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EDITORIAL

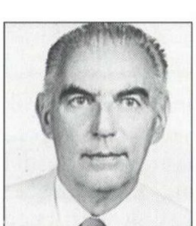
This issue of the JOURNAL ON COMMUNICATIONS is dedicated to the presentation of Hungarian results in modelling, simulation and optimization of communication links and networks.

Computer aided engineering is a significant branch of technical development. One of it's well-known areas is the computer aided design of electronic circuits. However, our main interest in this issue is not focused on electronic circuits but on communication systems, i.e. telecommunication links and complete telecommunication networks.

Simulation of communication systems is discussed in length in the IEEE Journal on Selected areas in communications, in CEI-Europe/Elsevier courses in advanced technology and on various workshops and conferences. This issue is intended to present Hungarian contributions to the simulation of communication networks. The development work took place mainly in research institutes and university departments, and the program packages have been applied in industrial enterprises and at public utilities. It is a great honour for the simulation community in Hungary to present these results, and we hope that in this software related subject, we can arouse interest in the readers of the JOURNAL ON COMMUNICATIONS. I am sorry that because of some overburdened outstanding Hungarian professionals, important subjects are missing, and I hope that in the future, a more complete review will be given.

Finally, I want to review briefly the individual papers of this issue. A. Jávor in his paper gives the general background of the simulation method. G. Csopaki, E. Halász and T. Trón present a computer program for simulation of baseband PCM transmission links. The paper of A. Benedek, I. Frigyes and B. Molnár describes a program package to simulate digital radio systems. T. Jakab, L. Jereb and M. Telek review the reliability problems of communication networks. Gy. Sallai presents a complete system of computer programs for the planning of telecommunication networks. All programs introduced in the four papers run on IBM (or IBM compatible) PC's. If you are interested in more details or in the basic science behind the programs, please contact the authors. They will be pleased to give you further informations.

Prof. K. GÉHER



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THE ROLE OF SIMULATION IN ELECTRONICS

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The role of simulation in various fields of electronics is overviewed. Reasons justifying the application of simulation in analysing and designing electronic systems, further problems of high complexity connected with extremely small or large physical sizes are dealt with.

The trends concerning the development of hardware and software tools of simulation, with special emphasis on parallel processing, object oriented programming and AI methodologies, are discussed.

1. INTRODUCTION

Simulation as a tool for designing and investigating various aspects of the World is widely used in a large number of fields. Electronics is one of the most rapidly developing area in our days in which simulation plays an outstanding role.

At first glance, one could assume that to use simulation is not justified here since the Maxwell equations determine everything in the field, and the only task that remains is to solve them for the given part of space, under appropriate boundary conditions. However, this is very rarely done because the complexity of the systems investigated is so high that in most cases this approach doesn't seem to be feasible.

In this respect I have to recall the most distinguished professor I had during my studies, professor Simonyi, quoting Michaelangelo who wrote that even the most beautiful sculpture is already hidden in a piece of marble, requiring only the removal of the unnecessary parts.

In the analysis and design of highly complex electronic systems of our time, not only the direct application of the Maxwell equations is outruled but, in a large number of practical cases, the use of simplified analytical methods too. There are, however, three questions to be dealt with, i.e.; why, how and when should simulation be used?

2. WHY?

As mentioned above, the main reason calling for simulation is the high complexity of the systems to be investigated. In numerous electronic systems, due to the large number of building elements and their interconnections, their operation is intricate to such an extent that it is extremely difficult to analyse them by conventional methods.

There is still another matter connected with complexity in our case. The keyword for this is size. A considerable number of electronic systems that have to be dealt with are either too small or too large for direct human access.

The advent of microelectronics has ruled out the conventional methods of breadboard model testing and modification because it became impossible to access and modify the thousands or hundreds of thousands of components placed on a surface of a few square millimeters. The number of elements on a single chip—in accordance with Moore's law [1]—grows exponentially. In the design and production of LSI and VLSI circuits, the expenses of prototype building are comparable with those of the whole mass production, and it is practically impossible to locate design errors by conventional measurements; the correction of even a single error would require the the manufacturing of a completely new chip. The problem of the extremely small size here has caused similar frustration as the one which Gulliver has faced; the sole difference being that now we hold a tiny chip in our hands instead of small creature.

As a result, simulation has become not only a widely used but also a badly wanted tool in CAD of electronic circuits. Simulation systems are used extensively for modelling electronic circuits on various levels such as

- procesor-memory-switch (PMS),
- register transfer
- logic
- switch
- circuit

or as a combination of them, by mixed and multimode simulation [2], [3], [4], [5], [6], [7], [8], [9], [10], [11].

The other extreme calling for the application of simulation is the case when the systems to be studied are too large. In communication networks, among them telephone exchanges and computer networks, distributed information processing and automatic control systems, a combination of highly sophisticated electronic hardware and software are utilized. The sole feasible way to get an insight into their dynamic behaviour due to their architecture and procedures controlling their operation is simulation. This approach provides also for investigating alternative structures and algorithms, and comparing them to find optimal solutions [12], [13], [14], [15], [16], [17].

3. HOW?

Electronic systems to be simulated, as it was outlined above, may be of extremely high complexity. This calls for tools capable to meet this challange. Due to the exponentional growth of complexity enabled by the development of technology [1] it became necessary to develop specialized parallel processing systems using 10^5—10^6 processors [18].

Considering the software tools applied in simulating electronic systems it can be seen that because there are many models to be simulated such as circuits, digital logic etc., the use of specialized user friendly systems has become the general practice as opposed to the general purpose simulation languages. In recent years, this user friendliness is more and more required in other fields of simulation as well.

An interesting phenomenon is that, not only in simulation but also in the discipline of programming in general, the term *object oriented* became an important catchword. This is presently a familiar way of thinking and model building, hence also for developing non-procedural simulators with objects communicating with each other, for electronic engineers [19].

Finally, it might be mentioned that the utilization of knowledge bases and methods of artificial intelligence may increase the efficiency of executing simulation experiments [11], [20].

4. WHEN?

The final question "when should simulation be used?" is a rather pragmatic one which can be answered in brief statements;

- when there is no other possibility to get the neccessary information, or
- when it is more economical to spend money on testing various designs in advance using simulation than choosing a non optimal solution, and
- always if we agree with Hamming who wrote, that "The purpose of computing is insight, not numbers" [21].

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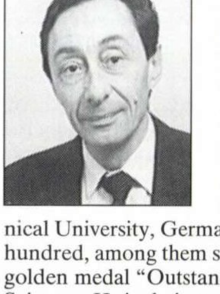

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graduated and received his M.Sc.E.E. and Ph.D. in computer science from the Budapest Technical University. He was awarded the scientific degree Candidate of Technical Science by the Hungarian Academy of Sciences. He is with the Central Research Institute for Physics, and he is also titular professor at the Budapest Technical University. He was visiting professor three times at the Aachen Technical University, Germany. His publications amount to more than one hundred, among them several books and patents. He was awarded the golden medal "Outstanding inventor" by the Hungarian Academy of Sciences. He is chairman of IMACS/Hungary, member of the editorial boards of two international simulation journals. He has been chairman and member of numerous scientific committees. His main field of interest is the methodology of discrete simulation and Artifical Intelligence (AI) in simulation, as well as the development of simulation systems.

SIMULATION OF PCM TRANSMISSION SYSTEMS

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The paper presents the main features of the computer program DLSIM developed for simulation of digital communication systems, especially of baseband PCM transmission links, on IBM-compatible PC's. The simulated system may consist of a large set of functional blocks, stored in a library, in free topology. The system description and its modification are based on a user-friendly menu-driven interaction. The program simulates the signal and noise transmission, and evaluates the system performance.

1. INTRODUCTION

Performance evaluation and tradeoff analysis are central issues in the design of communication systems. Unfortunately, except for some ideal and oversimplified cases, it is extremely difficult to evaluate the performance of complex systems by using analytical methods alone. Therefore, simulation techniques play an important role, and provide a useful and effective tool in computer aided analysis and design of communication systems.

In the past ten years, computer hardware and software technologies have undergone significant changes. Since the application of personal computers (PC) and workstations (WS) has become more widly, considerable efforts have been directed towards developing intelligent and user-friendly simulation packages that take advantage of PC's or WS's. PC versions of world famous programs (SYSTID [1], TOPSIM [2]) as well as new PC or WS based packages (MODEM [3], BOSS [4]) have been developed. The importance of this subject is manifested by the special issues of IEEE Journal SAC [5].

The main objective of this paper is to present the software package DLSIM (*D*igital *L*ine *SIM*ulation) [6,7,8] that was developed at the Technical University of Budapest for modeling, analysis, and design of digital communication systems, especially baseband PCM transmission links. The program can be used to model, test, and qualify the system in any stage of the development. This is particularly important in the early developmental stages when experimental investigations are nearly impossible or extremely expensive. The program can effectively be used in teaching by the demonstration of system operation and performance.

In the following, the main features of the program are first outlined, and the three basic modules of the program are discussed in detail. Finally, an illustrating example is given.

2. MAIN FEATURES OF THE PROGRAM

A simplified block diagram of the system to be simulated by DLSIM is shown in Fig. 1. The input bit stream is encoded to a multi-level (M-ary) symbol stream controlling the impulse source. The transmitter filter adapts the impulse shape to the channel characteristics. The transmission channel significantly distorts the signal so that consecutive symbols become overlapped and noise is added to the signal. In the receiver, the distorted symbols are reshaped to minimize the overlapping caused by intersymbol interference (ISI). The digital information is then recovered through sampling, decision and decoding. Because of the distorted and overlapped symbols, the decision is essentially influenced by the appropriate choice of the sampling time instant.

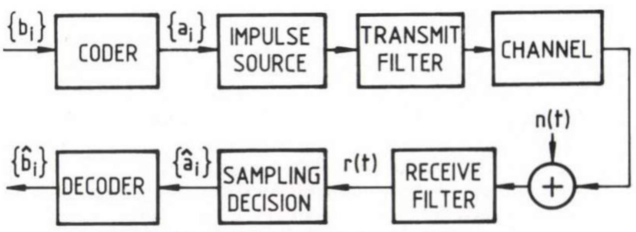

Fig. 1. Simplified block diagram of the system

The functional structure of the simulation package is illustrated in Fig. 2. It consists of three overlayed main parts using temporary files for data transfer. The program is supported by several libraries.

The *input module* deals with the system description. In addition to the basic function (the system definition), any modification of a previously described system can be carried out. Since this module represents the closest contact with the user, considerable effort was made to ensure user-friendly interaction through an appropriate menu system.

The *simulation module* determines the signal received at the input of the decision circuit. The present version of the program can be applied for linear systems only, hence signal and noise can be handled separately. Moreover, it is sufficient to merely consider the transmission of the elementary symbols, and then use superposition. By applying the monitor block, it is possible to continuously follow the signal transmission. Because of the linearity, frequency domain simulation is used.

Based on the simulation results, all informations characterizing the system are determined by the *evaluation module* which generates the eye-pattern for visual qualification of the transmission, and estimates the symbol error rate which is the most important measure of system performance. Graphical output is preferred whenever possible. In addition to the screen display, printed hardcopies are also available.

3. INPUT MODULE

As already mentioned, the input module is applied for the description of the communication system. It works by nested menu technique. The main menu, as shown in Fig. 3a, allows the user

(i) to define the system to be simulated, i.e., to specify all its building blocks,
(ii) to delete, insert, exchange or modify any building block,
(iii) to display or print the system description,
(iv) to start the simulation or to stop the program operation.

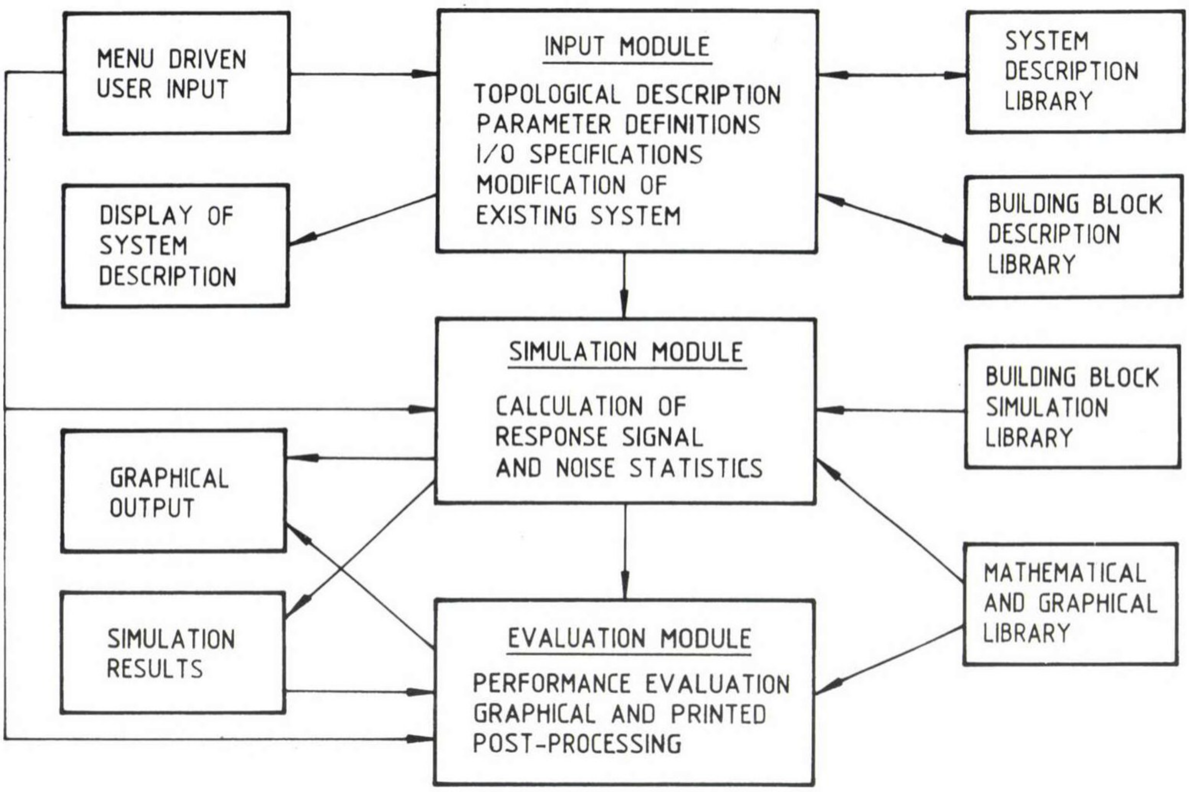

Fig. 2. Functional structure of DLSIM

The specified system descriptions are automatically stored in a library file for later use (simulation, modification or documentation).

To build up the system (*create* option), the building blocks are grouped according to Fig. 3b. Having selected a group, the corresponding submenu lists the set of optional group entries. For instance, a bit generator can be programmable or random, the impulse source may generate square, trapesoidal or half-sine pulses, AMI, HDB3, 4B3T or general block-codes can be selected, filters may be Butterworth, Chebyshev, Thomson as well as lowpass, highpass, bandpass or allpass, etc. Among the building blocks, there are special ones applied in individual systems such as Wandel & Goltermann type cables and special line built-out or equalizer circuits. Some blocks are defined by circuit diagrams (ladder filters, bridged-T equalizers), others either by measured magnitude and phase/group delay characteristics or their transfer function (given by pole-zero pattern or polynomial coefficients). In any case, the program asks for the necessary block parameters to specify the chosen building block.

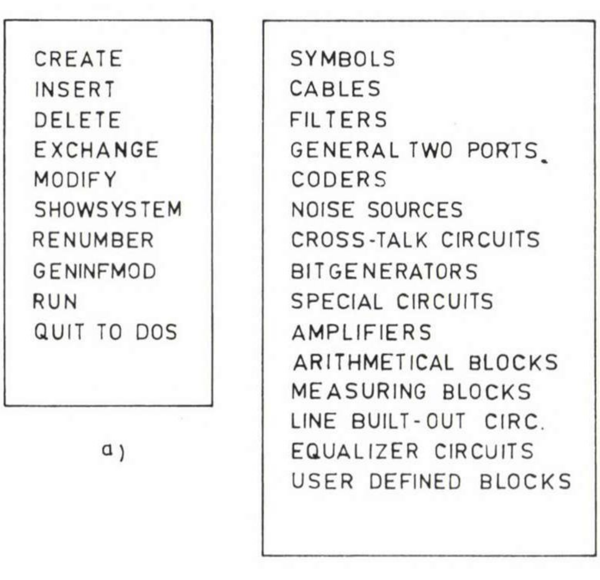

Fig. 3. Menus: (a) main menu, (b) building block groups

General system data (system name, identifying comment, bit rate, sample per symbol, etc.) can be modified by the *geninfmod* option. To check and/or correct any building block data, the *modify* option is used. Reference to a building block is possible by the name identifying the block, or by the serial number of the node preceding or following the block. The *renumber* option has to be used after *insert*ing or *delete*ing a block. Finally, the *run* option starts the simulation.

4. SIMULATION MODULE

Because of signal distortion, the transmission must be considered in continuous time. This module simulates the analog part of the digital system, i.e., the section between the encoder and the decision circuit. In the present state of the program, the system is assumed to be linear, thus it is enough to simulate the transmission of a single symbol in order to determine the impulse response at the input of the decision circuit. The response to an arbitrary stream of symbols can then be calculated by using the superposition theorem.

The actual simulation is carried out in the frequency domain where the overall transmission can be derived by simply multiplying the transfer functions of the building blocks, instead of using the corresponding convolution in the time domain. Whenever needed, the program calls for the Fast Fourier Transform (FFT) routine of up to 1024 points for transformation between the two domains. Simulation of a building block requires a call for the corresponding rout.ne which calculates the block transfer function and then the output spectrum.

In the simulation, it is possible to follow the signals at any internal point of the system by means of the monitor block which realizes a general measuring equipment. At any node where this block has been inserted, it is possible to display either the signal waveform *(oscilloscope mode)* or its spectrum and a partial transfer characteristic to this node *(spectrum analyser mode),* either from the input or from an arbitrary source node defined previously *(generator mode).* Shifting, windowing and rescaling actions are available on the plots. In addition to the display, printed hardcopies can also be generated. This block is especially powerful in teaching because it is easy to demonstrate either how any single building block effects the signal shape and spectrum, or what transient response and transfer characteristics a building block has.

The program can simulate additive noise as well. Noise sources may be either white noise or cross-talk from a neighbouring channel. Independent noise sources consisting of several building blocks and joining the signal path through an adder are allowed. The statistical noise simulation, separated from the signal, is also carried out in the frequency domain and yields the noise power spectral density (PSD) at the input of the decision circuit. The noise magnitude distribution is assumed to be Gaussian with a standard deviation of

$$\sigma^2 = \frac{2}{f_{eq}} \int_0^\infty N(f)\,\mathrm{d}f$$

after substituting the output noise, having a PSD of *N(f),* by an equivalent bandlimited white noise of bandwidth f_{eq}.

5. EVALUATION MODULE

The evaluation (or post-processing) module displays the results of the simulation and evaluates the system performance. First, the noise power spectral density and the impulse response signal are plotted in the simulated fre— quency and time window, resp., which are mutually related according to the FFT. Then, applying either a randomly generated or a user defined input bit stream $\{b_i\}$, the received signal can be determined by the equation

$$r(t) = \sum_{i=-n}^{m} a_i\, g(t-iT) + n(t)$$

where *g(t)* is the simulated impulse response, T is the symbol repetition time, $\{a_i\}$ is the encoded M-ary symbol stream, n and m are the number of overlapping symbols, and *n(t)* is the additive noise. The noiseless response is also displayed, together with the symbol series. Then, the module generates the eye-pattern, determines its characterizing data (opening, width, optimal sampling time) and evaluates the system qualifying quantities (ISI, signal-to-noise ratio [SNR], noise margin, worst-case symbol error rate [SER]). All these data are listed on the screen, together with the eye-pattern. The generation of the eye-pattern lines is controlled by the user.

