JOURNAL ON

COMMUNICATIONS

VOLUME XLVII DECEMBER 1996

RADIO BROADCASTING

JOURNAL ON COMMUNICATIONS

A PUBLICATION OF THE SCIENTIFIC SOCIETY FOR TELECOMMUNICATIONS, HUNGARY

Editorial office
Budapest I. Ostrom u. 23-25.
H-1525 Budapest. P.O. Box 75.
Phone: (361) 175-7777, (361) 201-7471
Fax: (361) 156-5520, (361) 201-7471

Subscription rates
Hungarian subscribers
1 year, 12 issues 8100 HUF, single copies 700 HUF
Hungarian individual subscribers
1 year, 12 issues 1000 HUF, single copies 120 HUF
Foreign subscribers
12 issues 150 USD, 6 English issues 90 USD, single copies 24 USD
Transfer should be made to the Hungarian Foreign Trade Bank, Budapest, H-1821, A/C No. MKKB 203-21411

JOURNAL ON COMMUNICATIONS is published monthly, alternately in English and Hungarian by TYPOTEX Ltd. H-1015 Bp. Retek u. 33-35. Phone/Fax: (361) 115-1759. Publisher: Zsuzsa Votisky. Type-setting by TYPOTEX Ltd. Printed by Dabasi Jegyzetnyomda, Dabas, Hungary HUISSN 0866-5583

EDITORIAL

If in the last third of the 15th century somebody would have predicted that at the end of the 20th century half a million printing houses will be working in Europe, the scholars being in contact with the written erudition would have had serious doubts. Who could put into words this legion of writings? – could they have said – namely the work time needed to produce the manuscript of a book is measurable in years, while the printing house produces many copies in a few days only. But even if there would be enough material to print, who would read all that, what these numerous printing houses would produce? Let us do some calculations: half a million printing offices publish yearly at least five million books, if we take one thousand volumes per book, this means five billion volumes to be read yearly by Europeans. The population of Europe is nearly 700 million people, that is 7 volumes per capita – from babies to graybeards. And of course many reading material is printed in other forms: newspapers, info-sheets, manuals, forms, announcements, etc. And, in addition to that, one printed material is usually read not only by a single reader.

Even if we suppose that everybody is lettered, and seizes eagerly everything, what is printed, it is impossible to read such a mass of written material. Considering the economic side of the problem the situation is similarly paradox: if there are unreadable books which can not be sold in quantity, either in the price of each book one has to pay for those going to the paper-mills or the printing houses will be losing money and will go bankrupt.

What is reality in contrast to the opinion above? The printing industry is flourishing, and we, the readers, tolerate quite well the "consummation" of only a negligible fraction of all that, what is printed.

Something similar happens today due to the amalgamation of electronics, telecommunications, broadcasting, computer and information technology. An incredible amount of information appears daily, and is obtruded upon us, consumers. The newspapers, radio, television, computer networks are flooding information to us and in addition to that information is disseminated in various secondary forms: video and audio cassettes, CDs, and CD-ROMs, one can buy books about films and even information on information itself has a vast amount of literature. Experts speak about information revolution and say, that the digitalization and the computer gave birth to processes which will transform the entire society, the whole human life. The large amount of information and the easy access to any kind of knowledge, will change totally the contacts and the communication between individuals, and the society on one hand and among groups on the other.

Let us calculate again. From one single satellite it is possible to transmit up to 120 digital television and about 400 radio programs. To cover Europe 60 to 90 satellites could be launched, i.e. Europe would be covered by about 10,000 TV programs and more thatn 30,000 radio programs. This would mean 240,000 TV-hours and 720,000 of radio-hours daily. The population of Europe is 700 million people. If all citizens of Europe – from babies to graybeards – watch 3 hours TV, and listen 1 hour radio daily – all different programs – than about 8,750 TV viewers and 972 radio listeners would "consume" the same program. Obviously it is an absolute nonsense to transmit programs for such a small number of viewers and listeners. If more people receive some of the programs, than other programs will become unnecessary. And we considered satellite-broadcasting only! Tremendous number of small and large terrestrial radio and TV-stations transmit programs, we experience the rapid growth of the computer networks, and one can buy recorded video and audio material. Who will fill all this media with program in an acceptable quality? And how will the economy react, how will the information market operate? Is this not a similar situation as that of the printing at the 15th century?

This issue of the Journal on Communications deals with the perspectives of Hungarian radio broadcasting. Our aim is to present the factors influencing the development and the future of the radio. Hungarian radio broadcasting has a very rich history. A Hungarian, Tivadar Puskás started in 1893 the "Telephone News Dispenser" the invention, which was in fact the first broadcasting in the World. Not the technical solution, the usage of the contemporary telephone technology was the ingenious in this invention, but the idea to deliver a continuous program to a number of "consumers", to transmit news, music, poems and jokes, as we do today. The continuation was also valuable, three years after the pioneering BBC, in 1925 the Hungarian regular medium wave service was started, and undoubtedly a new era has started already. From December 1995 the Hungarian Radio transmits its three main programs parallel in DAB.

Considering the possibilities for information delivery, and the immense quantity of it, the engineer has to live with the fact, that the contents of the information is almost independent of him, and on the other hand, if the contents makes the information not worthy for transmission, our efforts are fruitless.

I am firmly convinced, that human intelligence is strong enough find and create the value and the beauty. Our task is to produce the most adequate and advanced technology delivering for the beauty and the values of life to mankind.

J. RADNAI

Jenő M. Radnai graduated in electrical engineering at the Technical University of Budapest in 1961. He started his career in the Orion Radio Co. He has been working at the Hungarian Radio since 1963, in 1992 he became technical director. In 1985 he was delegated to OIRT (Prague) as Chief Engineer for six years. He is member of HTE, AES, Fellow of IEE.

MAN–MACHINE RELATION IN RADIO BROADCASTING*

G. HECKENAST
HUNGARIAN RADIO
H-1088 BUDAPEST, BRÓDY SÁNDOR U. 5-7.

The paper reviews the relation between man and machine in the process of radio broadcasting. It is shown that corresponding to development trends of technology, computers emerge in all sections of radio broadcasting serving as the means of human intervention. However creativity remains essential, in the software preparation and also in the utilization of up-to-date equipment.

