kozarov (FTN Novi Sad 1993), Željen Trpovski (FTN Novi Sad 1998). These PhD. theses extended the research interest of the Department: modulations, wireless systems, optical communications, digital image processing, digital signal processing, etc. Critical amount of teachers capable to perform and continue independent development of the Department exceeded the turning point. From a mere subgroup at group for Electronic and Telecommunication, the Department has created a powerful and independent group for Telecommunication that attracts the best and the most of the students of the generation.

With further development of digital signal processing that became one of the most developed research areas of the Department, the change of name became inevitable, and so the new name is Department of Telecommunication and Data Processing. Besides, it was unavoidable for Institute for Power Engineering and Electronics to change name as well, so now it is Power Engineering, Electronics and Telecommunication. All these changes are proofs of Department's experience and responsibility.

In spite of inevitable fluctuations of young engineers, especially in difficult nineties, Department of telecommunication and Signal Processing at Faculty of technical Sciences in Novi Sad now includes 7 professors (two full professors, three associate professors and two assistant ones), 8 teaching assistants and 15 research assistants engaged through various ways that Serbian Ministry of Science offers.

The Department organizes under-graduate courses at Group of Telecommunication within the Faculty of Electronics, Power Engineering and Telecommunication within the FTN; postgraduate studies of the same orientation and specialist studies "Contemporary Telecommunication Technologies". Besides the teaching and scientific achievements, the department has successful cooperation with industry both from Serbia and Europe. The cooperation includes the organization of courses of modern telecommunication knowledge for updating knowledge of engineers in industry, communication systems development, communication software development, ASR and TTS technology applications in Serbian, consulting and engineering within the field of communication systems, etc

In addition to many professors that have helped the Department to gain its nowadays responsibility (Prof. P. Pravica, Prof. D. Drajić, Prof. A. Marković, Prof. I. Stojanović, dr M. Ljikar, dr M. Topalović) the most significant place is reserved for Prof. Marinčić.

The most important research interest of the Department's current employees lies within the fields

[1]V. Milošević is with Faculty of Technical Sciences, Trg Dositeja Obradovića 6, 21000 Novi Sad, Serbia and Montenegro, e-mail: tlk_milos@uns.ns.ac.yu

[2]Dragana Bajić is with Faculty of Technical Sciences, Trg Dositeja Obradovića 6, 21000 Novi Sad, Serbia and Montenegro, e-mail: lmcdra@eunet.yu

[3]Vojin Šenk is with Faculty of Technical Sciences, Trg Dositeja Obradovića 6, 21000 Novi Sad, Serbia and Montenegro, e-mail: vojinsek@eunet.yu

[4]Miroslav Despotović is with Faculty of Technical Sciences, Trg Dositeja Obradovića 6, 21000 Novi Sad, Serbia and Montenegro, e-mail: ram_miroslav@uns.ns.ac.yu

[5]Vlado Delić is with Faculty of Technical Sciences, Trg Dositeja Obradovića 6, 21000 Novi Sad, Serbia and Montenegro, e-mail: tlk_delic@uns.ns.ac.yu

- Digital communication;
- Information theory and coding;
- Stochastic processes and sequences;
- Optical communication;
- Digital speech processing;
- Digital immage processing;
- Biomedical data processing.

Some of this areas are presented in the following sectaion of this paper.

II. Channel Coding

Channel coding (error control coding) has been in the focus of attention of the Department since the beginning of nineties. Block and trellis coding were both analyzed. All three aspects of channel coding were tackled: bounds, code construction and decoding algorithms. Channels considered varied from simple, such as BSC, to highly specialized, such as the magnetic recording and optical polarization channel.

Regarding the theoretical aspect of channel coding, we analyzed the bounds on the probability of error. These bounds are motivated by the geometrical insight into the error-generating process, and are uniformly better that the usual sphere packing ones. Tighter bounds than any known beforehand were given for any particular low-dimensional code, block and trellis, both upper and lower. The asymptotical behavior of those bounds was examined using their normalized logarithmic derivative, the error exponent, and several important conclusions were made. First, the upper and lower bounds are asymptotically tight, meaning that they coincide for any particular family of codes. Secondly, trellis codes need not have infinite memory to attain the maximum possible performance. Third, if decoding complexity is introduced into the refined definition of error exponent, rating of different codes changes towards ones that are proven in real life, and new criteria for constructing good codes emerge [8].

The criteria derived for evaluating codes were then used to construct codes that minimize error probability for fixed delay and decoding effort (both time and space, i.e. the number of instructions and memory occupied). A number of codes were constructed for different channels [7].