Finally, the main measure of the system performance, the SER is estimated. Assume that $\{a_i\}$ is a symmetrical equidistant M-ary symbol set with uniform probability. In the case of Gaussian noise, the SER is given by

$$P_e = 2\left(1-\frac{1}{\mathrm{M}}\right) \int Q\left(\frac{d-y}{\sigma}\right) f(y)\,\mathrm{d}\,y$$

where $Q(.)$ is the error function of the normalized Gaussian distribution, and $f(y)$ is the probability density of the ISI, practically always unknown. To overcome this difficulty, the iterative semianalytic Gaussian quadrature rule (GQR) is applied [9], yielding

$$P_e = 2\left(1-\frac{1}{\mathrm{M}}\right) \sum_{i=1}^{N} w_i\, Q\left(\frac{d-x_i}{\sigma}\right)$$

where x_i and w_i are the points and weights of the N-th order GQR. In the case of block coding, the correlated ISI moments are used [10]. The evaluated SER vs SNR curve is plotted displaying the actual values. Deterministic changes in the sampling time instant and in the decision level can also be taken into consideration.

6. EXAMPLE

To illustrate the capability of the program, a 140 Mbps coaxial PCM transmission system was simulated. In Fig 4, the elementary symbol carrying the encoded information is plotted at different nodes of the system. The figure shows the original source symbol, in addition to the distorted and equalized ones. The signal shape is considerabely distorted by the cable. Fig 4b implies the necessity, and the form of the output signal (Fig 4c) proves the effectivity of the equalization. The amplitude and phase characteristics of the equalized system are shown in Fig. 5. and Fig. 6., respectively. Fig 7 shows the eye-pattern with 4B3T (FOMOT) encoding. In Fig 8, the result of the iterative SER estimation is presented.

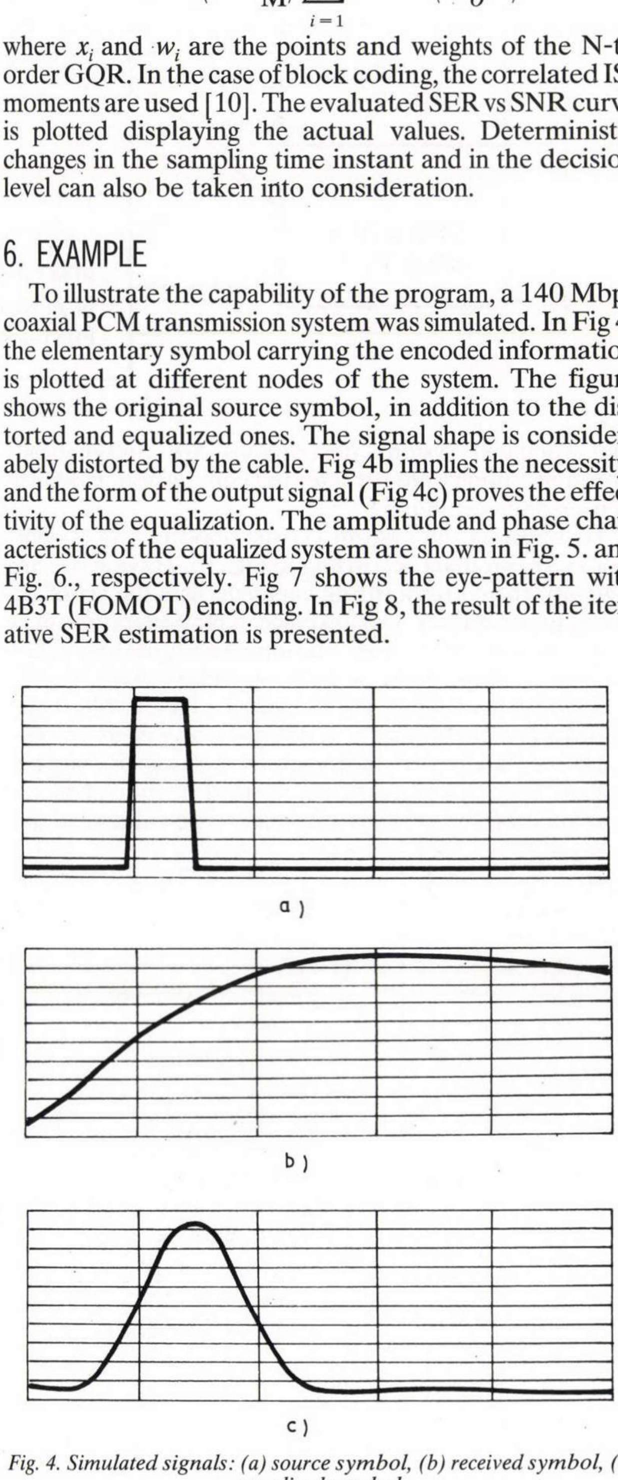

Fig. 4. Simulated signals: (a) source symbol, (b) received symbol, (c) equalized symbol

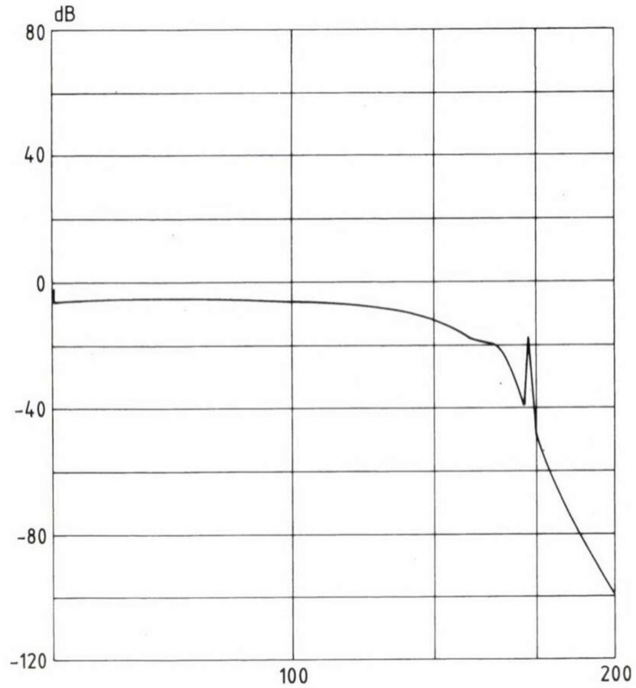

Fig. 5. Amplitude characteristic

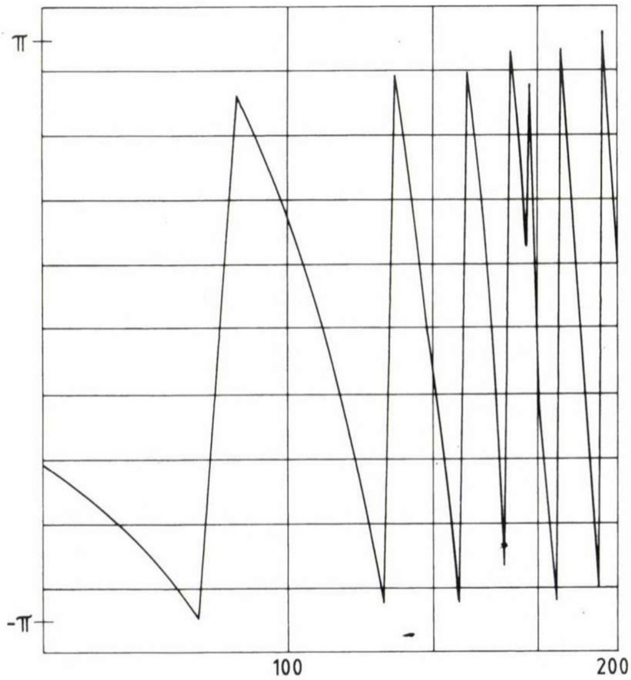

Fig. 6. Phase characteristic

7. CONCLUSION

In the paper, the program package DLSIM, developed for simulation of digital communication systems on IBM PC's was presented. The program is suitable to model, test and qualify the system already in the design phase. Several design variants may be compared. Furthermore, it is easy to model any modification of the system elements, and the effect of these modifications is simple to analyse. All these capabilities provide a simulation program which is a very efficient tool for the system designer.

During the program development, the possible use of the program in education was considered. Considerable efforts have been made for improving the demonstration facilities and making the simulation results more visible by using graphic output and printed hardcopies. According to our experience, the program DLSIM has turned out to be a very effective tool in teaching as well.

In further development, we intend to extend the program for simulating modulated and optical communication systems. To this end, nonlinearities have to be included and time-domain simulation has to be applied, furthermore signal and noise should simultaneously be handled.

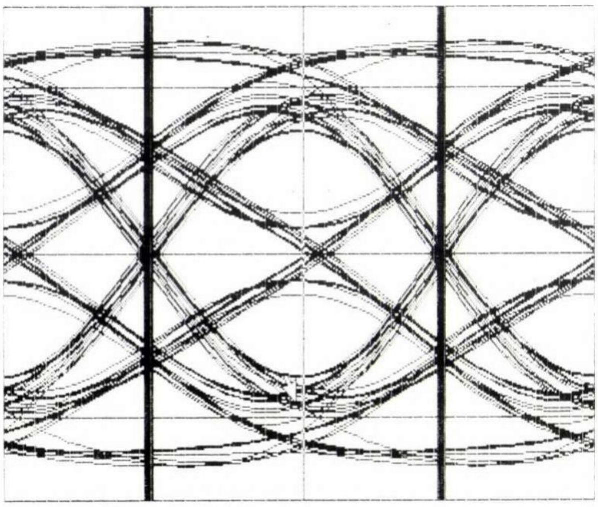

Fig. 7. Eye-pattern

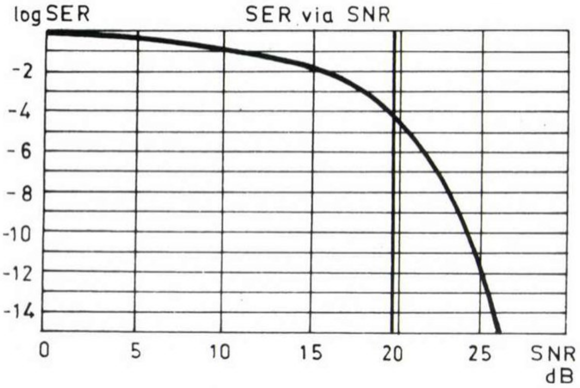

Fig. 8. Symbol error rate versus signal-to-noise ratio

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COMPUTER SIMULATION OF COMMUNICATION LINKS: SOME GENERAL PROBLEMS AND THE ASTRAS PROGRAM PACKAGE

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A communication system simulating program package ASTRAS is described. After a brief discussion of some general problems in system simulation, the structure and features of ASTRAS are given. Methods applied to estimate error probability are briefly discussed while a new method, effective in the case of very high linear distortions, is described to some detail.

1. INTRODUCTION

Computer simulation is an extremely effective tool in the analysis, design and evaluation of communication systems. This fact has been recognized two decades ago or so; since then, many simulation program packages have been developed and applied with success. Important aspects in this art are published in the literature, some of the early ones being [1, 2]; more recently, among others, two issues of IEEE SAC [3, 4] have been devoted to this problem.

A program package ASTRAS (Analog Simulation of TRAnsmission Systems) has been developed having the aim to simulate digital communication systems with the main concern to digital radio. The name (*Analog*...) is justified by the self-evident fact that factors influencing the performance of a *digital* communication system can more easily be described by *analog* than by digital characteristics. The present paper gives a description of ASTRAS after a short tutorial material.

In Section 2, some general points of communication system simulation are dealt with. Structure and main features of ASTRAS are given in Section 3. Error probability estimation methods, as applied in ASTRAS, are described in Section 4. And the particular problem of the high-distortion—noisless situation is briefly discussed in Section 5.

2. SYSTEM SIMULATION OVERVIEW

2.1. Networks and systems

Communication systems are complex structures of many components, which have complex characters themselves. Due to the high complexity, an analytical treatment of their behaviour would be a formidable task; experimental investigation, on the other hand, is extremely expensive, particularly in early stages of development. Computer simulation is therefore the preferable means of investigation/design. Further, simulation is a very effective tutorial means in explaining of system concepts and properties.

Computer methods of system simulation differ substantially from methods of network simulation/analysis. Of course, systems are built of the same electronic components as circuits. The differences are, however, essential, as summarized in Table 1. Due to the untractable characteristics in the central column of Table 1, an abstraction as characterized in column 3 is needed.

Table 1
Comparison of networks and systems

	Networks	Systems as networks	"Abstract" networks
Components	Simple	Simple	Complex
Number of components	High	Very high (∞)	Low to medium
Signals	Simple	Complex	Complex

2.2. Block structures

Figs. 1 and 2 show typical systems. These and other ones contain several blocks, further sources of useful signals and of noise, interferers, etc. These are arranged in series and parallel branches forming the structure to be simulated. The complex components of such a system are called *blocks*, having one or a few input ports and one output port. Blocks are defined by their input-output characteristics.

2.3. Signal representation

Signals characterized in Table 1 as *complex* contain random binary or M-ary sequences, Gaussian or more general noise, groups of sinusoids, etc. These appear either as baseband signals or as modulations of the appropriate carrier. Bandpass signals (apart from those of extremely wide relative bandwidths) can be represented by their complex envelopes. Further, either time domain or frequency domain representation is possible, the former being more appropriate in some cases while the latter in other ones. FFT or IFFT serves for the transformation from one of these to the other one and vice versa.

2.4. What to simulate?

Parameters to be determined by simulation may contain virtually any performance parameter such as signal shape, spectral density, signal-to-noise ratio, error probability; in digital transmission, error probability and eye pattern are the most important ones.

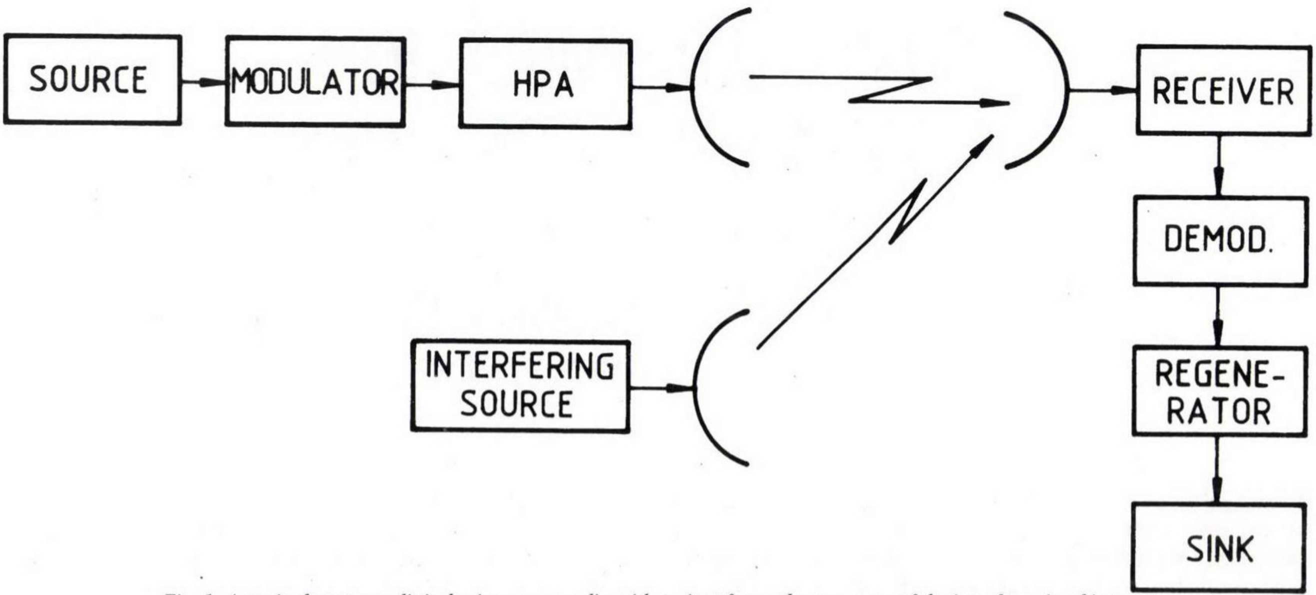

Fig. 1. A typical system: digital microwave radio with an interferer; the presence of the interferer is of interest

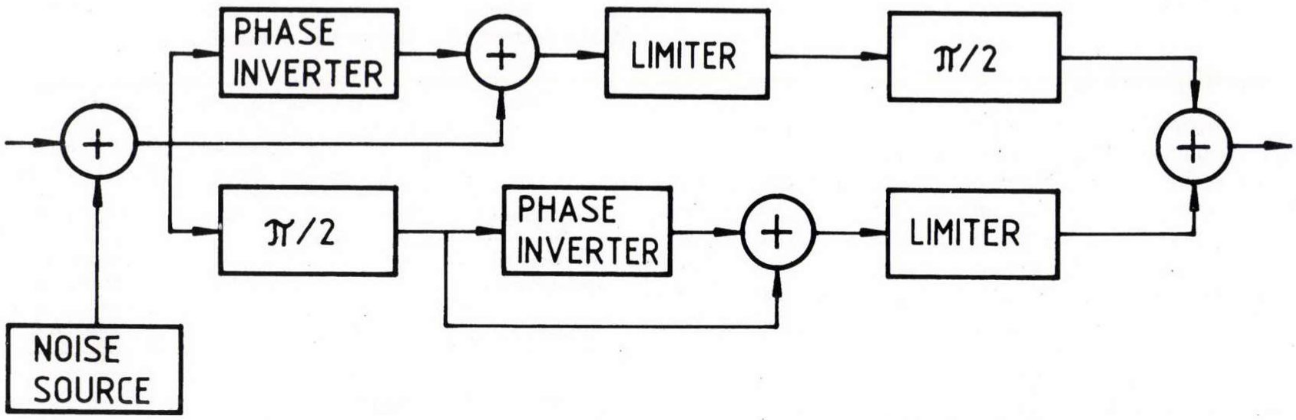

Fig. 2. A typical system: a QPSK direct phase regenerator; the presence of many parallel branches is of interest

The estimation of low error probability is one of the crucial points of simulation. Problems are discussed in detail in [5] and elsewhere.

3. THE ASTRAS PROGRAM PACKAGE

Taking the concepts of Section 2 into account, ASTRAS has been developed to fulfill the following tasks:

- to define the structure to be simulated;
- to carry the signals generated by the main source through the structure, taking the transfer functions of the blocks as well as signals of disturbing sources into account;
- to determine the appropriate measure of the transmission performance.

The main features of ASTRAS are:

- systems with virtually unrestricted complexity can be dealt with;
- it has a completely interactive character;
- program run can be interrupted and continued later, block data can be modified, etc.

Generality of the simulation is restricted in two points only: no feedback can be dealt with and only memoryless nonlinearities can be treated (although the latters may have AM-to-PM conversion).

These restrictions are justified by considering that microwave radios themselves usually do not contain feedback; the systems thus excluded are of marginally interest only. On the other hand, subassamblies containing feedback are better analyzed via network analysis programs; from the simulation point of view, these are regarded as blocks with known transfer functions. Nonlinearities having a memory can usually be modelled as a cascade of a memoryless nonlinearity and one or two filters.

The program package is composed of three main parts (Fig. 3): QINPUT defines the system in an interactive

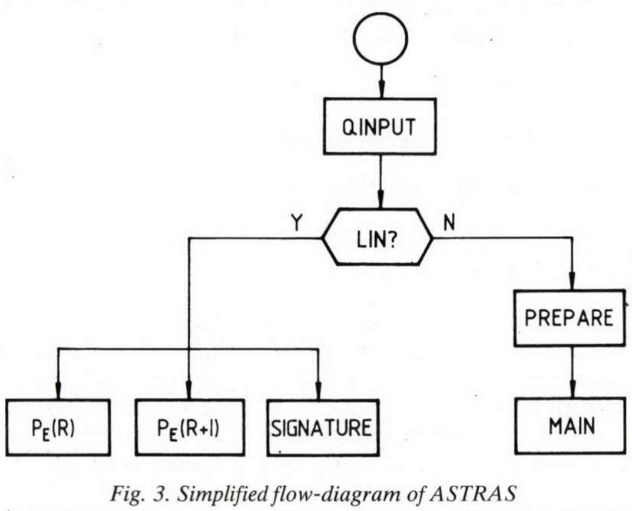

Fig. 3. Simplified flow-diagram of ASTRAS

way by presenting the structure and the block parameters on the display. A menu is assigned to each block, its type and parameters chosen by an ergonomically designed dialogue. The offered blocks contain predefined and user-defined ones. Among the predefined ones there are binary and M-ary signal sources, standard filters, modulators, amplifiers, carrier and clock synchronizers, etc. User-defined block characteristics have to be given in a file.

Simulation is executed by applying one of two possibilities: segment, called ASTRAS-QL is used if QAM modulation is applied and the system contains linear blocks only; it has three sub-segments, solving three different problems as seen in Fig. 3. In the more general case of nonlinear systems, ASTRAS-NL has to be applied. This distinction makes possible to use sophisticated and thus time efficient methods in the former case which are not applicable in the latter — as will be briefly discussed in the next Section.