1. INTRODUCTION

I am sitting at my radio set — what a rare thing nowadays — listening to music streaming from the loudspeakers. It is a good receiver, so is the program, it is like having the orchestral stage in front of me, I almost see the string and the wind sections and the suggestive movements of the conductor. But this is only a virtual orchestra, artificially produced by machines, and to experience the real orchestra a chain of machines should be connected. Even if not structures full of pistons, valves, clattering-.rattling moving masses of parts, still machines, electronic machines. Along this long chain people are working by all (or at least most) machines. I wonder what the relation is between man and machine at the links of this sophisticated chain.

Machines were invented to ease human labour. In ancient times, labour as an activity serving men, human life and society was merely based on biological energies, what had been beyond human performance was left uncompleted. Man was the means of production, products came into being as results of human physical-biological energies. However this way of production was very tiring and also quite inefficient, tools and simple machines were created, natural forces, first of all animal power were started to be used. Man still utilized his own physical power, but relieved the burden of work through self-developed tools. The first industrial revolution, mainly the invention of the steam engine brought major changes. By that time machines had become the major tools of production, man, instead of dealing directly with products, controlled machines; machines stepped in between product and producer. Machines had become more and more developed, tasks of man handling machines became easier at least regarding physical labour. Controlling machines needed less power and more skills. Thus machines were given more intelligence to increase productivity and also professional skills were automated. Automatic, self-acting machines came into being. Role of man was limited only to inspection, he was responsible for the operation of machines, thus he had to serve machines.

The next revolutionary process has started with the penetration of computer technology into the field of production; machines are no longer controlled and monitored by man but computers. This is a new challenge for man, his main task have become to program computers, to elaborate software, thus besides machine – i.e. the hardware – the software, a new link has stepped in between product and man.

2. THE RADIO PROGRAM AS A PRODUCT

The radio program is a product, in particular it is an object of mass production. However, we have to differentiate the process of content production from that of the technical realization, since the content of a radio program is the outcome of intellectual work, and as such it cannot be discussed by rules of industrial production. However to bring this program to the listener i.e. to make the product consumable it has to go through many processes of production. (Generally content accomplishment of intellectual products — i.e. creation itself — does not need work in the physical sense, but the realization of its form does.)

A further speciality of the radio program is that it cannot be created without the use of machines, and other machines are needed for its consumption. Thus the production process is characterized by the rules of industrial production, but consumers, i.e. listeners can only get the product with the help of another machine, and that means another man-machine relation. But instead of production, this relation relies to consumption, therefore rules of production processes cannot apply.

3. THE RECEIVER-LISTENER RELATION

The receiver is the last link and it is directly connected to man. This relation would be ideal if the receiver served its user without any adjustment, and in the very way what the consumer had wished. The less care the receiver needs, the better!

Radio receivers always had to be handled, but in the course of development more and more solutions appeared to serve the comfort of the listener, that is the set was not to be touched constantly. To lessen or even eliminate the troubles occurring during reception several automation techniques were elaborated (e.g. AFC, AVC, AGC), and instead of frequencies of stations, boards with names of towns were introduced, and to help fine tuning cathode ray and other types of tuning indicators (meters, LEDs, etc.) were built in. With the spread of UHF radio broadcasting displays with city names were wiped out, since one frequency could belong only to one city in one service area, and receivers were manufactured in much bigger quantities that could have been sold in one area. Tuning disks or pointer displays were not easy to manipulate therefore digital displays emerged. Later on synthesizer technology has brought real comfort; the listener keys in the frequency of the station to be set and

* The paper was edited by J. Radnai based on discussions with the author.

the receiver tunes up automatically with quartz precision. The optimum solution is, when only the program or program type is to be put in and the receiver tunes up itself. Frequented stations can be pre-programed and set with pressing one key. Receiver switch-on time can also be programed, and also the duration of the program.

These features require the application of computer techniques in the receiver, which, on the other hand, makes possible to realize several further functions. How pleasant it were if loudness relation of speech and music could automatically be set, even if it is a subjective thing and depends on listening circumstances. The best solution is when the listener can adjust it to his own taste and to the actual acoustic conditions. This is what RDS with UHF broadcasting and in DAB, digital radio broadcasting provides for, where loudness of speech and music can arbitrarily be set with the help of encoded signals broadcast together with program parts. In ideal cases the sounding of an original event is transformed with high fidelity (tone, sound stage, stereo image, room, etc.), and it cannot be changed by the listener. However long transmission chains can cause distortions, and the acoustics of the room can sometimes prevent to reproduce the original sound flow, so good receivers should allow adjustable correctors to be switched on. This leads to quality receivers full of manipulating elements, mysterious notes and the average consumer hardly finds at first sight even where to switch it on. Man-machine relation is linked to a certain knowledge or at least demands some technical sense. One may have an expensive hi-fi receiver, but if it is not properly adjusted quality reception cannot be achieved and the owner will blame the program provider.

This made the receiver market diversified: simple sets, hardly needing any setting and wonder-sets resembling to cock-pit dashboards emerged. Computer technology has entered into both types and when they are set, practically a computer gets commands that in many times performs very sophisticated tasks — instead of us. Man-machine relation has been simplified to man-computer relation. A short sighted country woman never searchs for 67,04 MHz, but for her favourite program, and until she has difficulties to find it on the display of her small portable radio, she rather does not handle the receiver, and if she finally gets what she needs she will only use the on/off switch.

4. MEN AT THE RADIO TRANSMITTING STATION

The link preceding the receiver is the "ether", and it is not a machine. Radio waves get into the ether through machines, and there are machines that help to receive them. When studying man-machine relation we find that the next link following the receiver is the transmitting station. The input of the transmitter gets the complete program — the finished product. The role of the transmitter is not production but service provision: it has to get the product to a consumable form, to "pack" it, i.e. to transform it to modulated radio frequency signals then transport it to the user, that is to radiate it to the ether. The service is good if it does not involve the product, if the listener with the help of his receiver, can "unpack" (demodulate) the very same product the producer prepared. However transmitters should be protected from overloading, distortions of modulated link should be equalized, band limitation might be needed (mainly at short wave transmitters), and better intelligibility might also need some changes on the signal. In the case of digital transmitters signal compression is often task of the station, and it is also a means to achieve higher efficiency. These more and more sophisticated operations would hardly be possible to complete manually, left merely to the control of the human ear.