One of those constructions emerged when a possibility to apply 3n-dimensional spherical codes to polarization shift keying (POLSK) systems was investigated under the guidance of Prof. Marinčić [12]. POLSK is a form of digital optical modulation. It exploits vector characteristics of the lightwave carrier. Information is contained within the state of polarization (SOP) of a fully polarized lightwave. Since SOP provides three degrees of freedom, POLSK is especially suited for nonbinary transmission. Demodulation and detection are based on Stokes parameters extraction at the receiver. A potential improvement in POLSK systems performance, resulting from increased dimensionality of signal space, were investigated through simulation. Signal point constellations were optimized in a way that maximizes the minimum squared Euclidean distance, while preserving the constant power signaling. The gain in system performance was achieved at the expense of an increased complexity at the receiver. In order to lower the decoder complexity, signal point constellations with perfect symmetry were extracted from appropriate 3N-dimensional lattices.

Finally, we analyzed decoding algorithms for channel codes. A number of new algorithms were devised, including a bi-directional systolic-type trellis search algorithm whose complexity is several orders of magnitude lower than any other known algorithm [6, 9, 10 and 11].

III. Stochastic Processes and Sequences

Among almost infinite varieties of random processes, the attention of the Department has recently been focused to the ones that arise from so-called “sliding-window search”. This process has been attracting attention of both mathematicians and engineers. The first ones are impressed by their beautiful structure, including a lot of “ε-vicinities” and “sufficiently small δs” into their analysis. The second ones simply try to optimise certain engineering solutions, trying to avoid abstract mathematical tools.

The sliding-window search means that a semi infinite sequence of L-ary symbols is observed through a window of size N, i.e. only N symbols can be seen during a single observation (test). Tests are performed one at a time, starting (obviously) from the first one, k=1. Search is finished if certain conditions are fulfilled.

The simplest search is performed within the sequence of random equiprobable data. The simplest finishing case is if a single fixed N-symbol pattern is found. These are the only cases (restricted to L=2!) covered by mathematicians and these are the cases for which an engineer, Tolstrup Nielsen, has introduced a bifix (the name proposed by the famous professor James Massey).

Bifix is a subsequence that is in both the prefix and the suffix of a particular sequence, e.g. 8-bit binary sequence 01001001 (N=8, L=2) has two bifices, 2-bit bifix 01 and 5-bit bifix 01001. Bifix indicator $h^{(n)}$, n=0,..., N indicates existence of a bifix of length n (in the mentioned cases, $h^{(2)} = h^{(5)} = 1$). The others equal to zero while, by default, $h^{(0)} = h^{(N)} = 1$. The utilisation of bifix enabled statistical parameters evaluation (p.d.f, i.e. probability that the k^{th} test would succeeded - $\Pr\{k\}$, and higher moments, i.e. expected value $E\{k\}$ and variance $E\{k^2\}-E\{k\}^2$) [13].

If the search is performed within a set of correlated symbols, and/or if the symbols are not equiprobable, the process is more complicated. Further derivation is if a known N-symbol sequence (known as frame-alignment word - FAW, sync. word, sync. marker, framing word) is inserted periodically within the stream of almost always scrambled (and therefore pseudo-random and equiprobable) data [14]. FAW analysis concerning the search process is an attractive problem itself, also within the scope of interest of the Department [15-19].

However, the search finishing when a single particular sequence is found is no more in accordance with contemporary applications. A process that ends if one pattern among a set of M patterns is found is more appropriate. For the sake of this analysis, a cross-bifix is introduced [20-22]. A cross-bifix of length n exists if an n-symbol suffix of the pattern P_i equals to an n-symbol prefix of the pattern P_j. In this case cross-bifix indicator $h_{ij}^{(n)}$ is equal to 1. The binary patterns P_i=0**001** and P_j=**001**1 have a 3-bit cross-bifix $h_{ij}^{(3)}=1$, while $h_{ji}^{(3)}=0$. If i=j, $h_{ii}^{(n)}$ denotes the classical bifix indicator $h^{(n)}$. The default values for the cross-bifix indicators are $h_{ij}^{(0)}=1,\quad h_{ij}^{(N)}=\begin{cases}0, i\neq j\\ 1, i=j\end{cases},\quad i,j=1,\cdots,M$.

Finally, related to this process is a trellis search – an example of this is binary process finished if the window contains more than K ones, while the remaining bits are zero. This is performed by state partition into "starting states", "ending states" and forbidden states" and applying a simple method described in [23]. This exposition is just an introduction, more analyses are to follow, among which might be the ones named "A note of impossible sequences" (what happens with p.d.f. if a hypothetical set of patterns with impossible cross-bifices); "Haeberle curves revisited" (re-derivation of famous Haeberle curves if errors within a sync. word are allowed, or if distributed sequences, with or without errors, are to be used); "Distributed sequence or distributed check" – is it better to use distributed sequence, or to apply distributed check of a contiguous sequence.