Linear blocks are handled in the frequency domain and nonlinear blocks in the time domain. Linear and nonlinear blocks may follow each other in any sequence. ASTRAS-NL recognizes automatically the character of the block and executes FFT or IFFT as needed. Further, a special and efficient method is applied to deal with parallel branches.

4. THE ESTIMATION OF ERROR PROBABILITY

Under usual circumstances, error probability is influenced by additive noise, intersymbol interference and phase errors of the recovered clock and carrier; additive interferers may also be present, while carrier is irrelevant to noncoherent systems. For coherent systems,

$$P_E(R) = \int\int\int P_E(R|g, \tau, \varphi)\, p(g)\, p(\tau)\, p(\varphi)\, \mathrm{d}v_g\, \mathrm{d}\tau\, \mathrm{d}\varphi \qquad (1)$$

where R is the signal-to-noise ratio,
P_E is the total probability of error,
g, τ and φ denote the intersymbol interference vector, the timing error and the carrier phase error, respectively,
$p(\,)$ the probability density of the appropriate variable, and
dv_g the elementary volume in the space of g.

In ASTRAS, the optimal value of τ and φ are estimated and taken into account during execution; they can, however, be varied interactively. The actual calculation involves thus

$$P_E(R) = \int P_E(R|g, \tau_0, \varphi_0)\, p(g)\, \mathrm{d}v_g \qquad (2)$$

with the subscript 0 meaning the actual value.

The first term in the integrand of Eq. (2) may or may not be known, depending on the noise statistics. Determination of the density function $p(g)$ is the task of simulation. The method of solution depends on the actual situation. Various situations are shown in Fig. 4.

If the system is purely linear, noise is Gaussian and $|g|$ is low, Gauss Quadrature Rule (GQR) can be applied to evaluate Eq. (2). And, as superposition holds in this case, GQR parameters can be calculated if the response function of the system to the elementary signal is available; to get this, one IFFT has to be evaluated. This is applied in ASTRAS-QL.

The situation is more complicated if, as in Fig. 4b, some of the blocks are nonlinear, but no nonlinear transformation is applied on the Gaussian noise. In this case, a quasi-analytic (QA) method is applied where the empirical $p(g)$ is determined via Monte Carlo simulation and applied in Eq. (2). This can be determined with a rather short run of simulation.

Finally, if Gaussian noise is also transformed nonlinearly, even $P_E(R)$ is not known a-priori. In this case, ASTRAS-NL applies Monte-Carlo simulation for the noise as well.

In the first and second case, output data are $P_E(R)$ curves while in the third case, the actual value of error probability together with the limits of confidence interval are the outputs. (The confidence interval is, of course, decreasing with the number of runs).

Error probability under the influence of interferers is determined either by Monte-Carlo simulation or by assuming a Gaussian distribution for the latters.

5. ERROR PROBABILITY IN THE CASE OF HEAVY LINEAR DISTORTIONS

Under special circumstances, linear distortion may be high enough to cause errors; error probability may then be influenced by multipath propagation, this being one of the main reasons of malfunctioning both in high speed LOS (Line-Of-Sight) and in mobile digital radio. In principle, the above methods for determining error probability can be applied in this case as well — e.g. for GQR or QA in the appropriate situations.

There is, however, an essential difference between the low and the high distortion case: in the former, it is the *mode* (i.e. the maximum of the probability density function) and its neighbourhood of the (usually unknown) probability density of g which are of interest; this can be determined by relatively low computational efforts. In the latter case, it is the tail of this distribution which is of interest; to get a reliable information on this, either a very high-degree GQR or a very long run in QA is needed. Both are extremely time consuming.

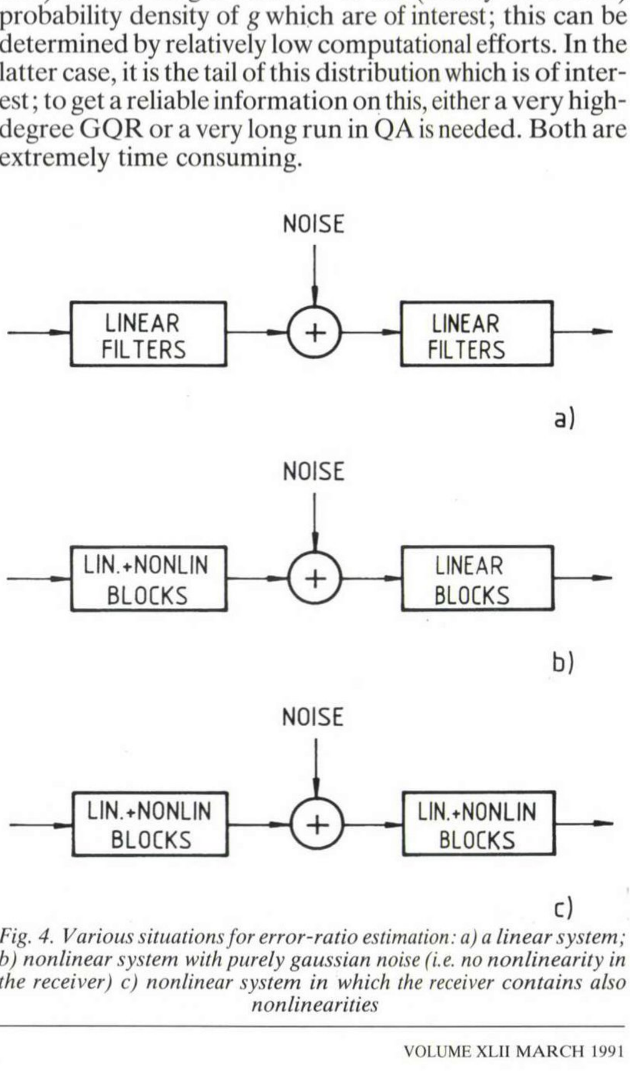

Fig. 4. Various situations for error-ratio estimation: a) a linear system; b) nonlinear system with purely gaussian noise (i.e. no nonlinearity in the receiver) c) nonlinear system in which the receiver contains also nonlinearities

A new method, applying orthogonal polynomial expansion for $p(g)$, has been developed [7]; the expansion is a generalization of the well-known Gram-Charlier method, using Hermite polynomials. (For sake of simplicity, one-dimensional situations will be dealt with in what follows;).

It is known that orthogonal polynomials can be used to expand rather general functions, probability densities falling in this class.

$$p(g) = w(g) \sum_{k=0}^{\infty} C_k P_k(g) \tag{3}$$

and

$$C_k = \int_a^b p(g)\, P_k(g)\, dg = \sum_{i=0}^{k} a_i^{(k)} m_i \tag{4}$$

where $w(g)$ is the appropriate weight,
$P_k(g)$ is the normalized polynomial of order k,
$a_i^{(k)}$ is the i-th coefficient of the k-th order polynomial,
m_i is the i-th moment of g and
a and b are the upper and lower limiting values of g.

As we are free in the choice of $w(g)$—i.e. in the choice of the actual orthogonal polynomial system—it is reasonable to take into account that we are interested in the tail-distribution; i.e. by truncating the expansion (3), approximation must be good in the tails while we are not too much interested in the mode-approximation. To find the appropriate $w(g)$ we note that a truncated orthogonal expansion minimizes the mean-square error, i.e.

$$\int_a^b \frac{1}{w(g)} \left[p(g) - w(g) \sum_{k=0}^{n} C_k P_k(g) \right]^2 dg = \text{minimum} \tag{5}$$

If the support of $w(g)$ is chosen as finite, its limits being a and b, respectively, further $w(a)=w(b)=0$, the contribution of the errors in the tails to the overall error will be enhanced. This assures better approximation at the significant domains. We can thus predict that this approximation will converge rather rapidly.

Jacobi-polynomials have the properties listed above if g is normalized appropriately. For these,

$$w(x) = (1-x)^{\alpha} \cdot (1+x)^{\beta}\,;\ \alpha, \beta > -1/2 \tag{6}$$

Here the domain $g \in (a, b)$ is transformed into $x \in (-1, +1)$. $\alpha = \beta$ seems to be a logical choice, leading to the ultraspherical Jacobi polynomials.

It turns out that the moments of g can be determined rather simply, further there exist simple recurrent formula to determine the a coefficients. As a result, error probability computation is rather fast.

The method outlined above has been applied to compute the signature of high speed digital radios. As in signature calculations a very great number of error probabilities have to be computed, the computational efficiency is a key point. (A more detailed discussion of the method is given in [6].)

6. CONCLUSIONS

A rather effective simulation program package has been developed and applied to various communication and educational tasks. Among others, the effectiveness of frequency and time domain equalizers, functioning of systems as complex as a direct phase regenerator chain, and many others have been determined and optimized. Some graphical results for various systems are shown in Figs. 5—8.

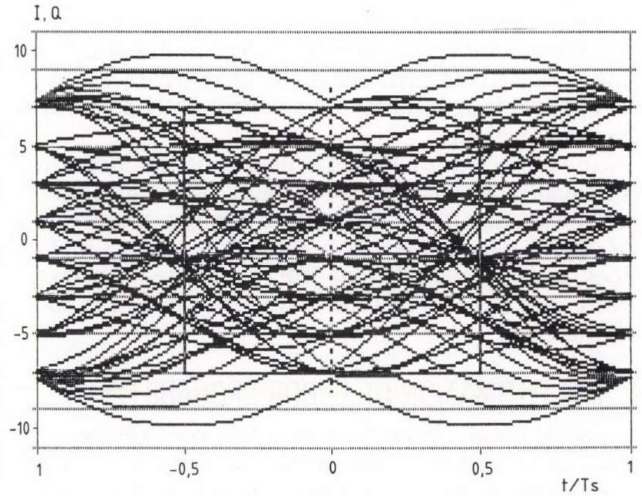

Fig. 5. Eye pattern of a 64 QAM system with a Tchebishev transmit filter and a Butterworth receive filter, as computed by ASTRAS

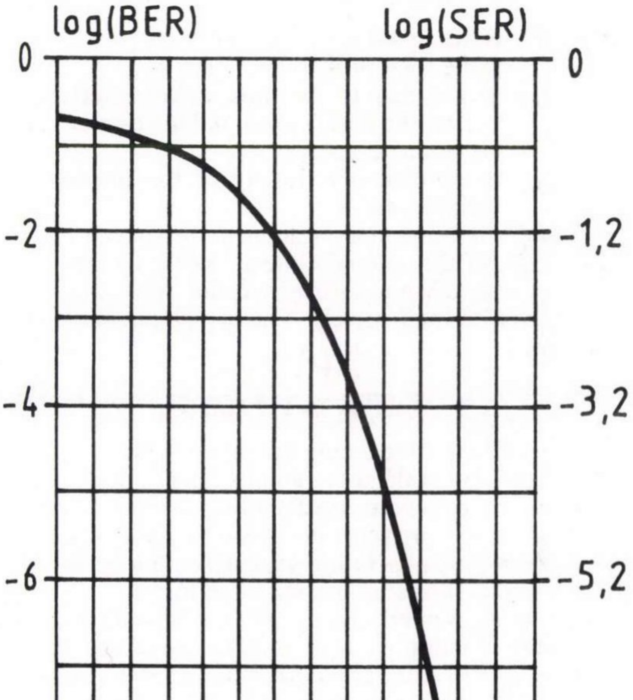

Fig. 6. P_E of a situation as in Fig. 5.

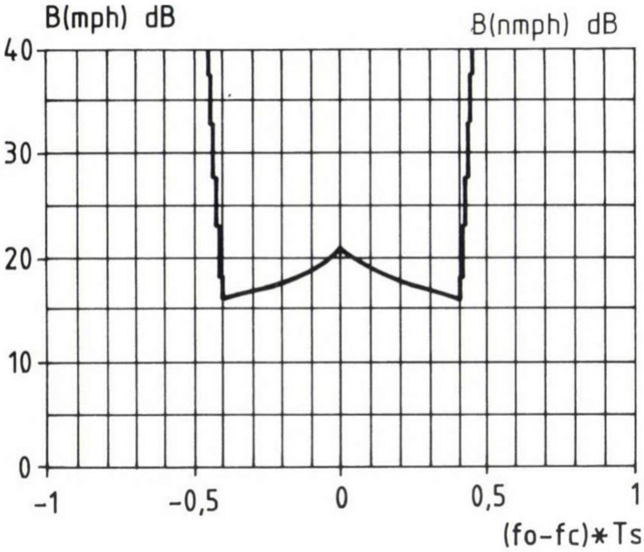

Fig. 7. Signature of a 16 QAM system with 5-tap linear transversal equalizer; (mph: minimum phase, nmph: non minimum phase)

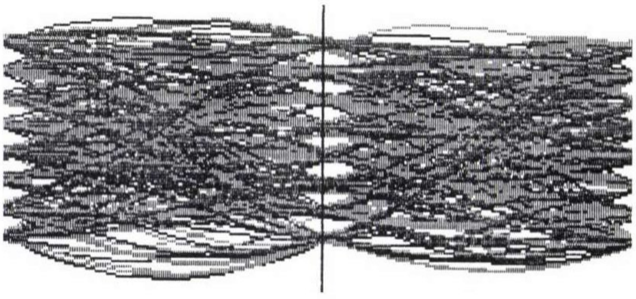

Fig. 8. Eye pattern of a 64 QAM system with a 0.5 roll off raised cosine filter and an amplifier having 0.5 °/dB AM-to-PM conversion

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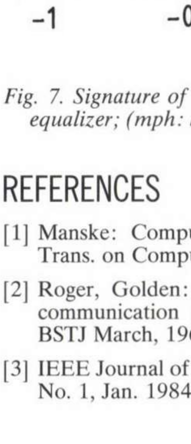

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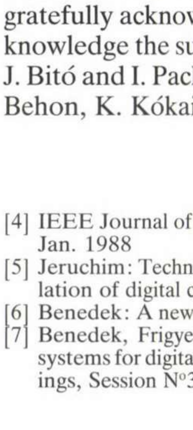

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COMPUTER AIDED RELIABILITY ANALYSIS AND PLANNING OF TELECOMMUNICATION NETWORKS

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An overview of reliability problems in communication networks is presented. In the first part, reliability constraints arising in the phase of the topological planning are summarized, and techniques used for the determination of the transmission network are introduced.

In the second part, the reliability analysis based on the topological and transmission plans is presented. By means of the proposed method, the reliability model of the network is generated by dividing the network into independent subnetworks, and by applying a state space reduction technique to obtain a model with a manageable size.

1. INTRODUCTION

As the quantity of services provided by communication networks increases the requirement for reliable operation plays a more important role. Due to this requirement, it is necessary to know the network reliability parameters and to take into account the reliability criteria in the planning stage.

The increasing importance of network reliability has resulted in a great number of publications about this topic. One of the basic similarities between the papers published about methods of determining network reliability is the decomposition of the problem into the network planning phase and the network analysis phase, and furthermore into different subphases.

The decomposition of the problem is necessary since even some of the individual steps of the network planning and analysis are NP-hard (see e. g. Spragins et al. [1], Jan et al. [2]). Due to the complexity of the individual steps, the authors usually focus their attention on the particular questions instead of dealing with the overall problem.

In this paper, we survey the problems arising in the phases of reliability planning and analysis of communication networks. The preliminaries of this overview are twofold. On the one hand, during the past 10–15 years, one of our basic research activities has been the development of software methods and tools used in the planning of topological and transmission networks (e. g. Jereb et al. [4], Sallai et al. [5]).

On the other hand, our recent work is based on the results achieved in the reliability modeling and analysis of (complex) electronic systems (e.g. Jereb [6], Begain [7]). In this paper, these results are extended and applied to the reliability analysis of the network.

The paper is organized as follows: In the next Section, the modeling assumptions are described. Then the planning method and the planning steps considering both the topological and the transmission network aspects are introduced. The network reliability analysis approach can also be found in this Section. Next, as a result of the previous Sections, a software implementation of the described method is demonstrated.

2. MODELING ASSUMPTIONS

The network can be either a metropolitan or a long-dictance trunk network which is modelled by an undirected, connected graph. The nodes of the graph represent either the switching centres or the points marking the topological places of the network between which the links can or must be routed and where no switching functions are realized. The edges of the graph denote the transmission links of the network.

At the beginning of the topological planning, the following data are given:

- The geographical location of the switching centres and the additionally defined points.
- The circuit demands between the switching centres (node pairs).
- The edges of the network which can be realized and used for the routing of circuit demands (initial graph).

Once the above assumptions are made, the aim of the topological and transmission planning is to determine the least-cost network which realizes the given circuit demands. If we disregard the possible failure of the network components, the edges of the suboptimal network and the necessary equipment can be determined (e.g. [3], [4], [5]).

On the other hand, if we are not only interested in the fault-free operation of the network but also in the quality of service in failed situations then the reliability aspect must be taken into consideration in the planning phase.

In this paper, the reliability assumptions of the network components are as follows:

- The nodes are ideal (i. e. they cannot fail).
- The reliability of the links is independent of the traffic realized by the network.
- Both the failure and the repair time distributions are exponential.
- The failure rate of the different components of the network are independent of each other. However, because of the maintenance, there are parts of the network which consist of dependent components within the independent subnetworks.

3. PLANNING OF RELIABLE NETWORKS

In the developed method, the handling of reliability aspects is different within the two main phases. In the planning phase, the random property of failures is ignored. The aim of this step is to create a network which enables the quality requirements to be fulfilled.

The topological and the transmission planning phases include the following steps:

- The determination of the network structure by considering only the connectivity of the graph (unirouting solution).
- The supplement of the unirouting graph and the routing of the circuit demands by taking into account the multirouting requirements.
- The determination of the circuit modules on the basis of the allowable groupings in the nodes of the network, and the different transmission technologies (optical, microwave, etc.).

Starting with the initial graph in the first step, a heuristic method is applied based on the results of [3] and discussed in detail in [4]. The result of this step is a suboptimal graph which is connected with regard to the switching centres. (The additionally defined points can be eliminated).

In the second step, an interactive technique is introduced. This technique creates the possibility to supplement the network in order to fulfill the multirouting (reliability) requirements. Although there are known methods which yield graphs with multiple connectivity, in the practical cases the interactive technique is available.

The application of the interactive method is based upon to the inhomogeneity of the multirouting requirements. If these requirements differ only for the different relations then the routes can be determined automatically. However, there are usually some additional planning aspects (e.g. alternate technologies of the routes) which can be taken into consideration significantly better in an interactive manner than by a method aiming at a least-cost solution. In this step, all of the demands are realized either on their minimal path or on their edge- or node-disjoint minimal multipaths. In this phase, we also disregard the capacity limits and module sizes of applied links.

The final step of the planning phase is the assignment of equipment to the links. In this step, the grouping rules in the different nodes, the module sizes of the different technologies (optical, microwave), and the improvement of the reliability are considered.

In this planning phase, the availability of relations can be significantly increased by splitting the demands into preliminary defined parts. These parts of a given demand can be routed via different equipment of a given path. However, the costs of this protected solution are the increased request for grouping and the increased amount of equipment.

The introduced planning steps diversely influence the parameters which determine the network reliability. However, the reliability and availability parameters of the network depend partly on the structure of the network, partly on the implemented equipment and also on the applied maintenance. The common impact of these aspects can only be taken into account by means of a detailed reliability analysis. The result of the analysis allows the comparison of the planning versions from the reliability point of view, and allows the revision of decisions made in the planning phases.

4. ANALYSIS OF NETWORK RELIABILITY

The reliability (availability) analysis qualifies the technical state of the network and the resources which can be utilized for the service of the users in the given state. With the above assumptions in reliability respect, the network can be described by a continuous time Markov chain and the steady state probability distribution, as well as some time parameters (mean up time – MUT, mean down time MDT) can be used in the reliability qualification.

The methods for providing the reliability parameters of a system described by a continuous time Markov chain are well known in the literature. However, in network reliability analysis, the basic problem is the size of the model. The problem is twofold. On the one hand, it is practically impossible to generate the total state space and to solve the full Markovian model. On the other hand, the solution of the complete model yields lots of useless results. For this reason, one of our basic purposes has been to develop a method which allows a reduced reliability model to be generated.

One of the solutions for reducing the state space and handling dependent failures can be found in [8]. However, in our solution, we handle these problems in a different way by reducing the Markovian state space of the independent subnetworks (containing the dependent components).