In the beginning transmitters were complicated equipment and needed constant control or at least inspection. Automation and computer technology have changed also this field and nowadays several unattended transmitters are in operation all around the world without any personnel. Those delicate operations influencing the lifetime of equipment or the safety of transmission originally managed by well educated technicians, today are based on algorithms and are computer controlled, therefore the role of software is increasing also in this field. So man creates the transmitter and the related software and then controls and checks his machine through a terminal.

5. STUDIO OPERATORS

We have arrived to the venue of program production, the studio house where the "product", the program is coming into being. Studio activities can generally be divided into two parts: program production and program providing. These two processes can be distinguished since the first is much similar to that of the works in record studios being not strongly connected to the radio. The latter can be done even if someone does not deal with program making, only buys and broadcasts it. In the beginning of the radio era, technicians working in the studios were carrying out both, operating very similar equipment for the two processes. It could not happen otherwise since there were only live programs, recording was not yet used. This work needed good sense of hearing, short reaction time and perfect knowledge of the equipment; the technician had to know the consequences of any interventions. All operations were manual and the knowledge and skill of the engineer were crucially important.

During recording real sound events had to be transformed into electronic signals in such a way that appropriate illusions should be raised when playing back the recording. Today we do not speak about "true fidelity transmission" because of technical limitations: bandwidth and basic noise level are given, and the impossibility of perfectly linear transmission and perfect transient reproduction. Features of the original sound have to be pressed into the transmission channel at least in such a way that the recording should provide as good an effect as that of the original sound event. The real sound field cannot be and is not worth to be transformed with today's means; recorded sound is such a product that gives the illusion of the original sound event but is not equal to it. Man always wanted to facilitate production by helping to ears with eyes (indicators, lamps), easing attention (by compandors and limiters) and by increasing the number of channels.

Recording technology used more and more sophisticated equipment, but it could not help sound engineers, who had to handle more and more manipulators simultaneously. At the high end of analog technology 40–50 channel mixers were used, manipulated by 2-3 sound engineers at the same time. Several functions were automated, sound engineers ruled less and less electronic signals, they tended to remind of a solo musician who knew which key made what effects, but did not care about the physical consequences of their movements.

Analog mixers with built in controlling electronics appeared and so computer technology has broken also into sound recording. When digital technology started to spread around, terminals for sound recording emerged and the greatest emphasis was put on software. Today operators do not have to understand physical processes of mixing, they have to communicate with the computer through special sound recording terms, computers translate, control and rationalize everything. The task left to the sound engineer is whatever is not possible to put into algorithms, that is creation together with actors, and to communicate artistic ideas with the computer.

Technology has released engineers from traditional technical work.

At program providing ready made materials are played in a prepared program order with introductions, announcements and small, easily automated corrections. Useless to mention that the whole process can be fully automated, human activities are limited also here to computer operations. Engineering creativity appears again in elaborating the software, machines can control program providing without supervision for days.

Program providing includes the selection and creation of signal path. In big radio houses many a hundreds of input and output lines are connected according to timetables. It is obvious that this task is also left to the computer. The good old manually handled, corded or plugged-in switchboards cannot provide for the solution of today's complicated activities. In some radio houses the length of the board in the switching room exceeded 28 m, the 8 operators were unable to access the cords and to comprehend who did and what. Today this work is done comfortably by one operator sitting in front of a terminal, since a part of the switching is repeated regularly by the day or by the hour and the remaining part can well in advance be programed.

6. GENERAL TENDENCIES

Man-machine relation is more superficial, less intimate — in any ways: different today — from that of in the beginning. In older times elements and parts were visible and seizable. Operators were aware of functions of all elements. Today there are incomparably more parts and in the world of microns they are neither visible, nor touchable, if at all, only chips and black boxes. Operators have no knowledge on their modes of operation or functions. As a result, operators or users do not need to have any technical education.

Technical development of our era generates professional fields to merge. Telecommunications and radio technologies cannot be separated, neither of them can be pursued without computer knowledge. Software has become the key factor, but good software cannot be created without thorough knowledge of the actual field.

With digitization any kind of information can be handled with the same means, for a transmission channel it does not make any difference if the forwarded digital signals are coming from sound, pictures or written text encoded. These digital signals can be processed if appropriate software is available.

In radio broadcasting as well, terminals started to turn to be the only means of man–machine relation; the whole chain is becoming computerized. The less effort have to be made on the operation of equipment, the more attention can be paid to creative activities, activities unsuitable to turn into algorithms, and to concentrate on achieving optimal sound quality. To ensure that whoever sits in front of the loudspeakers could not only hear the music but could really sense the strings, the winds, the conductor and the atmosphere, and could participate and experience the event.

Gábor Heckenast started his career at the Lakihegy Radio Station of the Hungarian PTT in 1948. He received his degree from the electrical section at the Faculty of Mechanical Engineering of the Technical University of Budapest. From 1949 till 1963 he was working at the radio studio of the Hungarian Radio and Television in the field of magnetic sound recording. His activities resulted in the construction of several new equipment and patents. Having spent an eight year period at the television branch mainly as deputy technical director, in 1971 he was transferred back to the radio field and as technical director of the Hungarian Radio from de-merger in 1974 till his retire at the end of 1992. He chaired the technical committee of OIRT, has been serving several important positions at the local branches of IEC and the Audio Engineering Society and received the Fellowship Award of the latter in 1993. Mr. Heckenast is member of the Acoustics Complex Committee of the Hungarian Academy of Science, the Scientific Society for Optics, Acoustics and Film Techniques and the Scientific Society for Telecommunications. He received the State Award in 1978 and the Officer's Cross of the Republic Award in 1992. At present Mr. Heckenast is working as a consultant and is busy with writing books, studies and technical articles.

CHANGES IN RADIO BROADCASTING IN THE 90'S*

L. UJHÁZY

HUNGARIAN RADIO
H-1088 BUDAPEST, BRÓDY SÁNDOR U. 5-7.

The last one and a half decade of our century passes in the spirit of the digital "revolution" from the acoustics point of view. Even the broadcasting organizations cannot elude the mostly good, however in some cases exasperating effects of digitalization. Hungarian Radio has been always highly responsive to the technological changes during its 70 years history, thus when we investigate the problems of the acoustical present or future of our Radio, effect of the digital technology cannot be neglected. However, the changes taken place in the field of listening-in in general, the changing habits of listeners, which may rearrange the radio genres, have the same importance at least.