IV. DATA RECORDING

Variety of communication channels were under scope of interest at the Department for communications and signal processing. One of them is data storage channel. Digital recording systems transport information from one time to another. In communication society jargon, it is said that recording and reading information back from a medium is equivalent to sending it through a time channel. There are differences between such channels. Namely, in communication systems, the goal is a user error rate of 10^{-5} or 10^{-6}. Storage systems, however, often require error rates of 10^{-2} or better. On the other hand, the common goal is to send the greatest possible amount of information through the channel used. For storage systems, this is tantamount to increasing recording density, keeping the amount redundancy as low as possible, i.e. keeping the bit rate per recorded pulse as high as possible. The perpetual push for higher bit rates and higher storage densities spurs a steady increment of the amplitude distortion of many types of transmission and storage channels.

In traditional magnetic disk drive systems, the data are recorded in tracks as a sequence of small magnetic domains with two senses of magnetization. Increasing areal density of the data stored on a disk can be achieved by reducing the length of magnetic domains along tracks (increasing linear density) and/or by reducing the track pitch (increasing track, or radial, density). Most prior research in disk drive systems has focused on increasing linear density.

Extremely high densities, viz. 10 Gbits/in^2, and data rates approaching 1 Gbits/s have been already demonstrated in commercially available systems. However, the rate of linear density increase in the future is not likely to be as high as in the past. This is because, as linear density increases, the magnetic domains on the disk surface become smaller and increasingly thermally unstable. The so-called super-para-magnetic effect fundamentally limits recording density.

Given that current linear densities are approaching the super-paramagnetic limit, the obvious alternative to an increase in linear density is an increase in radial density. Such approaches are required to meet constant the demand for increases in data rate and capacity of storage devices. In order to further increase radial density, multiple-head arrays have been developed enabling reading and writing data simultaneously on multiple tracks. Such heads can potentially provide both high density and high speed, but they suffer from inter-track interference (ITI). This ITI is a result of a signal induced in the heads due to the superposition of magnetic transitions in neighboring tracks. Today's recording systems have large track pitch, and therefore ITI has negligible effect on performance. However, significant advances in coding and signal processing for multi-track recording channels are required before the potential large radial densities together with head arrays may be realized.

The partial response (PR) systems are widely adopted as an appropriate model to describe the high-density magnetic recording process [24-27]. We focus to the class of two-dimensional (2-D) binary interfering PR magnetic recording channels. The case of two-track recording two-head reading channel, modeling intertrack interference in disc radial direction, is considered.

Soljanin and Georghiades have shown that multiple-head systems can better combat intertrack interference, which seriously degrades the error-rate performance of single-head detector, in high-density magnetic recording systems. In addition, for writing and reading on multiple tracks simultaneously the required redundancy for timing and gain control can be reduced. Furthermore, multiple-head systems have shown to be more robust to the head misalignment errors.

Iterative decoding and its application in magnetic recording have been studied extensively in recent years. Although the excellent error correction potential of LDPC codes has been demonstrated, many of unique features of these codes are still not fully understood, and some important aspects of their implementation in PR channels have not been addressed. In particular, the application of LDPC codes and iterative decoding methods to the multi-track/PR setting is an open problem.

Recent research at the Department [28,29] investigates the possible benefits of LDPC code implementation over binary 2-D magnetic recording channel, equalized to the E^2PR4 response. LDPC code iterative decoding implementation over 2-D AWGN E^2PR4 interfering channel, demonstrated very promising coding gain results, when 2-D detector was used.

For wide range of ITI values, two-head detector have shown robustness, obtaining 2dB coding gain at BER=10^{-5}.

Straightforward implementation of LDPC code does not possess ability to decrease the number of two-dimensional channel trellis states, so decoding complexity is high requiring further improvements to be closer to practical implementation.

V. Speech Processing

During the last fifteen years Department for Communications has extended their research into signal processing area and accordingly changed its name. Subjects on digital processing of speech, image and medical signals are covered. The best results are achieved in Automatic Speech Recognition (ASR) and Text-to-Speech (TTS) for Serbian language.