In the first step toward determining the model of the independent subnetworks, a data bank is established which contains the reliability models of different kinds of links. Using this data bank, the reliability model of the subnetworks is generated by a hierarchical method yielding the reduced state space in a step-by step manner. The reduced state space contains only the states of the highest probability. This method is described in detail in [6] and [7].

If we are interested only in the s-t (terminal) availability and if the circuit routing of the independent subnetworks does not depend on the other subnetworks then the available circuit capacity of the given s-t routes can be determined in the different states of the model. Further opportunities of state aggregations can be obtained by recognizing that there are some states in which the capacity of the given s-t routes are identical.

Similarly, another space reduction can be achieved by distinguishing only certain values of path capacities. The states which provide capacity values between two of the distinguished values are then lumped into the lower capacity state. This aggregation results in the lower bound of the availability parameters.

In the final step of network reliability modeling, the independent subnetworks which realize the paths of the given s-t demand are joined together. If the segments of the same route in the different subnetworks are independent of each other then this step requires only the calculation of probabilities of independent events. In this case, the resulting path capacities can be derived from the path capacities of the independent subnetworks. However, if they do depend on each other then the resulting path capacities can be determined only by the joint analysis of the given subnetworks.

5. THE PLANET PROGRAM PACKAGE

The outlined method is implemented by the program package PLANET on an IBM PC AT. This package includes the network planning phases TOPSYS and SYPLAN and the network reliability analysis phase RELNET. The input data structures of RELNET correspond to the output data structures of SYPLAN.

The implemented version of PLANET is illustrated by Figs. 1 to 6. Some examples of the menu driven general

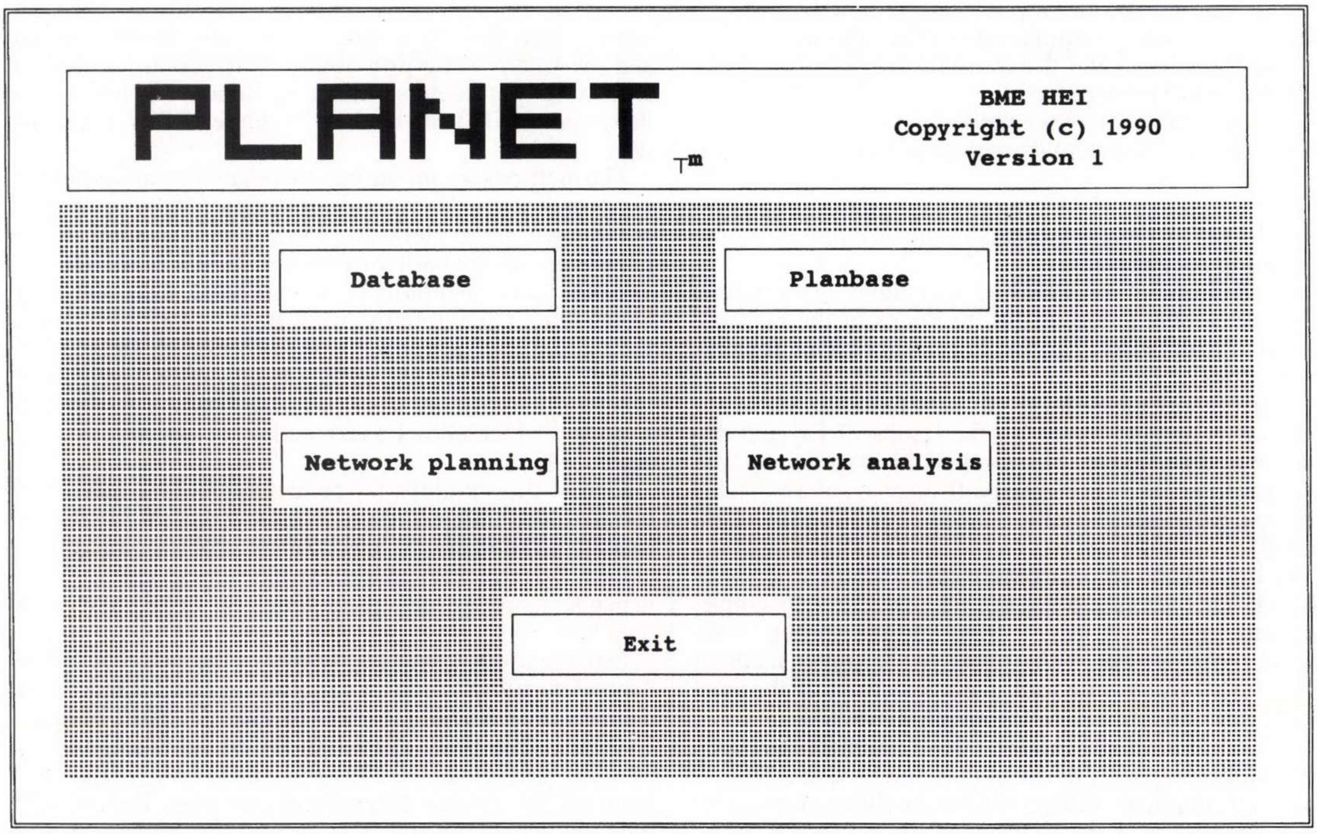

Fig. 1.
The main menu of PLANET

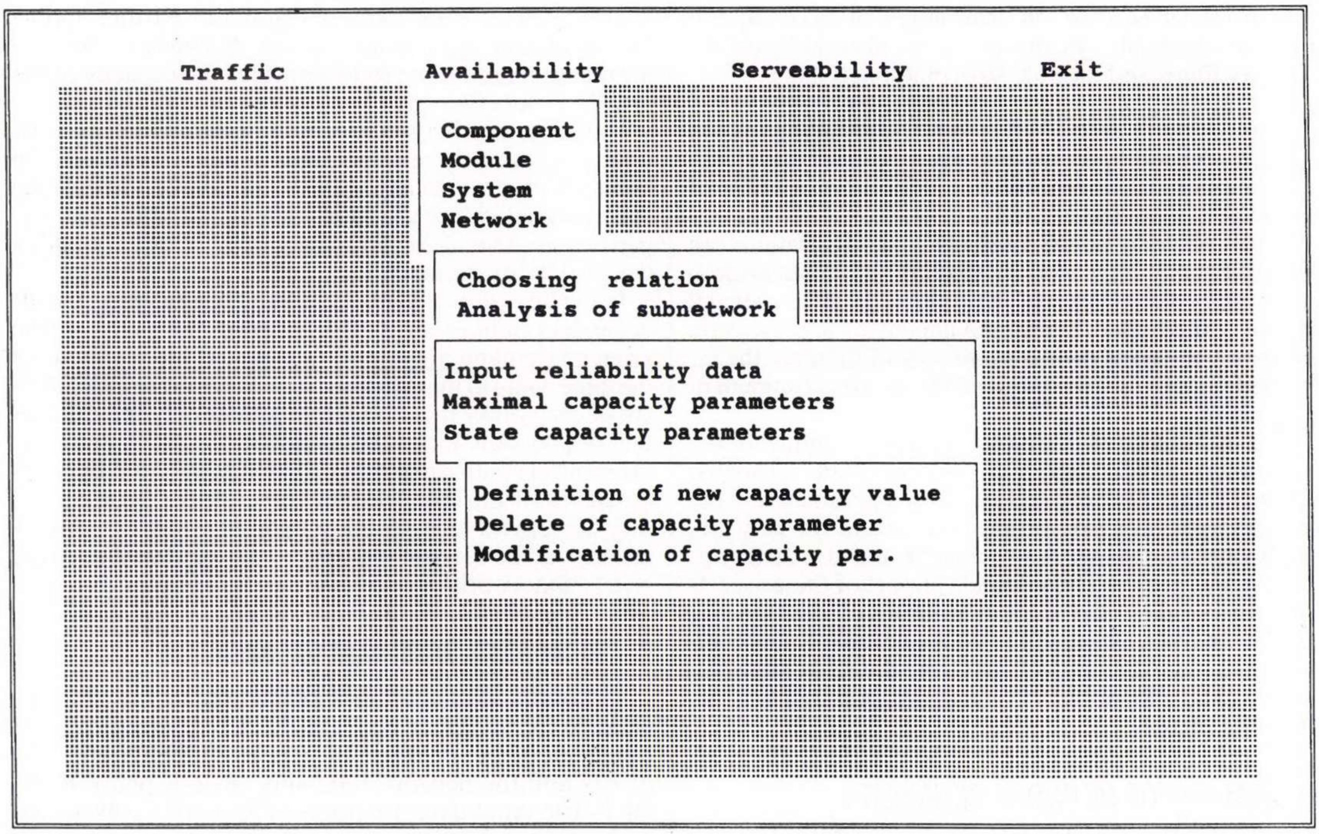

Fig. 2.
An internal menu of the network reliability analysis phase

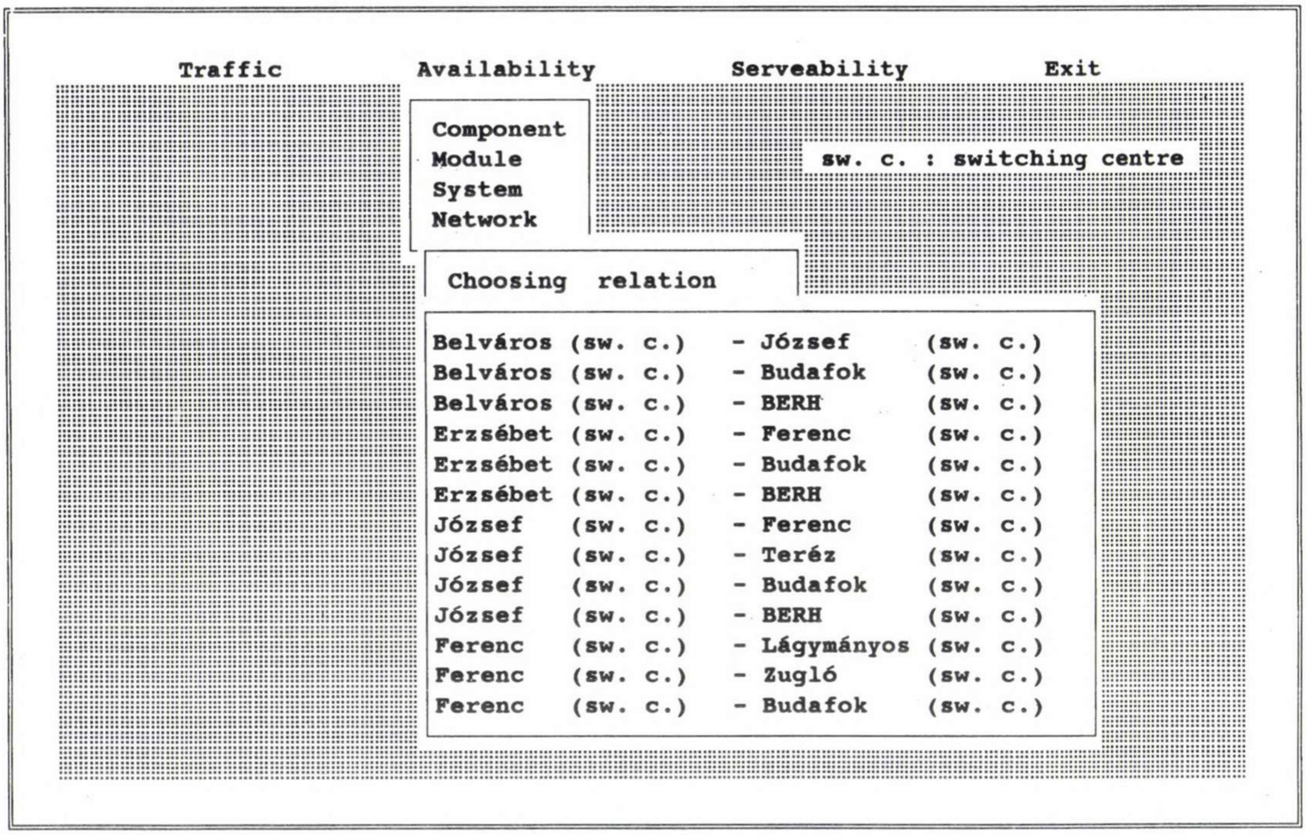

Fig. 3.
The window for the selection of a traffic relation

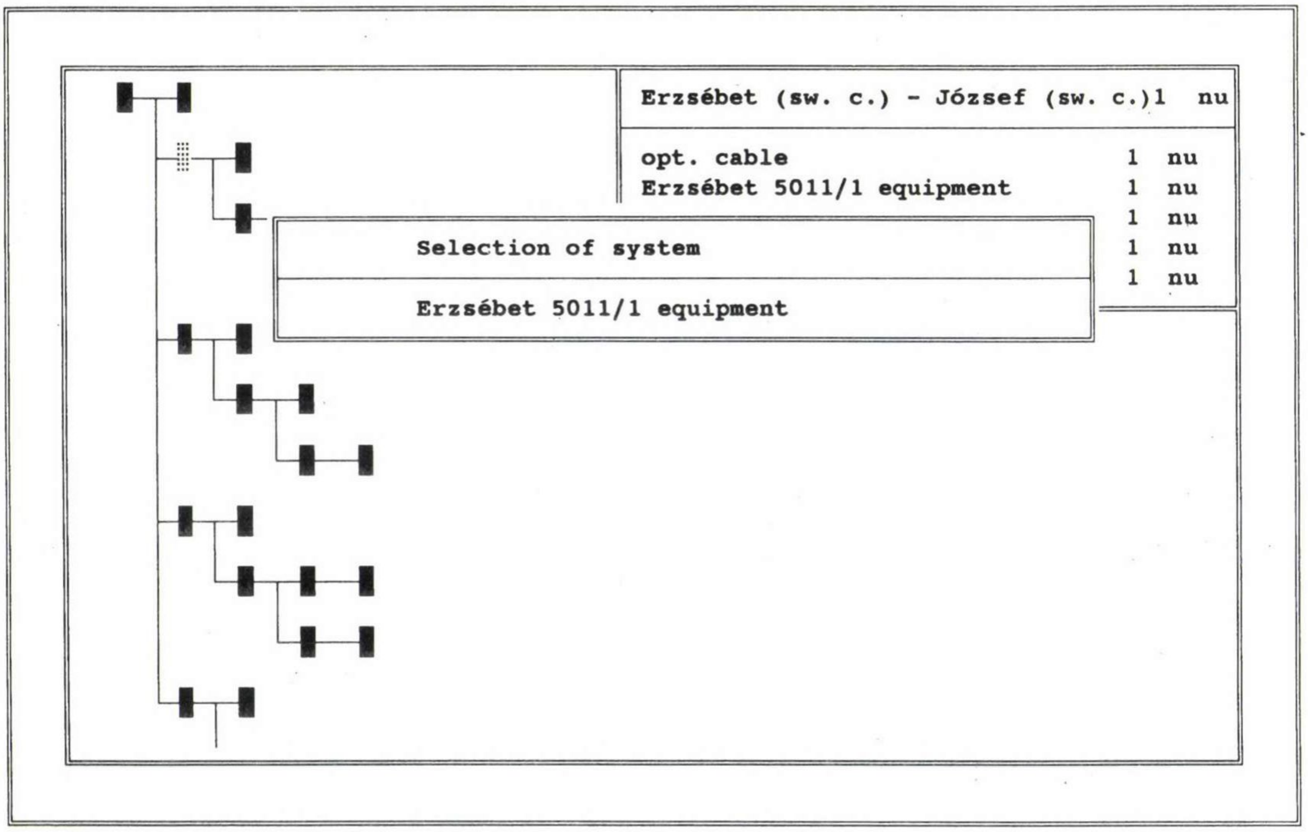

Fig. 4.
A part of the screen showing the structure of a given link

Compute exit probability

Number of states.......................... 5

Defined exit probability 2.50000E-06

Old computed exit probability 7.61317E-03

New computed exit probability 2.86780E-03

Press any key to continue!

Fig. 5.
An internal result of the reliability model generation

- 1 - **1990.10.12.**

Capacity values of relation

Network : mcdemo **Relation : Zugló (sw. c.)**
- Budafok (sw. c.)

index	capacity	probability
1	10	9.91545E-01
2	9	4.26888E-03
3	8	2.56693E-03
4	7	1.56023E-04
5	6	6.30951E-05
6	5	1.39690E-03
7	4	7.38620E-09
8	3	1.82623E-06
9	0	4.78421E-07

Press any key to continue! **ESC:abort**

Fig. 6.
The results of the availability analysis of a traffic relation

structure of the program package are depicted in Figs. 1 and 2, while Figs. 3 to 6 show the reliability analysis for a given traffic relation by using a window technique.

6. CONCLUSIONS

This paper introduced a method for integrated handling of network reliability problems. The described technique has been applied for the long- and medium-term planning of the Budapest trunk network and the Hungarian long-distance network.

There are two kinds of further activities. The first includes the handling of node reliability in a two-state manner and the improvement of network protection by rearrangable stand-by networks. These program packages are currently being implemented.

The extension of the node models to other areas is under preparation. These areas include the multistate reliability model of switching centres, the application of digital cross-connects in transmission networks, the adaptive routing of demands through the network, and the solution of the multiterminal network reliability modeling. Although there are known methods for solving these problems the implementation of these methods on a PC seems to be impossible. The purpose of these investigations is to implement these results by the new program packages for workstations.

7. ACKNOWLEDGEMENT

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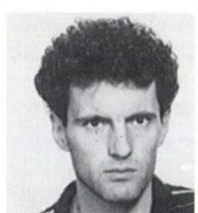

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COMPUTERIZED PLANNING OF TELECOMMUNICATION NETWORKS

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The paper reviews the reasons for using computers and heuristic design methods in telecommunication network planning. The major challenges of technological evolution for network design are then outlined. The paper presents a system of PC programs developed to assist in demand and traffic forecasting, optimization of exchange boundaries and locations, design of network hierarchy and structure, traffic routing strategy selection, circuit group sizing, network topology design, circuit routing etc. The PC planning system is available for metropolitan, long-distance and district-local telephone network planning. The basic concepts and the applications of the computer programs are also demonstrated.

1. REASONS FOR USING COMPUTERS

The purpose of telecommunication network planning is to reach an optimum compromise among the traffic demands of telecommunication services, the network modernisation objectives, the quality performance and the economic aspects by taking into account the existing networks, the equipment available and the fundamental technical and other constraints. Network planning involves the determination of the structure and design of the *intelligent* (service-controlling, management), the *logical* (functional, traffic-handling) and the *physical* (transport, transmission) *layers of the network.*

Normally, we try to find the *least-cost solutions* to the demands which satisfy the network planning standards, including grade-of-service, overload performance etc. We can have an inverse task, when the investment fund is limited and we try to find the *optimum allocation* of the fund available so that we satisfy the demands as close as possible. A third approach can be also stated, when we find the so-called *efficient solution,* with an optimum relation between the performance obtained and the cost required. This approach is very useful when the quality requirements are not completely defined or are under study (eg. in overload protection problems). General characteristics of the network planning and optimization problems can be summarized as follows:

- Great number of input data with uncertainties is to be handled.
- The variables have complicated relations with temporal and spatial couplings.
- A complex technical, financial and geographical constraint-system has to be taken into account.
- For several parameters, only certain integer values are feasible (eg. in digital networks, the size of circuit groups is a multiple of a module of circuits, eg. 30 circuits).

The planning tasks of network digitalization are extremely sophisticated, because

- the integration of switching and transmission,
- the utilization of stored program controlled (SPC) techniques and remote subscriber units,
- the mixed analogue-digital (A—D) network features, and
- the budget limitation of the A—D transition

initiate unavoidable temporal and spatial couplings among the decisions.

Therefore, the use of computers, computerized methods and data bases becomes indispensable so that

- systematic planning can be realized,
- more detailed models and more sophisticated procedures can be used,
- several alternate solutions can be stated, evaluated and compared,
- sensitivity analysis can be performed with respect to the changing and uncertain parameters.

For real-size problems the use of heuristics is inevitable even if high-performance computers are employed. Additionally, recently the professional personal computers (PC's) are widely used [1]...[5]. Great efforts have been made to find the most appropriate suboptimal heuristic planning methods, computation procedures, and to write computer programs to handle more and more aspects of the problems. Section 2 presents a typical heuristic optimization, procedure for solving large-scale network evolution problems. Section 3 summarizes recent challenges for network planners. Section 4 shows a computerized planning system for local and interexchange networks developed by the research institute of the Hungarian Telecommunications Company (recently: PKI Telecommunication Institute). [6]...[10].