Digital recording has been used since from 1988 in the Hungarian Radio and due to the rather fast running-up today all the activities in the majority of the music field from recording through post-processing to archiving are carried out in digital form. The thinking on the future raises the first question important: how can the long time storage of these recordings be ensured? To tell the truth, DAT is essentially not a professional system, it became a successful studio device from a consumer product with all of its advantages and disadvantages. This is why we are not quiet if a program is available only on DAT cassette. The most important — first of all music and radio play — recordings are stored in two identical copies on writeable CD. Since in some cases also the CD can cause surprise, the third copy is on a Dolby analogue tape of high quality, which hopefully can be played even 100 years after.

The digital technology gives much more new opportunities in the field of than in the post-processing, even process of recording. Striving after a sterility or purity of some kind can be observed in the performing art, because the repeated technical or performing "freckles" of the recording can be disturbing for the listener of a CD. Presumably, commonness of listening-in radio recordings is lower as compared with the CD, thus some concessions can be made at editing. However, nowadays nobody knows in advance, when will be recorded an originally radio recording on CD, thus editing should be carried out with a quality corresponding to the CD recording. Frequency of the cutting points is varying, in the serious music one cut falls to every 25 seconds in average. If this is examined from viewpoint of the homogeneous, undiminished "mood" of the recording, in these cases we may speak about misuse of the digital technology. The same can be observed on CDs of industrial product: cutting points, i.e. sounding of the individual tones without transients, can be heard well by means of earphone. Nevertheless, Menuhin believed already in his book (The Music of Man) published in 1979 that in the serious music returns the practice of recording by themes, but without cuts within those "to avoid loss of the living current of the performance". Sorry to say, just this current disappears frequently due to the opportunities provided by the digital editing technique of today and the question arises on the sufficiency of the compensation given by the sterility.

Hungarian Radio could be kept away from debates connected with the general sounding quality of the digital recordings so far, however we follow the dispute related to the issue with special attention. According to the general opinion the future is of the digital technology, even if sometimes we listen the soft sounding of an old "valve" recording with tears in our eyes. (Author of this article prefers digital recording in case of percussion instruments and all pianissimos, however for E-strings of high stringed instruments he would choose the analogue transmission.) Nevertheless, this technique should develop further on towards the higher bit rate and sampling frequency, provided the progress is supported by the already wide-spread systems, (CD, e.g.). The probable progress has a component at least important, if not more important as the previous one: improvement of the general acoustical experience of the listener. Now let us look back a little to the development of stereo systems. When in 1881 in Paris (and then in 1882 in Budapest) the first experimental stereo transmission was realized, idea of stereo presented itself too early. As long as the listener putting two rubber hose or receiver to his ears could not distinguish the bassoon from the saxophone, not the direction of the sound source was interesting, but that what sounds at all. Essential changes were necessary for spreading of the two-channel technique, as the microgroove record or the VHF broadcasting. The four-channel quadrophone experiments started in the early 70's also in the Hungarian Radio. Many 1 inch four-band tapes preserve the experiments of the 70's and 80's: entire radio plays, church music recordings, noise recordings, etc. From time to time we transmitted quadrophone programs even in the 90's by temporary linking up two stereo transmitter networks, e.g. in the last year a jazz concert from the studio and a church concert specially organized for this occasion from choral works written for 2 and 3 choirs. Both the quadrophone sound recordings and the four-channel transmissions proved that similarly to the mono-stereo change of experience an essential jump in sound technology is needed again for the change from stereo to quadro (or any multichannel) system. As in 1881 the idea of stereo came too early according to the sound transmission technique of that time, quadrophone "thinking" in the entire sound space of the 70's was too early (however not useless) in the same way. Such a big technological change as the analogue-digital transition should provide a change in

* Meditations of a sound engineer

sound experience to the listener much greater than the CD has no needle scratch and the themes can be found easier on the CD as compared to the empty grooves of the gramophone record. Opportunities provided by the digital technique deserve much more powerful efforts for heighten the listener's space experience, and this is valid for the radio broadcasting organizations as well. Although records conjure music to our homes, such important "genres" as living concert transmissions, radio plays remain always radio genres and transmission of those may provide especially great experience.

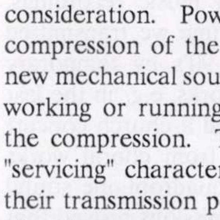
Two troops are fighting in the sound technology nowadays. On one side acoustical engineers make great efforts to provide sound as beautiful and realistic as possible to listeners and engineers dealing with information transmission on the other side endeavour to forward as much information through a given transmission system as possible during a unit of time. The result of this latter effort — among others — is the data reduction from which the Hungarian radio could be kept away so far and we hope that this will be unchanged in the field of artistic programs. (At the same time it is doubtless that sometimes data reduction is a physical or economic constraint.) Nevertheless the sophisticated technology of making radio programs involves the danger of occasional music transmission in a digital voice channel and the recorded version of the transmission — getting into the archives — may be published on CD after some years. And here arises the unavoidable question: in an era to come, when picture, sound and all other information are collected in a unified medium, will we able to preserve the quality of sound? Moreover it is thought-provoking that while all the earlier novelties of the sound transmission have been advertised with more beautiful, more realistic transmission, nowadays, in case of advertising DAB the undisturbed car-radio listening and information surplus are emphasized. Although DAB is essentially the response of the radio broadcasting organizations towards the CD technique, similarly to the relation between the VHF broadcasting and the microgroove gramophone record. This is why we have been informed with great pleasure on establishment of the ARA (Acoustic Renaissance for Audio) movement, what may mean simultaneously the renaissance of the exacting radio listening.

Change of radio listening habits cannot be left out of consideration. Powerful, sometimes aggressive dynamics compression of the commercial radios seems to create a new mechanical sound ideal. Background listening-in while working or running, in noisy environment really justifies the compression. This is also valid for the programs of "servicing" character of the public service radios, because their transmission period falls into the main working time. Thus background listening-in is an independent style from the acoustical point of view, however compression should be carried out rather in the studios to avoid damage of sounding aesthetics of the program as a whole.