ASR has a goal to train computers to understand human speech. On the other side, TTS synthesis has to learn computers to read any text. These tasks are considered very complex due to great variability of speech signal and complex grammar of Serbian language. ASR and TTS are language dependent and R&D projects in these fields require multidisciplinary teams of experts: acoustic, linguistic, programming, mathematic, as well as signal processing. Such an R&D group at our Department, named AlfaNum, has developed ASR and TTS engines for Serbian language. Many standard speech processing techniques were used, and some new features introduced, especially in case of prosody generation for Serbian language.

Large speech databases are necessary for R&D in ASR, uttered from several hundreds speakers. Their utterances are recorded either via telephone line using CTI card or in speech studio or, sometimes in office environment with PC noise. After recording of a speech database, every record should be listened to, and correct phonetic transcription should be formed. Department has the greatest collection of speech databases in Serbian language.

AlfaNumASR recognizes phonemes in context. Phonemes and subphonemes are modeled with continuous and semi-continuous Hidden Markov Models (HMM). Sequential model is used, so it is possible to transfer only to the next HMM state. This is the case when speaking about states within one phoneme. Transfers between different phonemes and words can be arbitrarily complex. Standard Gaussian mixtures are used with diagonal covariance matrix. For state probabilities calculation two approaches are used: calculation over weighted sum of Gaussian mixtures, and *winner take all* method [30]. Second approach has proved equally good compared to the first one, while providing significant time savings. After state probability calculation in certain time instants, optimal sequence is found by Viterbi algorithm. A part of the program for training requires presence of labeled audio sequences, i.e. label file with correctly set boundaries between speech units. It is obvious that this would require huge amount of work if all done by hand. Therefore we provided option for automatic labeling (realignment), if some initial acoustic models are provided and a label file with correct phonetic transcription. This is done by building trellis which would force recognizer to run through all phonemes in label, and then let Viterbi algorithm to find the optimal path. Obtained segmentation is written into the file.

Implementing TTS based on TD-PSOLA algorithm implies previous pitch-marking of the database, that is, detecting locations within phones most suitable for centering overlapping windows and extracting frames, in case a TTS algorithm which requires pitch-synchronous frame positioning should be used. Acoustic parameters such as f_0 contour were calculated in two steps. The first step consisted of analyzing the sentence, word by word, in order to get information such as stress types and locations, part of speech classes and functions of particular words in the sentence. Grammatical information is essential for synthesis of natural-sounding prosody, since words in Serbian language can sometimes be stressed in different ways depending on their morphologic categories, and sometimes even have different meanings if stressed differently. In some cases even syntactic analysis does not help.

Since stress in Serbian language is fairly unpredictable, a dictionary-based solution was adopted. A special dictionary including information on stress configuration, part of speech class and morphologic categories for each word was created. Furthermore, since stress can vary along with inflections of the same word, and those variations are predictable only to a certain extent, it was necessary to include all word forms as separate entries in the dictionary. Several part of speech classes were identified as having regular behavior when submitted to inflection and they were entered in the dictionary in a form which occupied little space, but was sufficient for correct determination of the stress of every inflected form. In such a way the dictionary contains more than two million entries (including inflected forms).

Such a solution is not entirely error-free, since it does not include syntactic analysis, nor does it solve cases when syntactic ambiguities arise, and semantic analysis, however primitive, must be performed. It was decided to leave these two problems for later stages of the project. Another problem that occurs is that some words may not be found in the dictionary. In that case, strategies for determining the correct way of stressing must be defined. Strategies currently being developed within our projects include making analogies based on standard prefixes and suffixes and rhyming.

Using information from the dictionary, the system is able to reconstruct a particular stress configuration of a group of words which form a metrical unit. In this phase of the project, several f_0 contours are assigned to each metrical unit, depending on its position in the sentence (beginning, neutral, before comma, ending), and the resulting f_0 contour is smoothed in order to avoid audible pitch discontinuities and tilted towards the end of the sentence [33]. Such a method does not take into account syntactical information, but relies only on punctuation marks. However, results are still significantly better than in case of synthesizers with constant f_0, available in Serbian until now.

Halfphones are considered as basic units which cannot be further segmented, but it is desirable to extract segments as large as possible, in order to preserve intelligibility. According to differences between existing and required values of parameters previously defined, each speech segment which can be extracted and used for synthesis is assigned *target cost*, and according to differences at the boundaries between two segments, each pair of segments which can be concatenated is assigned *concatenation cost* [32]. Target cost is the measure of dissimilarity between existing and required prosodic features of segments. Concatenation cost is the measure of mismatch of the same features across unit boundaries. Various phoneme groups are treated in different fashion. Some types of phonemes, such as unvoiced plosives, are more suitable for segmentation than the others, and have lower concatenation costs. The degree of impairment of phones is also taken into account.

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