2. NETWORK OPTIMIZATION HEURISTICS

The network planning process can be divided into roughly independent consecutive steps (Fig. 1):

1. Demand and traffic forecast.
2. Functional network planning (logical layer), including
   - definition of the traffic handling network structure (network hierarchy, location of transit/tandem centers, traffic routing principles)
   - traffic routing and circuit group sizing (optimization of the network inside the chosen structure under grade-of-service requirement in order to define the number of circuits for each relation).
3. Physical network planning (transport layer), including
   - definition of the transmission topology
   - circuit routing and grouping
   - link evolution planning.
4. Operational and management planning (intelligent layer) including the design of traffic controls and a standby transmission network.

Decomposition of the global planning task into these subtasks is associated with feedbacks and iteration. In addition to the decomposition, approximations are required to solve the planning subtasks. Approximations may involve

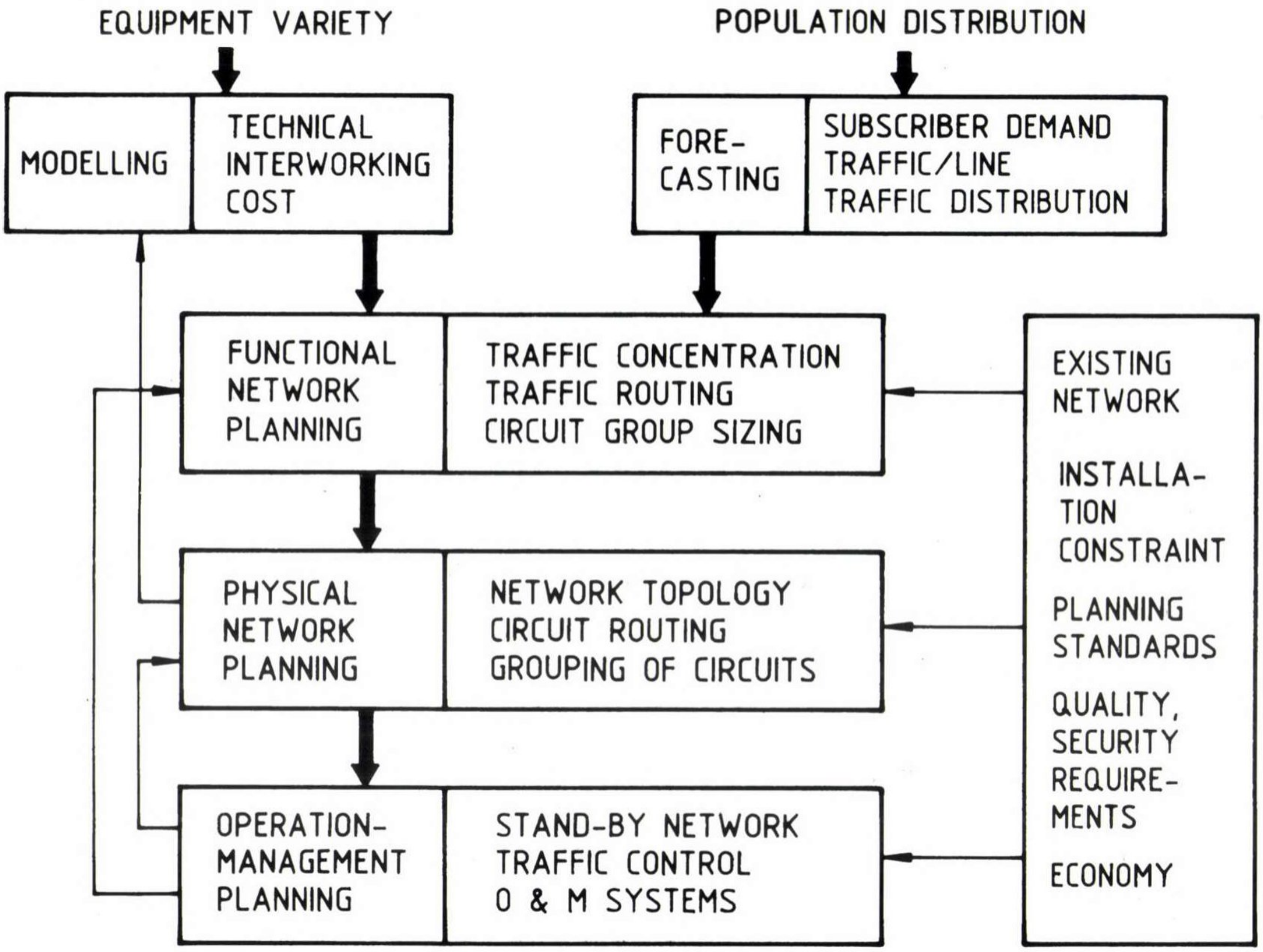

Fig. 1. The major steps of network planning

- construction of approximate network evolution models, eg. neglecting certain technical and financial constraints or network elements, modernization options (eg. the transmission costs are considered independent of the switching modernization strategy, the reuse of analogue exchanges is not considered);
- use of exact optimization algorithms to the partial problems of the subtasks and interlinking the partial solutions by iteration (eg. dynamic programming is used for planning the modernization of the switching centers separately, the interactions are taken into account through iteration);
- use of approximate, suboptimal planning methods. A typical example is the quasi-dynamic approach instead of dynamic programming: the optimization of a modernization process with respect to the intermediate time in carried out statically and sequentially, i.e. we consider the optimized network of the preceding time as the starting state and the target network as the evolution trend and take them into account as optimization constraints.

Thus a typical *heuristic optimization procedure* for solving large-scale network evolution and modernization problems is the following (Fig. 2):

1. The optimum modernization strategy for switching points are determined. (by the simplest approach the optimal transition strategies are determined for each switching point separately)
2. The circuit requirements between the switching points are dimensioned for every year of the concerned period under the selected traffic routing techniques.
3. The transmission network evolution is determined by using a quasy-dynamic approach to topology evolution planning, and dynamic programming to the link evolution planning.
4. The solution is analyzed, the estimations for cost parameters are checked. If necessary, a new iteration cycle is started with new cost parameters.
5. The least-cost solution (having minimum present worth of annual costs, PWAC) is analyzed with respect to the network protection requirements, annual budgetary constraints, etc. If the unconstrained solution is unfeasible, it is to be modified, eg. by rearranging evolution steps so that the total annual capital requirements remain within the specified budgetary constraints.

3. CHALLENGES FOR NETWORK DESIGN

The digitalization of the telephone networks including the introduction of the SPC techniques and common channel signalling (CCS) systems, as well as the establishment of integrated multiservice networks, in particular the integration of various bandwith services (broadband ISDN), require the reconsideration of network design methods and constitute several challenges. The major features of the digital network planning are the following:

1. *Both-way circuit groups* can be realized without extra cost over one-way groups, resulting in saving of cir-

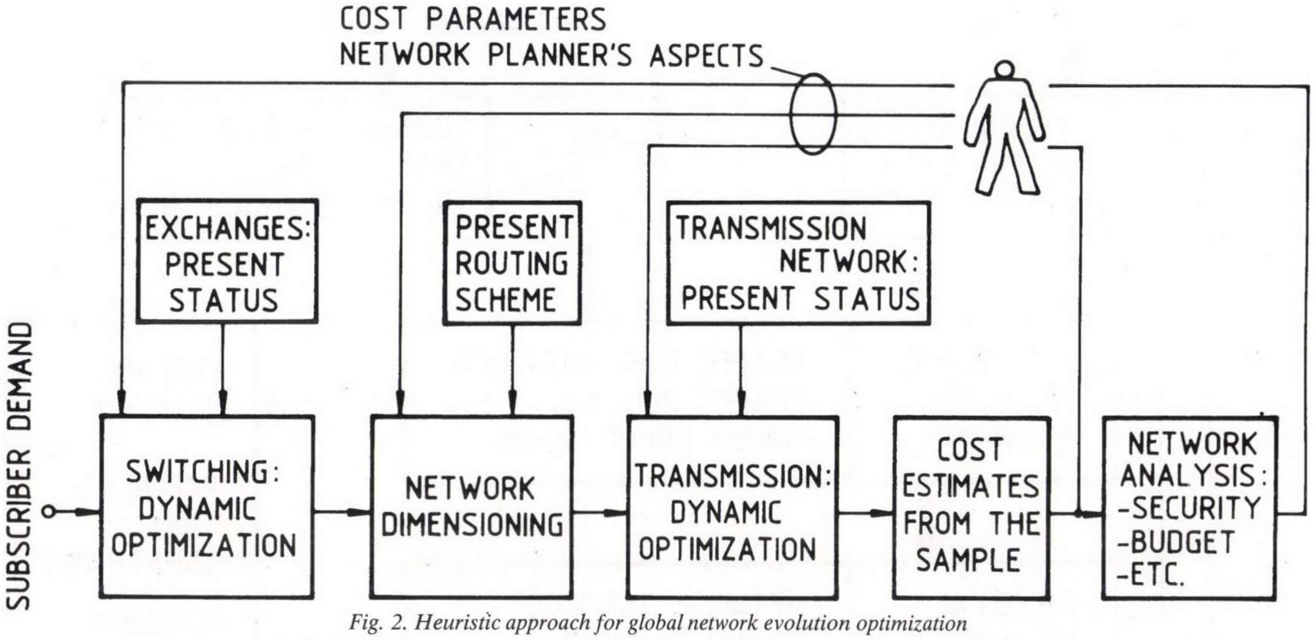

Fig. 2. Heuristic approach for global network evolution optimization

cuits groups, and usually require iterative circuit calculation.

2. Circuit groups must be provided on separate PCM transmission systems having intrinsic module sizes (24 or 30 channels), therefore *modular engineering,* the sizing of circuit groups in multiples of a fixed module size is required.
3. In SPC networks, alternate routing schemes are less limited, and can be *dinamically adjusted* to traffic situations with manually-controlled or preplanned program-based switch-over facilities. Multihour traffic dimensioning methods are further developed for these goals.
4. Using a CCS system, the hierarchical organization of the exchanges can be eliminated, and various *nonhierarchical routing schemes* can be introduced. Pragmatic, tractable routing schemes and design methods are investigated.
5. Efficient *traffic flow control* procedures may be easily implemented to avoid the degradation of network performance under traffic overloads and facility breakdowns. The unification of protective and expansive controls as well as routing procedures is extensively investigated.
6. In SPC/CCS networks, nonhierarchical *adaptive routing* can be established involving real-time traffic control with respect to the traffic fluctuations. The dimensioning problems are under study.
7. The integration of the various services, mainly the use of the *Asynchronous Transfer Mode* (ATM) as the target technique to be used in the future Broadband ISDN, raises new questions and problems to be clarified (eg. bandwith management).
8. The CCS system assists the digital telecommunication networks mainly as its nervous system to provide ISDN and IN (intelligent network) services, mobile services, advanced routing and control opportunities, etc. Special methodology for planning the *CCS network* is to be developed.
9. The new network elements (eg. remote subscriber units, digital cross-connect systems, digital radio acces systems), the large capacity digital switching systems and optical transmission systems provide new networking opportunities and radically change ratio of the cost components in the network models. The *hierarchical network structure* and the fundemental technical plans as routing, availability, charging, etc. plans are to be reconsidered to utilize the economic opportunities provided by digital technique, service integration and network intelligence.
10. The introduction of new services and technologies, necessitates the elaboration and evaluation of a lot of scenarios of network modernization, service diversification and integration. Multi-criteria decision making (MCDM) techniques, cost/performance studies are required to handle multi-goal planning objectives (life cycle costs, quality of service, security, etc.) and choose among planning options.

Consequence of these developments is a network with higher utilization of the resources and greater flexibility in handling abnormal situations.

The use of computers to solve the network design problems is inevitable. The use of heuristics, approximate models and methods is also needed to obtain results within a reasonable time.

4. COMPUTERIZED NETWORK PLANNING SYSTEM LONET-INTERNET

Since 1976, the PKI has been developing computer tools on PDP8/E and partly on IBM 370/170 computers to assist in telephone network planning. Recently, the program has been adapted to IBM PC AT professional personal computers, and developed to handle mixed analogue-digital and hierarchical-digital networks.

Use of a low-capacity computer requires the forced segmentation of the design process, a closer interaction of computer programs, and novel approaches, with emphasis on heuristics and probabilistic methods. Only a wellstructured, user-friendly system of computer planning tools is able to give consistent evaluation of the various development alternatives, to test the sensitivity of the optimum solution, and to provide flexibility in both the planning and in the further development.

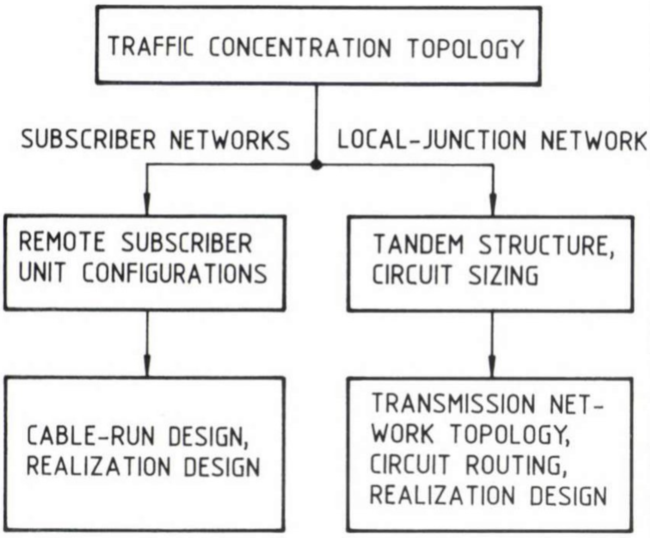

Fig. 3. Segments of urban network planning process

From the point of view of network modelling and applied methods, a certain dissociation of urban and interurban network planning is favourable. A more detailed diagram relating the functional and physical planning of multiexchange urban networks is shown in Fig. 3, indicating that as a result of the traffic concentration topology planning (including number of exchanges, their locations and boundaries), the whole network is divided into subscriber networks and the interexchange junction-network. A similar diagram may be drawn for interurban network planning, with branches of district (rural-area) networks and interdistrict (long-distance) network.

The computer program system has been developed for the local (urban) network planning (LONET subsystem) and for the interurban network planning (INTERNET subsystem). Presently it covers the forecasting, cross-sectional (static) functional and physical planning, and some elements of the design of network management. Several programs are used in both subsystems. A short definition of programs is indicated in Table 1 in alphabetic order, the main flow of their use is sketched in Fig. 4, not showing the feedbacks in the planning process.

The programs can be classified into two groups: generic programs providing orientations and basic concepts for the network planners, and specific programs aiding the planning of a defined network.

4.1. Generic programs

The generic programs (indicated by asterisks in Table 1) have increased importance due to the digitalization. To

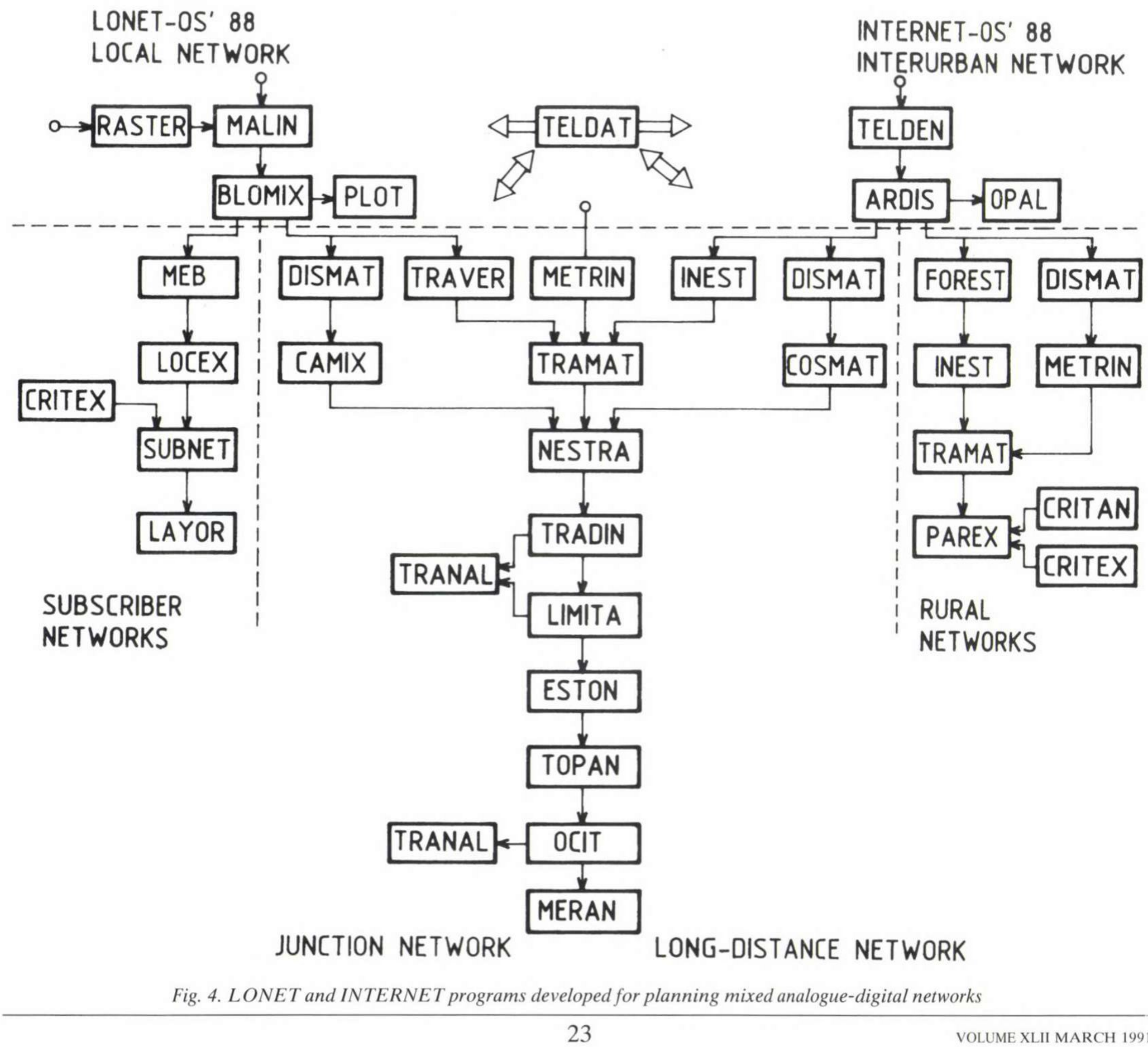

Fig. 4. LONET and INTERNET programs developed for planning mixed analogue-digital networks

Table 1: Program functions
(Generic programs are inidicated by*)

Name	Function	Application
ARDIS*	Finding least-cost assignment of localities to network levels under a given number of hierarchical levels and switching and transmission configuration, based on a statistical network mode	Nation-wide
BLOMIX	Optimal division of total urban area into exchange areas, optimal locations and capacities of exchanges, taking into account the type of the exchanges (analogue or digital, existing or planned), the geographical and transmission constraints as well as the applicability of remote subscriber unit	multi-exchange urban
CAMIX	Calculating the cost of interexchange circuits in mixed network, taking into account transmission and routing plan	junction
COSMAT	Calculating the cost of long distance trunks	LD
CRITAN*	Making orientative tables for deciding the establishment of a tandem switching point	rural
CRITEX*	Making decision tables for selecting the least-cost subscriber network configuration (switching and transmission)	subscriber and rural network
DISMAT	Determination of the length of circuits and their routing	junction LD, rural
ESTON	Economic optimization of the topology of physical network, starting from a feasible topology	junction, LD, rural
FOREST	Calculating the number of subscribers and originating traffics of localities	rural
INEST	Calculating the outgoing and incoming traffics of localities or territories and traffic interests between them	LD, rural
LAYOR	Determining optimum layout to a defined subscriber distributions taking into account the installation of decentres e.g. remote subscriber unit, multiplexer, cross-connecting point	subscriber network
LIMITA	Determination of the limited accessibility equivalent of the full accessibility circuit groups calculated by NESTRA and TRADIN	junction, LD
LOCEX	Optimal location of an exchange on a given area (described by subscriber matrix)	subscriber
MALIN	Calculation of matrices of subscribers of different classes from population and working place distributions, with density factors	urban
MEB	Separation of a part of total area	subs.netw.
MERAN	Rearrangement of an analogue-digital direct-routed urban network under line-failure condition, minimizing the maximum point-to-point congestion	junction
METRIN	Processing of the measured traffic data and calculation of the interest between the exchange	junction LD, rural
NESTRA	Evaluation of a defined traffic routing structure by approximatie optimum dimensioning of circuit groups, for two level alternate routed networks, under given grade-of-service criteria	junction, LD
OCIT	Determination of necessary analogue and digital transmission capacity expansions taking into account the existing capacities and the multirouting transmission requirements	junction, LD
OPAL*	Optimal allocation of available investment funds to improve the traffic throughput of a hypothetical hierarchical network	nation-wide
PAREX	Selection of optimal traffic concentration and facility configurations from the feasible ones, taking into account the transmission standards, and any other constraints, e.g. an existing network	rural
PLOT	Plotting the map of the concerned urban area, the distribution of the subscriber demands, the exchange locations and urban boundaries, etc.	multi-exchange
RASTER	Converting the map of an urban area into a grid form, used to describe the distribution of the population and working places by matrice	multi-exchange urban
SUBNET	Optimizing the locations and area boundaries for a given set of decentres (e.g. remote subscriber unit, multiplexer, cross-connecting point)	subscriber network
TELDAT	Database of localities, including global information to forecasting and on the existing facilitie	urban, LD, rural
TELDEN*	Calculating the telephone density for typical localities	nation-wide
TOPAN	Optimal multipath routing of circuits on a given network topology, determining the capacity demands	junction, LD
TRADIN	Dimensioning of alternate (and direct-tandem) routing networks based on Wilkinson's ERT and Moe's optimization methods taking into account the routing pecularities of the exchanges and the design constraints of the circuit groups	junction, LD, rural
TRAMAT	Determination and Kruithof-type correction of traffic matrix, taking into account the calculated interest between the exchanges	junction, LD, rural
TRANAL	Calculating congestion values for circuit groups and between exchanges in a given alternate routing network, under normal and overload situation	junction, LD
TRAVER	Calculating the originating and terminating traffics of local exchange areas	junction network

find the optimal hierarchical structure of target network, allocate the available investment funds in the existing network, preselect the efficient network configurations out of the numerous options, etc. All these require generic network models [10]—[14]. Using probabilistic approaches, well-tractable network models and reliable statistical solutions can be obtained even if only personal computers are available.