Programs of magazine character have significant role in the program structure of the Hungarian Radio. Several of those have extremely high listening index, while the same can be ranged among the most sophisticated programs from acoustic point of view. Sound technicians of these programs are in the most defenseless position, because they get all the program components ready-made excepting the microphone of the program leader sitting in the studio. Since the reports are recorded and even edited in many cases, the sound quality is not determined in the magazine studios. Similarly, the quality of a telephone connected in the transmission is also questionable, since if the listener has a wrong microphone, the quality cannot be restored even by the best telephone hybrid.

Concerning the studio system engineering and the studio layout in general, a decisive breakthrough is needed in this genre, as instead of the earlier "classical" arrangement consisting of studio and technical rooms, magazines require a studio complex, consisting several rooms of different purpose. The central, auxiliary and preparatory rooms, the studio and the news commentator studios, telephone rooms and editing workstations should form a unified totality mutually not disturbing their work. Instead of the earlier view of separating the editorial rooms from the technical ones, integration of the same is practical regarding a certain program. By slow liquidation of the continuity room these joint studio groups will be suitable for the arrangement of the complete daily program flow. Contrasted with the commercial radio broadcasting, personality and creativity of the sound engineer will be significant factors in the public service radio broadcasting in the future too. Since in the present difficult situation of the public service radios the actual programs reacting rapidly upon the daily events have more and more important tasks, a flexible technical view following rapidly the demands is badly needed for executing such programs.

The third type of the public service broadcasting, the radio play has as great tradition in the Hungarian Radio. However, the future is all the more questionable: whether will good scenarios inspiring the sound engineers to the artistic re-creation be written and in our visual era will be able to allure the listeners to be tied to the loudspeakers? Leaving one of the genre of really radio character to its fate would be a considerable loss of the European culture! Simultaneously with the fast and great acoustical development, the aesthetic shallowness threatens us. This is, why all of us: professional technicians and non-professionals have to make efforts in order to preserve the position occupied by the public service radios in the European culture.



László Újházy graduated from Faculty of Electrical Engineering at the Technical University of Budapest in 1960 and from Bartók Béla conservatory in 1961. He has been sound engineer of the Hungarian Radio since 1960. He is founder member of the Camerata Hungarica antique music ensemble and of the Hungarian Society of Musical Sciences and Musical Criticism. Main fields of his research activity are studio system engineering, stereo and multichannel sound recording, history of musical instruments and acoustical problems of the performance practice of the antique music. For the time being he is Deputy Technical Director of the Hungarian Radio and professor of St. Stephen Conservatory of Musical Art, where he is teaching audio technique, acoustics and sound culture. In 1995 he has been decorated with the Small Cross of the Medal of Republic.

TRENDS IN STUDIO ACOUSTICS

É. ARATÓ-BORSI

HUNGARIAN RADIO
H-1088 BUDAPEST, BRÓDY SÁNDOR U. 5-7.

In the design of radio studios several aspects of acoustics have to be considered. With the development of technologies newer and newer requirements are to be met. New requirements mean new design and measuring methods, and these are in close relations with each other. For the improvement of sound quality the development of methods and approaches in room acoustics is inevitably necessary.

1. INTRODUCTION

The development of sound recording and broadcasting technologies inspires acoustics professionals to steadily reconsider acoustics requirements towards studios. Certain significant changes in technology can have specially strong impacts on the field, and this happened when digital technology emerged.

Demand and taste went through permanent changes during the past 70 years of radio broadcasting, but since the introduction of stereo technology the issue of sound quality has never been so much in focus as it is now in the era of digital recording.

It is certain that acoustic features of sound fields are considered to be more and more important, acoustics research works are in abundance.

If we look back to the history of room acoustics, we see some periods of fast development, but also years and decades passed with hardly any changes in methods and approaches in this field.

The acoustics behaviour of a closed field is so complicated that it is a very difficult task to design and calculate the processes of sound events, even with the help of up-to-date technical tools.

2. SOUND EVENTS AS TIME FUNCTIONS

At the beginning of this century scientists already found out that sound events (growth and decay) as functions of time have a strong effect on the forming of acoustic judgement.

2.1. Reverberation time

The importance of reverberation time was realized first. It was the first room parameter defined. Observations have proved that our sensations on sound phenomena in rooms are, among others, governed by reverberation. The wording of exact definition, deduction of the formula for calculation, and elaboration of the measuring method were of great importance.

Many scientists have been engaged with this issue, including György Békésy. He made experiments to adjust reverberation time when he designed Studio 6. — already a historic monument today — of the Hungarian Radio. He wrote about his experiments in an article [1] published in 1936 as follows:

"... It seemed to be reasonable to study reverberation time referred to the whole extent of hearing and to find its optimal value for the complete sound area. The impact of sound damping as a function of frequency on the acoustics of a room was obvious in the case of the building of the Budapest studio, since the courtyard of the building provided significantly more beautiful and natural tone, even if strongly echoed, of speech and a musical performance specially introduced at the spot than the studio room itself.

The room kept its dullness even when all curtains had been rolled up thus making the reverberation time equal to that of the court at the frequency of 500 Hz, that is the reference for the reverberation time in literature..."

So Békésy proved the frequency dependence of reverberation time. This exploration lead to a more exact method in designing acoustic covers. At that time even if in a solely experimental way, materials and structures proper to absorb sounds of different frequencies were found.

Since Békésy's works several recommendations have been elaborated for the reverberation time of studios of different purposes, obviously of different values. Fig. 1 shows a set of characteristics being in use today by the Hungarian Radio, Fig. 2 shows reverberation time values recommended by the BBC [2].

At the start of building and designing studios and for a long time afterwards mostly short reverberation times were considered ideal, independently from purposes (naturally always matched to the volume of the room). It was obvious, since mainly for economical reasons, studios of such volumes that allowed long enough reverberation time were not possible to build. Smaller volume needs shorter reverberation time, avoiding the impact of improper reverberations through sound absorption. In case of studios for music natural fullness of sound should be complemented with artificial acoustics means.

Sound engineers expect acoustics designers to design studios of clear, airy sounds. The aim is to reduce the needs of artificial changes to be made on the original room acoustics. It is not by chance that the world's leading recording companies either build studios of enormous volume or chose outside locations, churches for their classical music recordings.

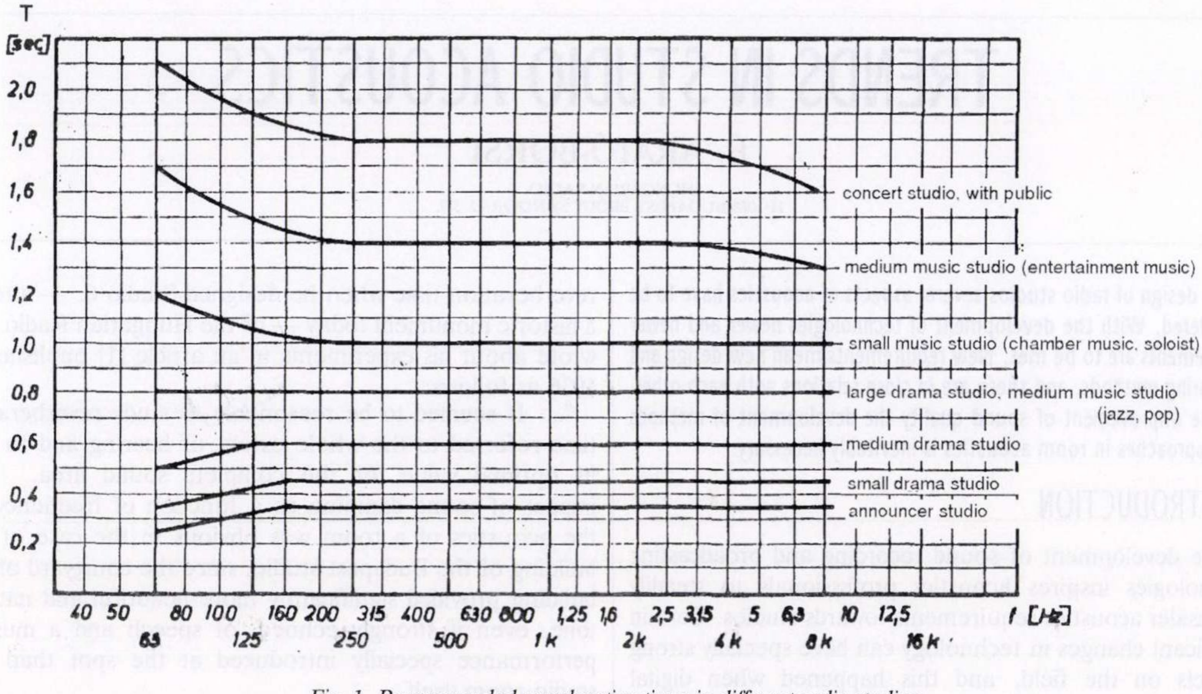


Fig. 1. Recommended reverberation times in different radio studios

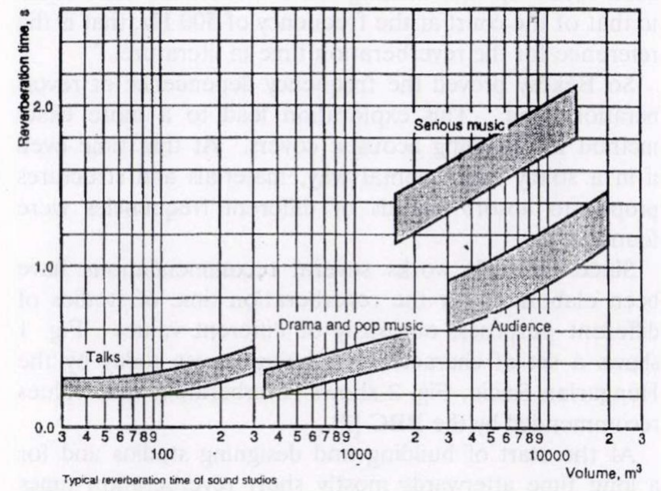


Fig. 2. BBC recommendations for reverberation times of radio studios

In certain types of studios, mainly for pop music, in recording technologies artificial reverberating techniques are generally used tools. In such cases also this condition is taken into consideration already during the period of planning by the designer of the studio.

Nevertheless we do not have to concentrate merely on music studios. In a studio for speech also to reach airy, clear sounds is the aim. In overdamped studios no beautiful sounds can be achieved.

This demand had lead to rediscover how important the diffusity of field it was. In the case of big concert halls to increase field diffusity has always been considered during design; while in smaller studios such considerations have rarely been taken into consideration, and if they were, only in a simplified form.

2.2. Diffusity

Basic designing methods of up-to-date diffusers were created by Schroeder [3]. Today's diffusers are designed with efficient diffusity in a wide band, and like sound absorbing structures, these diffusers are tuneable to several frequency bands.

With the application of diffusers the impact of bad reflections is not reduced through absorption but through dispersion of sound energy thus beneficially increasing time of sound decay. More natural and clear acoustics can be achieved this way. This is the direction present demands in room acoustics tend to.

The experiments taken place at the Hungarian Radio also have justified that if merely absorbing materials are used to avoid wrong reflections, the room would be overdamped. In an overdamped radio OB van with a few diffusers so much energy was re-sent to the field that a significant improvement in psycho-acoustic sense have been achieved. Figs. 3 and 4 show waterfall diagrams before and after the installation of diffusers. It can be seen that within a given time and frequency domain the energy increases and also distribution improves significantly.

Today's research concentrates on bringing diffusity calculations to perfection. Several calculation methods have been created, e.g. Jeans Holger Rindel's new approach in 1995 [4].

2.3. New parameters of room acoustics

Measuring technology has developed much with the application of digital signal processing. Advanced measuring methods allow to measure more room parameters and contribute to the development of designing methods.