The most useful generic program for planning a nationwide analogue-digital mixed network is ARDIS [10], [11]. For calculating the average line length, the population centres are supposed to be randomly dispersed. The total area to be served is described by the distribution of the number of subscribers of the population centres. The network is supposed to be star-connected, defined by structural parameters expressing the number of nodes on each hierarchical level (distribution point, concentrator, local switch and transit levels). The traffic outgoing from a territory is counted by a parametric theoretical function obtained from experimental requirements and depending on the relative number of the territorial subscribers. The transmission, switching and building facilities, telecommunication infrastructures are characterized by fundamental technical, capacity and cost parameters. The cost of switches situated at a given hierarchical level depends on the capacity and type. From the transmission systems satisfying technical requirements, the cost-effective one is chosen. Program ARDIS is appropriate to determine the statistically optimal network hierarchy and structure (i.e., for example, the statistically optimal number of district exchanges, but not the actual district exchange locations) for a given technical configuration. Alternatively, this program can be used for comparing consistently the cost of different switching and transmission solutions, to examine the global cost consequence of the deviation from the optimal structure, and the dependence on the various cost and design parameters. The digitalization strategies can be investigated by various analogue-digital configurations for switching and transmission functions. The probabilistic approach is particularly suitable to model rural networks involving great numbers of population centres [10].

4.2. Specific programs

The specific programs support the planning and optimization of a defined local area or long-distance network, handling at least two different techniques (e.g. analogue crossbar and digital switching). The programs cluster around a node program which is the main program of a dedicated segment, and the solution to the segment requires their multiple interactive use.

Main features of the applied models are:

- The subscriber forecasting is based on normative methods. The traffic forecasting is also normative, but the results of traffic measurements are also considered.
- The programs BLOMIX and SUBNET use matrix description of the considered urban area [6], [15], the PAREX uses a graph model to represent the population centres of the considered rural area [16].
- The traffic routing is supposed to be a deterministic, hierarchical alternate routing. Sizing methods for handling the modularity and protecting the network against traffic overloads are involved [17], [18].
- Additional nodes to the switching nodes can be defined in the transmission topology but their location should be given externally [19]. Multirouting requirements are handled, taking into account the use of a standby network [20], [21].
- The network protection planning is based on the efficiency approach. The protection methods (multirouting of circuits, splitting final routes, etc. are dimensioned by optimizing the performance/cost of the concerned method, and the efficient combination of the various methods is calculated [21], [22], [23].

All specific programs are linked up with a number of files through which data are transferred from one program to the other.

4.3. Practical applications

The outlined computerized planning system is used in the strategic, long- and medium-range planning of metropolitan, urban and suburban networks, as well as long-distance and rural networks.

The LONET subsystem has been used in the planning of the future network of some Hungarian towns among them the Budapest metropolitan network, to determine the optimal number and locations of exchanges, the optimal network configuration for every exchange area, and the optimal solution of the junction network.

A long-term plan of the *Budapest multiexchange network* to cater for 900.000 main lines (44/100 inhabitants) was prepared in 1978 using crossbar exchanges. The optimal structures of the Budapest area network had 33 exchanges. The plan has been recently revised, investigating the impact of the introduction of digital switching technique. The long-term calculations show a significant reduction in the overall cost (13,5%) compared with the fully analogue switching configuration, as well as a reduction in the number of exchanges to be established. The optimum total number of exchange locations is 26 to 31, depending on the type of digital switching system.

The programs are also used for planning the *Hungarian interurban network.* A routing plan was prepared in 1977 for the long-distance network planned with ARM crossbar transit exchanges. The routing plan was revised in 1988 to involve the analogue-digital mixed network pecularities.

Using the developed probabilistic methods, the impact of the introduction of digital techniques into the interurban and rural network has been extensively studied.

We elaborated and compared several versions for *the architecture of the Hungarian telecommunications network* related to different evolution plans, rates of the digitalization and administrative organizations of the localities. We investigated the impact of the introduction of different digital switching systems while tripling the number of subscribers during the next decade. The results obtained have been applied to define the target network, eliminate two hierarchical levels, determine the number of regions (secondary zones), districts (primary zones), local exchanges and remote switches. The 9 regional exchanges are fully interconnected. The majority of the 56 districts has a combined local-transit exchange and about 10 remote units. The elimination of the two hierarchical levels results in a saving of 16% or so. As a result of extended numerical studies, we can state in general sense, that:

a) A least-cost digital nation-wide target network typically has a regional level with fully interconnected transit exchanges, and a district level with combined local-transit exchanges. The subscribers are connected directly or via remote switches.

b) The average cost per subscriber depends less on the subscriber's class (residental or business) and the hierarchical level of the subscriber's locality if the number of levels is reduced or the ratio of digitalization is greater.
c) Although the total network cost has a flat minimum versus structural parameters, the optimum structure is robust to the design and cost parameters, and the actual structural parameters have been given in a 10 percent vicinity of the statistically optimum values.

In addition to domestic applications the Hungarian network planners also utilized their experiences in the planning of the telecommunication of other countries, and the network planning computer tools are being adapted to the particular requirements of other telecommunications administrations. At the same time, the LONET-INTERNET system is permanently further developed by taking into account the experiences of other personal-computer aided design systems and the peculiarities and opportunities of digital networking. [24]—[28].

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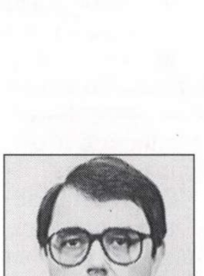

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ON SPEECH CODING IN A NUTSHELL

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In recent years speech researchers witnessed an unprecedented evolution in the field of low bitrate speech coding. The introduction of the revolutionary analysis-by-synthesis concept has been facilitated by an equally impressive development in DSP hardware technology, which culminated in the birth of floating-point signal processors.

In this contribution after a short classification of speech coding methods the concept of analysis-by-synthesis hybrid coding using long-term prediction and perceptual error weighting is introduced. Multi- and regular-pulse excited codecs are shown to provide toll-quality speech around 10—16 kbit/s at medium to low complexities, while stochastic codecs deliver near-toll-quality at about half the rate and quadruple complexity. A brief system-oriented speech-quality/complexity/bitrate/robustness comparison concludes that code-excited stochastic codecs with a bitrate around 7 kbits/s constitute today's challenge for the microelectronics and tomorrow's answer to bandwidth-efficient telecommunications.

1. INTRODUCTION

In designing a telecommunications system one of the most salient parameters is the number of subscribers that can be accommodated by the transmission media utilised. Whether it is a time division multiplex (TDM) or a frequency division multiplex (FDM) system, whether it is analog or digital, the number of subscribers is limited by the channel capacity needed for one speech channel. If the channel capacity demand of the speech channels is halved the total number of subscribers can be doubled. This gain becomes particularly important in applications like power and band-limited satellite or mobile radio channels where the urging demand for free channels over-shadows the inevitable cost-constraints imposed by a more expensive narrow-band speech codec. In the framework of the basic limitations of state-of-art LSI technology and theory the design of a speech codec is based on an optimum trade-off between lowest possible bitrate and highest possible quality, at the price of lowest possible complexity, cost and system delay.

Although these factors impose rather contradictory requirements, they broadly predetermine a few classes of speech codecs for a given application. In this contribution, motivated by perspective application in digital mobile radio and satellite systems, we focus our attention on digital speech coding methods yielding toll or near-toll quality speech at bitrates around or below 16 kbit/s, with the provision of surmountable technological complexity and price. Due to the evolution of the integrated services digital networks (ISDN) we are also interested in the transmission of other types of information such as signalling and data, hence methods which are possibly signal independent are greatly preferable. As we intend to propose speech codecs for rather hostile channels, their robustness against transmission errors is also a very important factor in our quest. Finally, apart from the broad subjective term of "toll-quality", we have not precisely defined the targeted speech quality as yet. Well, this involves a quality which is perceptually hardly influenced by the encoding/decoding process. Since the reliable objective measurement of speech quality is not a solved problem yet, it can only be characterized by a rather extensive set of objective distance measures [1] such as various weighted signal to noise ratios, weighted spectral distortion measures, spectral envelope measures, articulation index, or the Itakura logarithmic likelihood ratio. Hence, the subjective assessment of speech quality by the Mean Opinion Score (MOS), Diagnostic Acceptability Tests or using the popular Modulated Noise Reference Unit (MNRU) to quantify the Opinion Equivalent Q (Q_{op}) [dB] is indispensable, although for quick comparisons we use in our discussions the well established segmental SNR (SEGSNR) measure.

In this short contribution we give an elementary description of some speech coding techniques and elaborate on analysis-by-synthesis codecs using perceptual error weighting in somewhat more detail. Finally, a simulation-based rudimentary comparison is given to highlight the pertinent system's trade-offs when opting for a specific codec for a given communications network.

In harmony with well-known classical references [2]-[10], speech coding methods are broadly categorised as *waveform coding, source coding* and *hybrid coding.*

2. WAVEFORM CODING

Basically, waveform codecs are designed to be signal independent by mapping the input waveform of the encoder into a faximile-like close replica at the output of the decoder. Due to this advantageous property they can encode also secondary type of information such as signalling tones, voice band data, or even music. Naturally, because of this transparency, their coding efficiency is usually quite modest. The coding efficiency can be improved by *exploiting some statistical signal properties,* if the codec parameters are optimised for the most likely categories of input signals, while still maintaining good quality for other types of signals as well.

The waveform codecs can be further subdivided into:

- time domain waveform codecs and
- frequency domain waveform codecs.

2.1. Time Domain Waveform Coding

The most well-known representative of signal independent time domain waveform coding is A-law companded pulse code modulation (PCM), which has been standardised by the CCITT at 64 kbit/s, using non-linear companding characteristics to result in near-constant signal-to-noise ratio (SNR) over the total input dynamic range. Also well-known are the 32 kbit/s adaptive differential PCM (ADPCM) standardised in the CCITT Recommendation G.721 and the adaptive delta modulation (ADM), where usually the last signal sample or a linear combination of the last few samples is used to form an estimate of the current one. Then their difference signal, the prediction residual, is computed and encoded usually with a lower number of bits, since it has a lower variance than the incoming signal. This estimation process is actually linear prediction with fixed coefficients. However, owing to the non-stationary statistics of speech, a fixed predictor can not consistently characterise the changing spectral envelope of speech signals. Adaptive predictive coding (APC) schemes utilise in general two different time- varying predictors to describe speech signals more accurately. Namely, one short-term predictor (STP) and one long-term predictor (LTP), where the STP is utilised to model the speech spectral envelope, while the LTP is deployed to model the line-spectrum-like fine-structure representing the voicing information due to quasi-periodic voiced speech.

In summary, time domain waveform codecs treat the speech signal to be encoded as a full-band signal and try to map it into as close a replica of the input as possible. The difference amongst various coding schemes is in their degree and way of using prediction to reduce the variance of the signal to be encoded so as to reduce the number of bits necessary to represent it.

2.2. Frequency Domain Waveform Coding

In frequency domain waveform codecs the input signal undergoes a more or less accurate short-time spectral analysis. Clearly, the signal is split into a number of sub-bands, and the individual sub-band signals are then encoded by using different numbers of bits, to obey rate-distortion theory on the basis of their prominence. The various methods differ in their accuracies of spectral resolution and in the bit-allocation policy (fixed, adaptive, semi-adaptive). Two well-known representatives of this class are sub-band coding (SBC) [21] and adaptive transform coding (ATC).

3. SOURCE CODING

The class of source coders is based on a-priori knowledge about the way the signal to be encoded was generated at the source. Accordingly, instead of trying to produce a close replica of the input signal at the output, the appropriate set of *source parameters* is found to characterise the input signal sufficiently closely for a given period of time. These source paramaters are quantised and transmitted to the decoder to synthesize a replica of the original signal.

To be more specific, in the linear predictive coding (LPC) sub-class of source codecs the following speech production model is utilised. An autoregressive (AR) or autoregressive moving average (ARMA) [2] time-varying linear system (TVLS) is used to characterise the vocal apparatus, and an excitation signal of varying complexity is used to describe the voice generating source. Accordingly, once the vocal apparatus has been descibed by its AR or ARMA model, the central problem of coding is how to find the simplest yet adequate excitation for high quality parametric speech representation. Strictly speaking this separable model represents a gross simplification of the vocal apparatus, but it provides the only practical approach to the problem. The simplest possible source representation model is where, after a voiced/unvoiced decision, the excitation generator supplies either a pitch-spaced pulse train or white noise to the TVLS constituted by the so-called all-pole synthesis filter H(z). It is plausible that the quality of this type of systems is predetermined by the adequacy of the model, rather than the accuracy of the quantisation of these parameters. This means that the quality of source coders can not simply be enhanced by increasing the accuracy of the quantisation, i.e., the bitrate, it is fundamentally limited by the fidelity of the model used. The main advantege of source coding tecniques is their low bitrate, with the penalty of relatively low, synthetic speech quality. A well-known representative of this class of vocoders is the 2400 bps American Military Standard LPC-10 codec [16].

Source coding techniques can also be categorised into frequency domain and time domain sub-classes. However, frequency domain source coding methods, such as the channel vocoder, are more effective than their time domain counterparts [6].

4. HYBRID CODING

Hybrid coding methods constitute an optimum trade-off between waveform coding and source coding, both in terms of speech quality and transmission bitrate, although usually at the price of higher complexity. Every speech coding method, combining waveform and source coding methods to improve the quality and reduce the bitrate, falls into this broad category. Yet, adaptive predictive time domain techniques used to describe the human spectral shaping tract combined with an accurate modell of the excitation signal play the most prominent role in this category.

In such schemes we tacitly assume that the speech production process can be well approximated by a separable linear model, where the excitation generator is independent of the time varying linear system $H(z)$. This is equivalent to sayins that the synthesized speech $S(z)=E(z)$. $H(z)$ can be generated in various combinations of the excitation $E(z)$ and the spectral shaping system $H(z)$. However, once the prediction coefficients a_i, $i=1\dots p$, describing the spectral shaping tract

$$H(z)=\frac{1}{A(z)}=\frac{1}{\left(1-\sum_{i=1}^{p}a_i z^{-i}\right)} \tag{1}$$

have been found [2], the problems is reduced to finding the excitation, which results in the smallest mean squared error (MSE) between the input speech $s(n)$ and the synthetic speech $\hat{s}(n)$. In other words, we input all the legitimate excitation models to the synthesis filter, compute the sample-by-sample difference of the input speech signal $s(n)$ and the synthetic speech $\hat{s}(n)$ over the whole LPC analysis frame, and choose that specific excitation model which results in minimum mean squared error, i.e., in the most similar synthetic speech to the speech to be modelled. Due to this excitation optimisation this family of codecs is often referred to as 'analysis-by-synthesis' codecs. The dilemma to be resolved is what error minimisation criterion leads to the *perceptually best synthetic speech.* To answer this question we again use our a-priori knowledge about the characteristics of speech signals and modify the spectral distribution of the error to exploit the masking property of the ear. Accordingly, we deemphasize the error in the frequency regions around the formant frequencies, because in these spectral domains we can afford a higher error component, as also the signal power is relatively high. This results in uniform SNR across the frequency band, rather than uniform error distribution. This policy provides us with a perceptually meaningful error criterion, whereas the appropriate error weighting filter $W_1(z)$ is practically available anyway, since it is a transformed version of the all-pole filter $A(z)$:

$$W_1(z)=\frac{A(z)}{A\left(\frac{z}{\gamma}\right)}=\frac{1-\sum_{i=1}^{p}a_i z^{-i}}{1-\sum_{i=1}^{p}a_i \gamma z^{-i}}, \qquad (2)$$

where the parameter $0<\gamma<1$ controls the degree of error deemphasis in formant regions, as portrayed in Fig. 1. Observe that by introducing this perceptual error weighting we have *actually degraded the quality of waveform reproduction in favour of the perceptually better speech quality.*

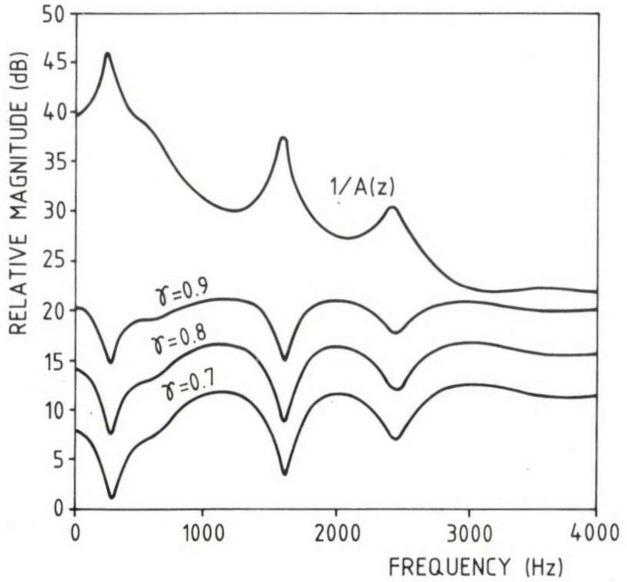

Fig. 1. Spectrum of $1/A(z)$ and $A(z)/A(z/\gamma)$ for different values of γ

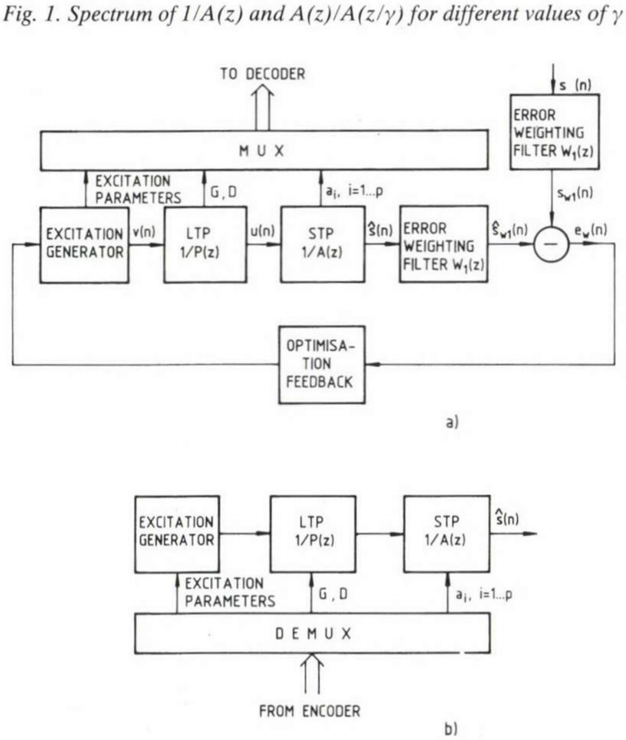

Fig. 2. Time-domain hybrid c. dec with LTP

Based on the above ideas a fairly general time domain hybrid coding scheme can be derived, the operation of which is best understood from Fig. 2. A 20 ms long frame of low-pass filtered original speech samples $s(n)$ is weighted by the error weighting filter $W_1(z)$, which deemphasises the speech signal in the formant regions to produce the weighted speech $s_{w1}(n)$ for comparison with the weighted synthetic speech $\hat{s}_{w1}(n)$. The synthetic speech is produced with the help of the excitation generator, outputting the so-called 'innovation sequences' $v(n)$, which are then filtered through the long-term predictor (LTP)

$$1/P(z)=\frac{1}{1-\sum_{k=-m}^{m}G_{kZ}{}^{-(D+k)}}, \qquad (3)$$

to generate the short-term prediction residual $u(n)$. Here $m=0$ yields a one-tap, $m=1$ a three-tap LTP.