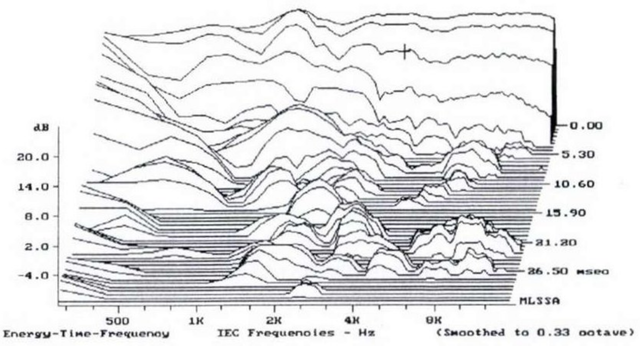


Fig. 3. Energy decay in an overdamped OB van

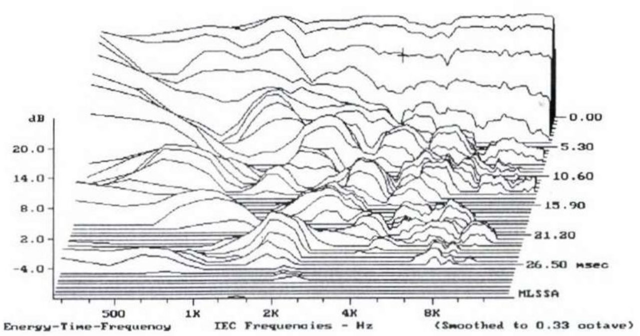


Fig. 4. Energy decay in the OB van of Fig. 3 after installing diffusers

The general use of pulse response measurements simplified the determination of energy quotient of time intervals thus calculation of new parameters became easier.

The concept of sound clarity was already introduced some time ago, but only recent development of measuring technologies have made it widely used. E.g. Gade gave the following definition for big halls [5]:

$$C = 10 * \log \frac{E(0, 80 \text{ ms})}{E(80, 1000 \text{ ms})} \tag{1}$$

where:

$$E(t_1, t_2) = \int_{t_1}^{t_2} h^2(t)dt \tag{2}$$

The same clarity parameters for smaller rooms can also be defined with applying other time intervals. E.g. the energy quotient coming in the first 20 msec can be related to the quotient of the 20–80 msec period, since these two time intervals are generally the time domains of the arrival of early reflections in smaller rooms.

In this case:

$$C = 10 * \log \frac{E(0,\ 20 \text{ ms})}{E(20,\ 80 \text{ ms})} \tag{3}$$

The new MLSSA measuring system of the Hungarian Radio enables us to determine sound clarity for the above mentioned time intervals from pulse responses measured in different types of studios. The collected data are compared to subjective evaluations. Our goal is to determine such values of C, that different studios can be declared as of good clarity with them.

3. REQUIREMENTS FOR CONTROL ROOMS IN RADIO STUDIOS

It is obvious when talking about sound recording and related sound quality we mainly concentrate on the acoustic parameters of the field of the studio.

However requirements of control rooms have also gone recently through great development.

Due to the usage of advanced recording tools and electroacoustic transducers we have to pay more attention to the field where sound engineers are working on sound recordings.

We have to admit that even if excellent loudspeakers are available, wrongly placed in a room of bad acoustics, the sound, that the sound engineer should hear without any distortion in tone and direction, will not be sensed properly.

To find acoustic parameters that can exactly be designed and calculated we had to make a lot of experiments. Psycho-acoustic experiments have lead to many results. Among them those are of great significance proving that the initial time period of sound decay is very important in forming the acoustic sensing. Though it has been known since long, only recent research revealed how very early sound reflections influence acoustic sensing. The 1996 AES award was given to a Danish colleague, Soeren Bech for activities in this field. His research work was carried on in the framework of the EUREKA program named "Archimedes" [6].

With our new measuring technique we can expressively show the sound decay in time. In the so-called waterfall diagrams strong, disturbing sound reflections can be found and if possible, be eliminated.

Five channel stereo recording technology provides a new field for research. Though not much experience and measurement data on listening rooms for these type of recordings are available so far, the recently completed Studio 24 of the Hungarian Radio fits to the requirements of five channel stereo listening.

4. ACCEPTABLE NOISE LEVEL

The notion that digital technology does need improved acoustics, has resulted in changing our approach to noise level.

The widening of the dynamics domain has brought about more severe requirements on sound recording studios. If we consider our own studios, our old, not "house in house" systems do not conform to the noise level criteria of digital technology.

Basically it is a critical and difficult task to define criteria since tightening of requirements leads to serious economic consequences. It is worth to review existing recommendations and standards on maximum noise level values accepted in different studios.

Fig. 5 shows the accepted noise level in music studios. Recommending organizations compared:

BBC United Kingdom
IBA Independent Broadcasting Authority, U.K.
ARD Germany
AIJ Japan
OIRT East European radio and TV organization

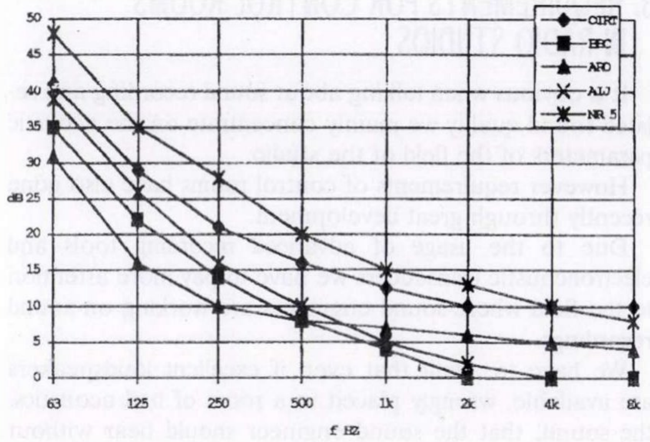


Fig. 5. Acceptable noise level in orchestral music studios [dB]

At the Hungarian Radio we were using OIRT recommendations up till recent past, and it is an interesting issue since the EBU has not yet elaborated any similar comprehensive set of recommendations.

NR curves are standard curves accepted in acoustics, the EBU makes efforts to spread their use.

One can think about the big variance of these criteria. Most likely this is due to different economic approaches. In studios of a one-man continuity center less strict noise level requirements should be accepted. Studios with several microphones, used in magazine type programmes should match more severe requirements. In any ways all criteria used so far should be reconsidered.

5. SUMMARY

Many aspects have to be considered during the acoustics design of radio studios. With the development of technologies newer and newer requirements should be met. New requirements mean new design and measuring methods, and all are in close relations with each other.

The improvement of sound quality needs to improve room acoustics methods and to change acoustics approaches.