Details of long-term prediction are available in the literature [14], in our practical approach here we simply state that the optimum LTP delay is the specific time-lag in the buffer storing the previous history of the STP residual, where the highest correlation between the present and the buffered previous ones is found. In practical terms this is either the pitch period or a multiple of it in case of voiced sounds. Once this most highly correlated segment has been found in the search buffer, it has to be appropriately scaled by the gain G to minimise the MSE and subtracted from the present STP residual: $v(n)= =u(n)-G\cdot u(n-D)$. Incidentally, the MSELTP prediction error

$$E=\sum_{n=0}^{N-1}e^2(n)$$

is minimised if the scaling factor or gain is computed as the normalised cross correlation of the present STP residual and its highly correlated previous history at delay D, which is expressed as:

$$G=\frac{\sum_{n=0}^{N-1}u(n)\,u(n-D)}{[u(n-D)]^2}, \qquad (4)$$

where $40<N<60$ is the length of the segments correlated. Then the minimum MSE is yielded as:

$$E_{\min}=\sum u^2(n)-\frac{\left[\sum_{n=0}^{N-1}u(n)u(n-D)\right]^2}{\sum[u(n-D)]^2}. \qquad (5)$$

Clearly, the second term of Eq(5) has to be maximised over the range of the legitimate delays in the STP residual buffer. This might appear prohibitive for some 100 possible positions in terms of complexity, but a simple step-by-step updating procedure, reduces the number of operations needed for the computation of Eq(5) to $(N+3)$ for each delay.

Filtering the short-term prediction residual $u(n)$ through the short-term predictor (STP) synthesis filter $1/A(z)$, where $A(z)$ is the z-domain transfer function of the all-zero LPC analysis filter, gives the synthetic speech $s_{w1}(n)$. Observe that the polynomials A(z) in the numerator of $W_1(z)$ and in the STP synthesis filter cancel, which is a significant simplification, justifying the separate weighting of $s(n)$ and $\hat{s}(n)$, rather than weighting $e(n)$ in the optimisation feedback loop. The source model is represented by that particular innovation sequence v(n), which minimises the mean squared weighted error $e_w(n)$ between the original and synthetic speech. Once the optimum excitation has been found, the relevant excitation parameters (for example pulse positions and amplitudes, or codebook indeces, etc), the one-tap long-term predictor gain G and LTP delay D, as well as the short-term prediction parameters a_i, $i=1\ldots p$, are appropriately quantised, multiplexed for transmission and sent to the decoder to recover the original speech. In the decoder the parameters mentioned are decoded and used in exactly the same way, as in the encoder to generate the synthetic speech.

The removal of redundancy is comprehensively portrayed in Fig. 3, where 120 ms female speech (six 20 ms long LPC frames) is depicted when fed through consecutive STP and LTP filtering using one, three or five LTP taps. Observe the gradual prediction error amplitude reduction from the top to bottom, as well as the removal of periodicity, i.e., that of predictable redundancy. After STP analysis filtering the LPC residual still exhibits the characteristic quasi-periodic pitch-peaks, which are removed by the LTP, rendering the LTP residual to become unpredictable, i.e., noise-like. It is also worth noting that the deployment of three LTP taps rather than one further reduces the prediction residual, but increases the computational complexity and requires further side-information transmissions. Therefore in practical systems one-tap LTPs are utilised. To further augment exposition it is interesting to review briefly the effects of prediction also in spectral domain. The highly non-uniform formant structure of the input speech clearly indicates the inherent redundancy, and due to quasi-periodicity we have a line-spectrum-like fine-structure. The spectral envelope prominences around the formant regions are flattened and the pronounced fine-structure disappears in the LPC residual after STP analysis filtering. Finally, the LTP analysis filter produces an almost noise-like 'coloured' spectrum exhibiting minimum redundancy. Clearly, we represented the predictable component of the speech by modelling the vocal tract's spectral shaping apparatus using the STP and LTP parameters, and we are left with the problem of finding the most efficient parametric source representation to describe the unpredictable.

4.1. Subclasses of Hybrid Coding

Hybrid coding techniques can be categorised on the basis of how the excitation $v(n)$ to the LTP has been generated in Figure 2. A historically important approach has been suggested by Atal [11], which is called the *multipulse excitation LPC (MPE-LPC) method,* in which no voiced/unvoiced decision is necessary, instead a number of pulses are allocated (typically 8—16 per 20 ms LPC analysis frame), one at a time in a very intelligent way. Namely, all the possible locations in a frame are associated with their optimum amplitudes and used to synthesize the speech. That specific position is chosen for the next pulse to be allocated, which minimises the weighted error between the original and synthetic speech. The entire innovation sequence is determined, if all pulse positions and amplitudes have been computed.

A closely related method is the so-called *regular pulse excitation LPC (RPE-LPC) scheme,* in which the prediction residual is computed by LPC analysis, as usual, and then decimated by a factor of 3 or 4 for example, to give 3 or 4 candidate excitation sequences, each shifted by one position with respect to the previous candide excitation. It can be shown [12] that the particular candidate innovation sequence with the highest energy represents the prediction residual most closely and hence results in the smallest error between the original and synthetic speech, i.e., best speech quality. Although in this scheme the number of pulses is typically higher, than in the MPE-LPC method (usually the framelength/3—4), it is sufficient to transmit only the gridposition or "phase", associated with the sucessful candidate excitation, plus the pulse amplitudes, no pulse positions have to be transmitted to the receiver. Hence this method results in similar bitrate at similar quality, as the MPE-LPC arrangement at somewhat lower complexity. Recently this method has proven to have very advantageous general properties around 13

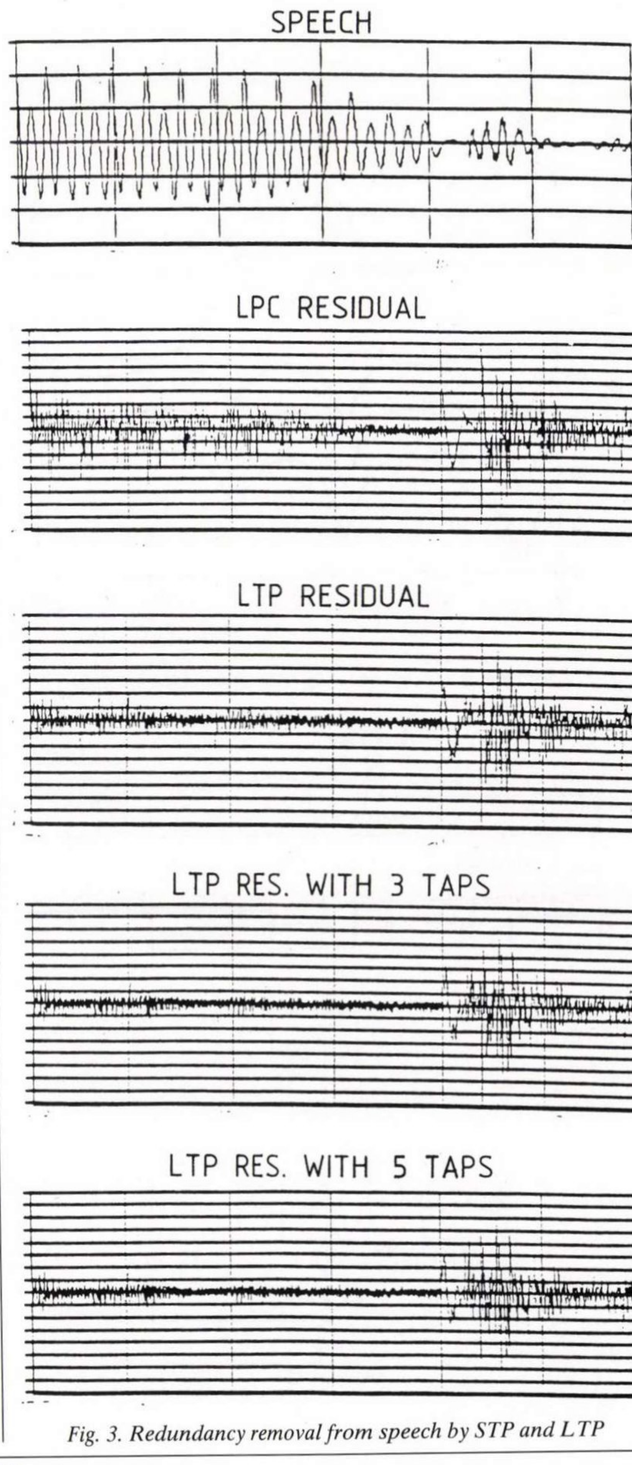

Fig. 3. Redundancy removal from speech by STP and LTP

kbit/s bitrate and hence it has been selected and standardised for implementation in the Pan-European Mobile Radio Network [13], where the MPE-LPC scheme was one of its closest competitors.

Now we focus our attention shortly on a new and around 4.8 kbit/s transmission rate surprisingly high-quality hybrid coding subclass, the group of *stochastic codecs.* In this category sometimes several long-term predictors (LTPs) are deployed to result in as random an LTP residual as possible. Then this noise-like LTP residual does not have to be transmitted at all. The decoder is initialised by a random noise process and then consecutive LTP residuals are recovered by correlation techniques from the past history of the excitation itself, by using an appropriately selected gain factor, which leads to the so-called *self-excited vocoder (SEV)* [14]. Alternatively, an identical zero-mean, unit-variance random code-book can be used in both the encoder and the decoder, along with a gain factor to model the noise-like excitation. Then simply the address of the particular code-book excitation, resulting in the lowest weighted error has to be sent, which is the algorithm of the so-called *code-excited LPC (CELP) method* [15].

The CELP method is slightly more computationally complex, than the SEV scheme assuming the same bitrate and quality. This is because in the SEV the correlated past history of the exacitation is generated by the codec itself, while in the CELP method the code-book has been generated by a totally uncorrelated external process. Therefore the same quality usually requires a quadruple-sized CELP codebook and whence yields higher complexity. Nonetheless, since CELP codecs are more robuts against error propagation, they are more favourable over noisy or bursty channels.

The importance of the CELP codec is also reflected in the fact that it was selected as the 4.8 kbps US Military Standard codec, and it is the most likely candidate for the half-rate GSM codec. Finally, it performs very well upto bitrates of 16 kbps and whence it is also the strongest candidate for the CCITT 16 kbps standard.

5. COMPARISON OF SPEECH CODECS

Speech codecs belonging to the various categories reviewed have profoundly different attributes in terms of speech quality, complexity, robustness against channel errors, transmission bitrates, etc. This complex interplay of various codec parameters is broadly characterised in Fig. 4 for the three speech codec categories mentioned.

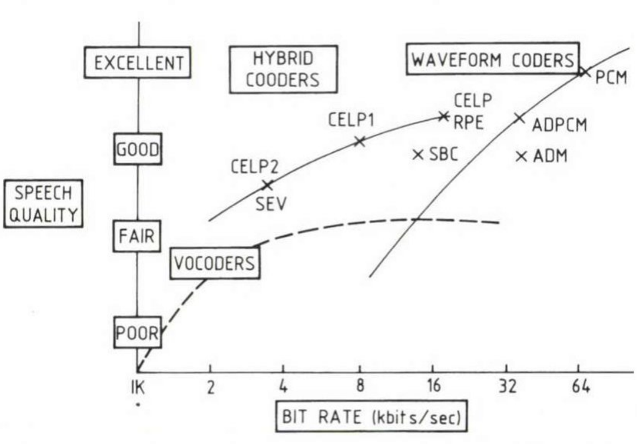

Fig. 4. Subjective speech quality versus bitrate for various speech conding classes

Generally speaking waveform codecs are unsuited for toll-quality (MOS ≈ 4.0) speech transmissions below 32 kbit/s transmission rate, although they perform very well above this bitrate. Albeit vocoders have very low transmission bitrates, their quality is fundamentally limited by the quality of their source representation, therefore it cannot be improved by simply increasing their bitrate. High complexity hybrid codecs nicely fill in the gap between the former two classes in terms of both bitrate and speech quality. Whence analysis-by-synthesis codecs play a prominent role in recent developments such as those for cellular mobile radio systems and mobile satellite systems.

In the framework of this short contribution it is impossible to give any rigorous comparison of the bewildering plethora of exotic speech codecs. Nonetheless, in Fig. 5 we attempt the impossible, and based on our simulations we provide a rudimentary comparison of some of the codecs reviewed. Details of the codecs listed on the horizontal axis are available in references [17], [18], [21], [19], [20], [15], [22] and [14], respectively. The bitrates measured on the right-hand-side vertical axis are ranging from 32 kbps down to 4.8 kbps. The SEGSNR values shown on the left-hand-side vertical axis are deceptive in comparisons, because the subjective quality or MOS is non-linearly related to it and we cannot even claim that two different codecs with similar SEGSNRs have similar subjective qualities.

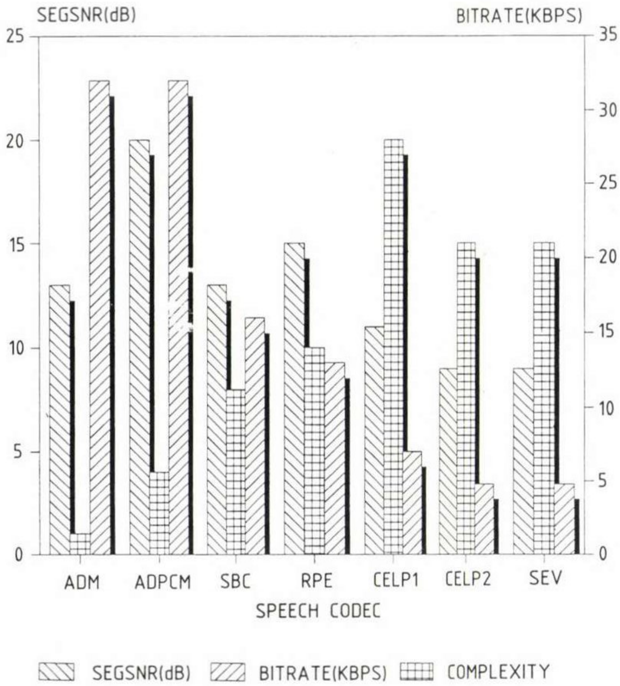

Fig. 5. Quality/complexity/bitrate comparison of various speech codecs

For pure waveform codecs like the ADM [17] and ADPCM, the SEGSNR gives a somewhat better idea about the perceived quality. Therefore, when comparing both of these codecs, at identical bitrates approximately quadruple complexity is needed to improve the SEGSNR by around 7 dB in a specific implementation using a variety of male and female speakers. Due to linear prediction more redundancy is removed in the ADPCM codec, thence the lower-variance prediction residual is more finely quantised using the same number of bits. However, as a consequence of lower redundancy as well as due to the differnet weights of the four ADPCM bits of a residual sample the codec becomes less resilient against channel errors than the ADM codec transmitting bits of identical importance.

The SBC codec [21] is more intelligent in allocating the channel capacity, i.e., the bits available, than the above mentioned codecs, therefore it reduces the bitrate to around 16 kbps, while still maintaining the SEGSNR of the ADM. Subjectively it has a somewhat better quality than the ADM at half the bitrate, an estimated eight-fold complexity and slightly lower robustness. The difference in robustness is attributable mainly to the fact that, while in ADM all bits have identical significances, in SBC the corruption of the most significant bits (MSB) of the ADPCM coded subband signals has rather disastrous consequences. Although more intelligent than full-band waveform codecs, the SBC still has the fundamental limitation of an 'open-loop' system when compared to the most advanced analysis-by — synthesis codecs.

The optimum RPE codec [12] is a closed-loop analysis-by- synthesis scheme with perceptual error weighting, where most of the complexity arises from the complex excitation optimisation process. However, the 13 kbps GSM-RPE codec portrayed in Fig. 5 is an open-loop version of the optimum RPE with the objective of achieving more cost-efficient one-chip integration, higher yield and lower power consumption, This 13 kbps codec still outperforms the 16kbps SBC codec both subjectively and objectively at marginally higher complexity [13], [19].

May we remind at this stage that the perceptual error weighting degrades waveform representation (i.e., SEGSNR) for the sake of higher perceptual quality. At the higher end of the complexity scale reside the stochastic codecs, the CELP and SEV. Clearly, they represent a different league with their low SEGSNRs and low bitrates, yielding high computational complexity. Deploying special measures to enhance their robustness makes them ideally suited for hostile mobile applications, where the subscriber is prepared to sacrifice speech quality for high grade of mobility. When comparing the 7 kbps CELP1 [20] scheme with the 32 kbps ADM scheme, similar subjective qualities and robustness are achieved at four- or five-times lower bitrates, and an estimated twenty-fold complexity as well as considerably higher power consumption.

The 4.8 kbps CELP2 [22] and SEV [14] schemes are near-toll quality codecs for mobile communication, where channel capacity is at premium. These codecs have rather similar attributes with the CELP2 system being slightly more robust than the SEV, which makes it more suitable for fading mobile channels.

In summary, if the most cost- and power-efficient mobile hand-held portable phone is our goal and the five-fold band width requirement is not a strict limitation, ADM is a serious contender in the 1991 speech-coding 'beauty contest'. However, since bandwidth is always at premium as far as we telecommunications engineers are concerned and 1μm CMOS technology is available for civil applications, can we please relax and wait for the microelectronics practitioners to implement our favourite *7 kbps CELP codec?*

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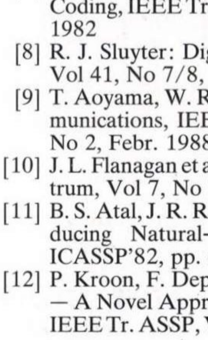

Lajos Hanzo

received his MSc and PhD degrees from the Technical University of Budapest in 1976 and 1982, respectively, and since 1976 he has been with the Telecommunications Research Institute in Budapest. Between 1976 and 1980 his main research interests included high-speed data transmission, digital filtering and modulation. From 1980 to 1981 he was on sabbatical leave working on parallel DSP modems at the University of Erlangen-Nürnberg in West Germany, while between 1981 and 1986 his research endeavours encompassed digital speech and data communications via satellites. In 1986 he joined the University of Southampton where he is teaching telecommunications. His recent research interest is the joint optimisation of source (speech, image, etc.) and channel coding as well as modulation schemes for dispersive fading mobile channels. Over the years he authored and co-authored some eighty scientific contributions and was awarded various distinctions.

WORLD BANK SUPPORT IN HUNGARIAN TELECOMMUNICATIONS INVESTMENT PROJECTS

1. INTRODUCTION

The World Bank supports provide not only financial resources for telecommunications development projects. The Bank prepared a complex system of proposals which could substantially affect the legal, economic and regulational environment of the telecommunications development and the internal operation of the Hungarian Telecommunications Company as well. As a result it could promote the economical telecommunications development and finally, modern Telecommunications in Hungary.

2. DEVELOPMENT OF HUNGARIAN TELECOMMUNICATIONS AND THE ROLE OF THE WORLD BANK

At the beginning of the mid-1980s the Government of Hungary, recognizing the importance of telecommunications to economic development, decided to give a higher priority and to increase investment substantially. As a result of the decision in the Seventh Five Year Plan (1986–1990), approximately 40 billion HUF have been spent which is roughly the double the amount in the Sixth Five Year Plan.

The strategy of this period has focused on the rehabilitation and expansion of the local and long distance cable and transmission infrastructure rather than on switching because it was felt that it was better to wait for the expected relaxation of COCOM control of digital equipment before making major investments in switching and improvement of transmission infrastructure.

This strategy has followed during this period although there was a consensus in Hungary that a major program for expansion, modernization and reform of the telecommunication sector was needed in order to keep pace with Hungary's effort to catch up with the rest of Europe.

2.1 The Government of Hungary's Telecommunications Development Program

The overall telecommunications development goal is to achieve a rough parity with the Western European average by about 2010. Such a program must address all three segment of the sector: services, manufacturing, and the user community. Recognizing the need for a comprehensive sectoral approach, the Hungarian PTT prepared a Concept Paper of Telecommunications Development from 1991–2000 for the Government. In June 1988, the government accepted the Concept Paper as a basis for further work.

The proposed 10-Year Plan is ambitious. The percentage of GDP devoted to telecommunications development would rise from the 0.3% in the early 1980's to over 2.0% in the 1990's. The rate of growth in direct exchange telephone lines would rise from 3.8% p.a. in the early 1980's to 11.3% p.a. in the 1990's. According to the experience of several percentage of their GDP in telecommunications and have achieved similar or grater rates of growth, while the plan is ambitious, it certainly feasible.

2.2 Previous World Bank Activities and Experience

The Bank has been involved in Hungarian telecommunications development from 1985 and had made a loan for First Telecommunications Project in 1987. The main objective of the project, scheduled to be implemented during 1987–1991, was to support Hungary's program of reform and development of the telecommunications sector by increasing the financial and administrative autonomy, flexibility and efficiency of the Hungarian Post, as well as providing modest expansion of the network and technical assistance for finance and technical matters. The project has been sucessful so far, the physical targets have been slightly overachieved, technical assistance was very effective. Institutional goals have been met both regulatory, authority and postal operations have been separated from telecom operations. The Hungarian Telecommunication Company as a commercial company was reconstituted at the 1st of January, 1990. Further institutional improvements are called for and expected during the Second Project.

The other substantial work performed in the sector was an Infrastructure review, with a separate telecommunications section. The main benefit was an objective valuation of the difficult situation in the Hungarian telecommunications manufacturing industry and the recommendations. For practical reasons, assistance to the industry is being channeled through the Second Industrial Restructuring and the First Technology Development Project.

2.3 The Bank's Strategy

The Bank has consistently supported the reformulation of the telecommunication policy in Hungary both at the enterprise and at the government level.

At the enterprise level, the policy focuses on the faster development and modernization of telecommunications system, the increased commercialization of the Hungarian Telecommunications Company, and the self financing of telecommunications investment on commercial terms.

At the government level, the policy emphasize the separation of operations from regulatory functions, the definition and legalization of regulatory regime, and the orderly introduction of competition.

From these strategic objectives, we could deduce the most important policy aims of the World Bank for the 1991–1993 three year period, and specify the proposed project as:

- continuation of the ongoing reorganization of the Hungarian Telecommunications Company's activities into an efficient, independent commercial company;
- digitalization of the telecommunications network;
- serving business subscribers by improving traditional services and rapidly introducing new services to promote term contributions to the economy' supply response;
- restructuring of the telecommunications manufacturing sector into a modern, competitive industry;
- introducing new entrants into selected telecommunications markets, together with the development of a regulatory regime to manage this process;
- preparation and introduction of outside finance and managment expertise into the new company.

2.4 Project Objectives

The Second Telecommunications Development Project has two principal objectives:
The project focuses on improvement and expansion of the telecommunications network and services. In this sense it is very important to build a new long distance digital network, expand international communication capability, connect the majority of business subscribers to these new facilities.

The other most important aim of the project is the institutional improvement of Hungarian Telecommunications Company designed to increase efficiency, quality of service and commercial orientation. The institutional strengthening will be realized through human resource management and development, measures to increase quality of service, measures to increase efficiency and productivity, and technical assistance to support this process.

3. FINANCING OF THE PROJECT

Total project cost for 1991–1993 is estimated around 84 billion HUF with a foreign exchange component approximately 22 billion HUF. These costs are estimated on 1990 price basis.

The proposed sources of financing are the internal resources of the Hungarian Telecommunications Company in 54% of total costs, loans from The World Bank and the European Investment Bank in a share of approximately 19% of total project cost, and other external resources in a share of 27%.

4. PROCUREMENT

The World Bank loan would finance the purchase of equipment, materials and use of associated services. It would be procured through International Competitive Bidding in accordance with the Bank's procurement guidelines. Participation of Hungarian suppliers in the bidding is expected to be limited since equipment and supplies to be procured under the loan are generally not yet manufactured in Hungary or not manufactured in sufficient quantity.

Procurement of items financed by European Investment Bank would be in accordance with EIB's procurement regulations which are acceptable to the World Bank. Import items not financed by the Word Bank would be procured by the Hungarian Telecommunication Company under International Competitive Bidding.

5. PROJECT IMPLEMENTATION

The Hungarian Telecommunications Company will be responsible for overall project implementation.

In order to improve cost effectiveness, especially at a time when the network is fast developing, investment cost per Direct Exchange Line must be very closely controlled. As material and equipment costs constitute a substantial part of investment, competitive procurement is essential to keep costs down. During the First Telecommunications Project, domestic procurement was mainly from monopoly suppliers without competition. But during the Second Project, a limited competitive environment is gradually coming into place. The Hungarian Telecommunications Company believes, and the World Bank supports the idea, that the gradual introduction of a competitive purchasing process will benefit not only the company but the suppliers as well. However, the plans for more aggressive development, together with elimination of the state support will necessitate still more efficient and cost conscious mechanisms.

6. PERFORMANCE INDICATORS

A series of performance indicators was defined for monitoring the execution of the project including development indications (connected Direct Exchange Line, switching capacity), quality of service indicators (completion ratio, fault incidence,...), efficiency indicators (staff per 100 DELs, cost and revenue per DELs,...), financial performance indicators (internal cash generation, debt service coverage, rate of return,...).

On the basis of these indicators, the Hungarian Telecommunications Company has to provide semi-annual reports to the World Bank in which the company has to analyze its performance.

6.1 Future Financial Performance

As a result of the Second Telecommunications Development Project, the economic performance of the company is expected to improve. The operating revenue is projected to increase by a yearly average 17% while operating expenses are projected to increase at rate of 13%. So the operating ratio is projected to decrease from 55% in 1990 to 46% in 1993.

Net profit after taxes, as a percentage of operating revenue, is projected to increase from 27% to 36% during this period. The rate of return on net fixed assets is projected to increase from 21% in 1990 to 26% in 1993.

Although the level of debt financing of the investment program is rather high, the self financing ratio is projected to increase from 60.0% in 1990 to 62.7% in 1993.

7. CONCLUSION

The World Bank supports provide not only financial resources for telecommunications development projects. The Bank prepared a complex system of proposals which could substantially affect the legal, economic and regulational environment of the telecommunications development and the internal operation of the Hungarian Telecommunications Company as well.

As a result, it could promote the economical telecommunications development and finally, modern Telecommunications in Hungary.

PETER KIS
HUNGARIAN TELECOMMUNICATIONS COMPANY

WORKSHOP ON CELLULAR NEURAL NETWORKS AND APPLICATIONS

The first IEEE International workshop on cellular neural networks — CNNA—90 was held in Budapest in December 1990, with the subtitle "A new multidimensional approach in "neural" computing technology with real-time image processing applications." It was a surprise, even for the organizers, how many excellent papers were presented in this brand-new area.

The first positive response came from the European Sections of IEEE Region 8 when the idea was first introduced during their late 1989 meeting in Vienna. In addition to the IEEE Region 8, the following Sections were co-sponsoring the Workshop: IEEE Austria, BENELUX, Germany, Central and South Italy, North Italy, Poland, Portugal and UKRI. (United Kingdom and Republic of Ireland). It was organized by the Hungarian Section of the IEEE and the Computer and Automation Institute of the Hungarian Academy of Sciences.

About 80 persons attended the workshop including internationally reknown scientists and many Ph.D. students from about a dozen countries, including Unites States, Japan and many European countries.

Let us briefly describe the technical area and summarize the main events.

Since a few years, computing arrays with uniform nonlinear analog processors placed on a regular grid have become more and more important. Some of these structures are called neural networks. Their programmability is achieved by changing the interconnections between these processors. Hence, they are also called connectionist models. In general cases, all the processors are connected to all the others. In some cases, the processors are merely capacitors, their voltage/charge representing the information while the interconnections are being realized by resistors. These so-called resistive grids are used also in some picture processing applications.

In image processing applications, it is quite reasonable that the picture elements (pixels) should be in a one-to-one correspondence with the analog processors. In fact, they can be considered as analog nonlinear cellular automata. In an entirely general setting, this paradigm has been called "cellular neural network" or CNN (L.O.Chua and L. Yang, IEEE Trans. CAS, Vol. 35, pp. 1257—1290 (1988)). Hence, in the following, when saying CNN, it will mean an analog nonlinear processing array with locally connected processors placed on a regular 2D grid (layer). Even multilayer CNN-s are of special interest. A crucial aspect of the CNN is its local interconnectivity and the two independent inputs (here, the initial state is also an input information).

Artificial neural networks are of wide-spread interest. The cellular neural network paradigm is emerging as an important area, partly as a means to solve new problems, partly to become a solid theoretical framework for locally connected analog processing arrays. These models are also useful in modelling living organisms e.g. the retina.

The aim of the organizers of this Workshop was twofold: to provide a forum for the exchange of ideas of those working in this field and sharing their hand-to-hand experiences, and to organize a one-day tutorial session with live demonstrations for those not yet active in this new research area but wishing to get aquainted with it. Thus, the large number of Ph.D. students is an asset for the future.

It was a very pleasent surprise for the organizers that even at this early stage of this new research trend, more that 50 papers have been submitted for reviewing. The papers which have been accepted and presented contain interesting new results.

The main sections of the Workshop were as follows.

- Tutorial and demonstrations
- Theory
- Applications
- Hardware-software design simulation and testing tools
- VLSI realizations
- Related other topics

All papers, except the tutorials, are contained in the Proceedings which is available through the IEEE Press or through the Organizer Institute. Let us mention only some areas which were considered by several important papars.

The CNN paradigm was extended by including nonlinear and delaytype templates. Various qualitative properties including multiple solutions were considered. Learning algorithms for determining the templates, comparison with 2D digital filtering as well as application areas like motion analysis, character recognition (including Japaneese characters), small object counting, special types of image processing were key areas of applications. The design questions of testing systems, simulators, a hardware accelerator board, and various VLSI realizations (including programmable ones) proved the maturity of this new technology in high speed applications. The impressive first optical realization experiments as well as cortex-like architecture analysis highlighted the new directions to be explored.

The tutorials by invited scientists covered various topics related directly or indirectly to the CNN paradigm. This first session and the last round table discussion (moderated by Professor Leon O.Chua) showed the tremendous research and application potentials of this new area.

Special thanks are due to the members of the International Scientific Committee who prepared the reviewing on a very tight time schedule. The excellent work and dedication of the secretaries, our students and coworkers as well as the generous help of the Region 8 Committee of the IEEE are gratefully acknowledged.

TAMÁS ROSKA
CHAIRMAN OF THE ORGANIZING COMMITTEE

TEKTRONIX TO PROVIDE BETTER SERVICES FOR THE HUNGARIAN MARKET

At a recent press conference in Budapest, three contracts have been signed on the distribution of Tek products in Hungary by Mr. Günther Graf, general manager of the Vienna Tektronix Office.

Tektronix is one of the leading manufacturer of test, measurement and communications equipment, graphical devices and computer peripherals worldwide. Founded in 1946, Tektronix has grown to a company with more than 12 thousand employees and sales of $1.4 billion in the last fiscal year.

After introducing the first triggered oscilloscope in 1947, Tektronix now owns a number of industrial patents realized in high frequency power supplies, oscilloscopes triggered by a television signal, plug-in and vector scopes, general purpose spectrum analyzers and hand-held storage oscilloscopes. The reliability track of some three thousand Tek products is among the best in the industry.

Tektronix products have been present in Hungary since the early fifties, and a service department has been operating for fourteen years in Budapest. From 1983 to 1988, all sales activities in Hungary were handled by the Vienna Office Tektronix Ges.m.b.H. Since 1988, a Tektronix Representation in Budapest has been dealing with the requirements and problems of Hungarian end-users. The scope of the Representation — headed by Mr. Ágoston Temesi — covers technical support, application advice, price information, warranty claims, etc.

During the past few months, COCOM export regulations have been drastically eased, with a great number of commodities becoming licence free. As a consequence of these changes and the consolidated political and economic situation, sales to Eastern and Central European countries are expected to grow rapidly.

In order to provide the same services for the developing Hungarian market as for the Austrian one, three Hungarian companies had been chosen as distributors of Tektronix products. Metratek Llc (1149 Budapest, Varga Gyula park 7—9) will be the distributor of test, measurement and communications products while Jura Llc (1066 Budapest, Podmaniczky utca 20) and SZKI Recognita Ltd (1015 Budapest, Donáti utca 35—45) will cover the market with high quality Tek colour thermal-wax and ink-jet printers for the Hungarian MacIntosh and PC world, respectively. The three companies will provide demonstration equipment, direct sales and full service for all Tek equipment used by Hungarian customers. ■

Informations for authors

JOURNAL ON COMMUNICATIONS is published monthly, alternately in English and Hungarian. In each issue a significant topic is covered by selected comprehensive papers.

Other contributions may be included in the following sections:

- INDIVIDUAL PAPERS for contributions outside the focus of the issue,
- PRODUCTS-SERVICES for papers on manufactured devices, equipments and software products,
- BUSINESS-RESEARCH-EDUCATION for contributions dealing with economic relations, research and development trends and engineering education,
- NEWS-EVENTS for reports on events related to electronics and communications,
- VIEWS-OPINIONS for comments expressed by readers of the journal.

Manuscripts should be submitted in two copies to the Editor (see inside front cover). Papers should have a length of up to 30 double-spaced typewritten pages (counting each figure as one page). Each paper must include a 100—200 word abstract at the head of the manuscript. Papers should be accompanied by brief biographies and clear, glossy photographs of the authors.

Contributions for the PRODUCTS-SERVICES and BUSINESS-RESEARCH-EDUCATION sections should be limited to 16 doublespaced typewritten pages.

Original illustrations should be submitted along the manuscript. All line drawings should be prepared on a white background in black ink. Lettering on drawings should be large enough to be readily legible when the drawing is reduced to one- or two-column width. On figures capital lettering should be used. Photographs should be used sparingly. All photographs must be glossy prints. Figure captions should be typed on a separate sheet.

For contributions in the PRODUCTS-SERVICES section, a USD 110 page charge will be requested from the author's company or institution.

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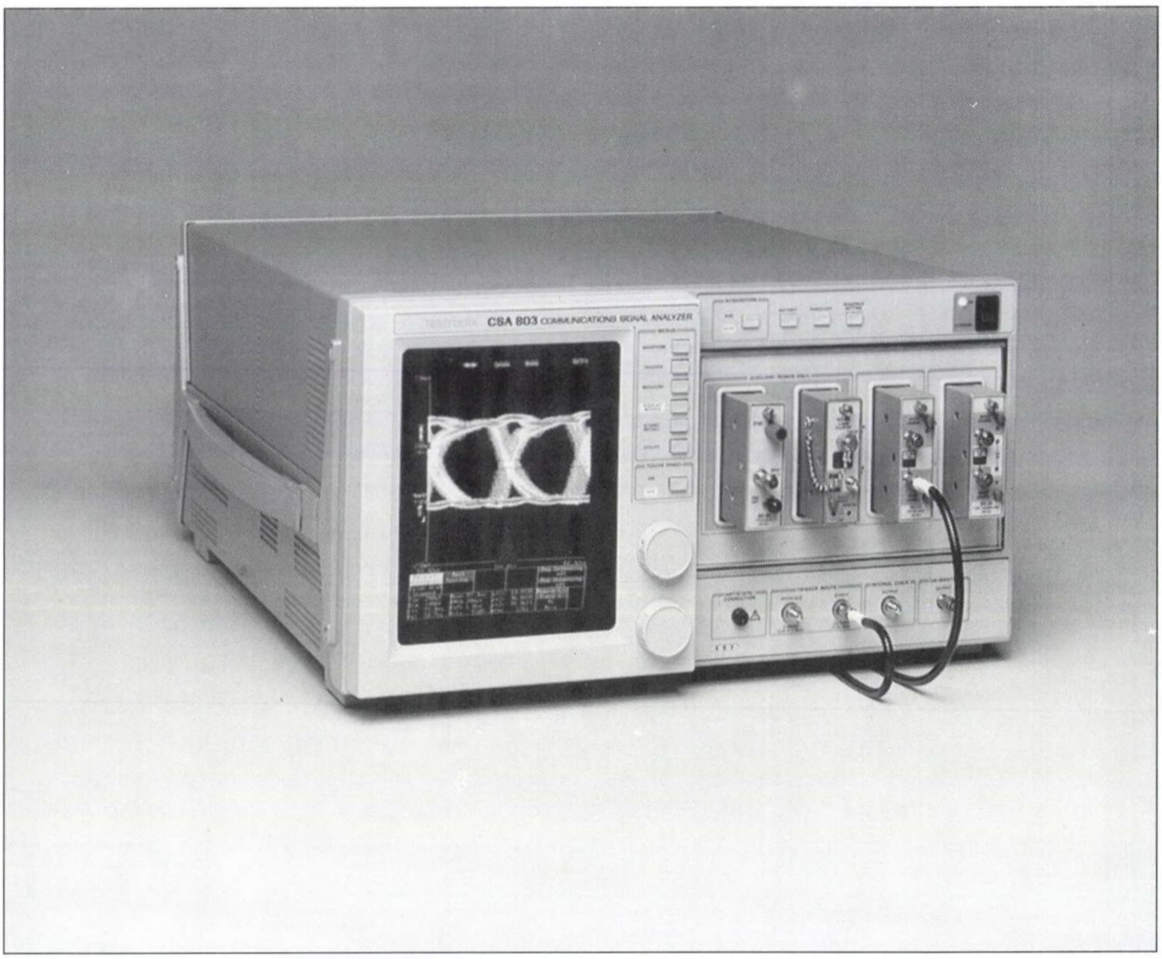

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Three-Year Development Program of Hungarian Telecommunications

Our Company-according to its mission and basic goals-is working to ensure that within the shortest possible time, telecommunications becomes generally available in the Hungarian economy.

As the only provider of basic telecommunications services in Hungary, in accordance with the Telecommunications Law now under preparation, we have started preparations for cooperation with future partners in the competition to come. To accomplish dynamic development in telecommunications, our Company has prepared a three-year development program which contains following major strategic principles:

- development on a commercial basis
- attracting local, regional and foreign capital
- separating many telecommunications services (payphone and PABX services as well as packet switched data transmission) from the basic services, and development of these in a competitive enterprising form
- entering the market of personal paging systems and cable television
- acquiring dominant market shares during the development. Our goal in the practical realization of the above principles is to develop the infrastructure of a regional telecommunications network, in this way providing basis for local initiatives, and ensuring the possibility of applying following market oriented services:
- establishing the digital backbone network that would connect 56 nodes in the country and would be practically unlimited in its capacity of traffic performance
- installing digital exchanges with a complete service coverage in the 19 county towns to satisfy foreseeable demands (in terms of exchanges to be connected, facilities to be used and traffic to be handled)
- significantly expanding international and national telephone exchanges in Budapest
- establishing the digital change-over network in Budapest and installing two large central exchanges.

Based on the above project the previously planned increase in the number of subscribers can be reached, and the Hungarian Telecommunications Company and its competitors will able to connect 800 thousand new subscribers.

Given proper enterprise-spirit and capital investment, completely automated, modern telecommunications services can be developed in the entire country within 4–5 years.

Utilizing the backbone network soon available everywhere, data and text-communication services will be developed, based on enterprises. We will also undertake developing areas and different new services (VSAT, personal paging system, etc.) in cooperation with foreign partners.

In order to achieve goals and objectives more efficiently, our company will change into a joint stock company according to the World Bank loan contract.

The strategic goal of privatization is capable of satisfying a large demand with the help of an organisation that is service oriented and efficient. Privatization will also contribute to our Company becoming uncreasingly competitive, renewing its organization and developing its management methods.

HUNGARIAN TELECOMMUNICATIONS COMPANY