REFERENCES

[1] Békésy, Gy.: "Sur la réverbération optimum des petites salles", *Revue d'Acoustique*, Vol. V. 1936.
[2] "Guide to Acoustic Practice", *BBC Engineering*, 1990.
[3] Schroeder, M. R.: "Number Theory in Science and Communication", Springer, 1984.
[4] Rindel, J. H.: "Diffusion of Sound in Rooms", *15th International Congress on Acoustics*, Trondheim, 1995.
[5] Gade, A. C.: "Prediction of Room Acoustical Parameters", *121th Meeting of Acoustical Soc. of America*, 1991.
[6] Bech, S.: *Final Report on Activities of EURECA project "Archimedes"*, 1995.

Éva Arató-Borsi graduated at the Faculty of Electrical Engineering of the Technical University of Budapest in 1975. At the end of the eighth semester she was admitted to study research and development engineering at TUB, and received her second diploma in 1976. She started her career at the acoustic department of the Construction Quality Control Institute. She has been working at the Hungarian Radio since 1978. Till 1992 Ms. Arató was a member of the development division, today she is head of the technology group of the operations division. Her main task is the acoustics design and measurement of radio rooms of acoustics nature and everyday problems in studios.

DAB – REGIONAL AND LOCAL

Z. VAJDA

HUNGARIAN RADIO
H-1088 BUDAPEST, BRÓDY SÁNDOR U. 5-7.

Though for Hungary the introduction of DAB on a larger scale comes only second after the transfer of the FM broadcasting from Band I to Band II, the formulation of some basic principles in connection with DAB and the examination of their practical consequences in the technical as well in the regulatory field is an actual task. We have at least try now to foresee on the one hand the development trends of radio broadcasting within the country and worldwide and, on the other hand, the possibilities of DAB known today and evolving in the next decade. By the start of more and more regular programmes and — hopefully — the considerable increase of the number of receivers and listeners all over the world, lots of experience will be accumulated which will inevitably bring about new solutions and changes in the coming years.

1. INTRODUCTION

One of the important questions not solved satisfactorily up to now is that of local or small area broadcasting using DAB technology. There is 'a frequently expressed fallacy that DAB is not appropriate for local radio' [1]. Is it really a fallacy or is there a bit of truth in this statement? The answer to this question could influence very much the conditions, the extent and the price of the introduction of DAB, not only in Hungary.

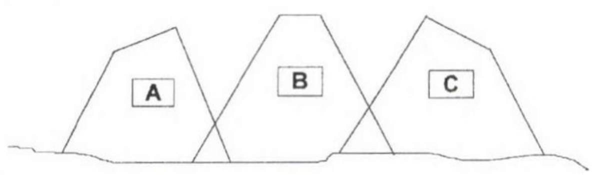
The main requirement for a local radio is individuality in terms of coverage and quality on one hand and economy on the other. FM satisfies this requirement optimally within the limits of the quality offered by the technology, within the limits of spectrum availability and within the inevitable restrictions of the regulatory environment.

DAB was designed to go beyond the quality limits of FM in a multipath mobile environment using low priced receivers. To achieve this the only feasible solution was a multicarrier, wide band transmission channel making multi-service usage a must, and the application of the guard band which in turn offers the single frequency network possibility. Only by exploiting these could the fundamental requirement of spectrum — and power — efficiency be met. However both the obligatory 'cohabitation' of services and the SFN seem to put insurmountable limits to the individuality required for local broadcasting.

There are, however, solution proposals based upon different philosophies by which local broadcasting could be realized meeting the requirements of the standard ETS 300 401 and of the draft of the second edition of this standard [2], [3] retaining the individuality of the participants in a similar extent as they have it now using FM technology.

2. THE CONSERVATIVE SOLUTION [1]

The starting point of the conservative solution is the rule that within an SFN the signals emitted by the different transmitters belonging to the SFN have to be exactly the same. If so, individual local coverages can be realized only using the methods of the FM network planning together with the exploitation of the multi-service capabilities of DAB. An example [1] in Fig. 1 helps to explain the principle. In the Figure three areas are shown which has to be covered by nine services of different coverage requirements. Four programmes have to be present in all three, one in two and three in single areas each. The multiplexes are configured for six 192 kbits/s stereo programmes.



Area (block)	Audio channel					
	1	2	3	4	5	6
A	g	e	a	b	c	d
B	h	f	a	b	c	d
C	i	f	a	b	c	d

Fig. 1. Coverage of three areas with nine services of different coverage requirements

As different services are required for different areas the only solution possible is to use different DAB blocks, one for each area. The three single area services are transmitted separately in the three blocks, the two area coverage is realized by using a channel in two and the three area coverage by using a channel in all three blocks for the same service. (In the example the number of areas, i.e. the number of blocks and the number of the services required are matched, all blocks are filled, no request declined, a highly unlikely occurrence in reality.)

A simple block assignment and transmitter network planning principle can be generalized from the example: assign separate blocks to those areas, cells, where among the services required there is at least one, serving one single area only. Meet multi-cell coverage requirements by transmitting the same service using channels in the blocks covering the required area. Using several low power transmitters, and gap fillers and coverage extenders if needed, i.e. local SFNs for cell coverage, the solution, the availability of the necessary number of blocks provided, is feasible and can be even economically acceptable [1].

The basic argument against this solution is, that the spectrum efficiency is lost in this scheme. In practical terms, at least until much more spectrum is available for DAB, the necessary number of blocks are not available on the Continent. The Wiesbaden Plan allocated two blocks to each region. For the solution above the theoretical minimum needed would be four.

Another practical drawback is worth to mention also. Each time when a mobile receiver passes over from one cell to the other, it has to retune and resynchronize which takes time. This means an inevitable discontinuity in the reception even of the regional programmes or of programmes available in more than one cell.

3. THE 'LOCAL WINDOW' SOLUTION [4]

The starting point of the local window solution is that accepting some disturbances of the local services — and only of the local services — in the regions between different local coverage areas, in the interference zones, the signals emitted by different transmitters of a single frequency network may be different. This is made possible by the structure of the DAB signal allowing the division of the channel capacity into a regional and a local part in a way that the interaction between the two can be minimized.

3.1. Technical background

The transmitted DAB signal is a sequence of transmission frames. The structure of a frame (without the null and phase reference symbols) is shown in Fig. 2. A transmission frame consists of a series of symbols transmitted consecutively. Each symbol carries a definite number of bits transmitted simultaneously on DQPSK modulated, closely spaced carriers, two bits on each carrier. As the modulation is differential there is no error propagation, the disturbance of a carrier can falsify only two bits in the worst case.

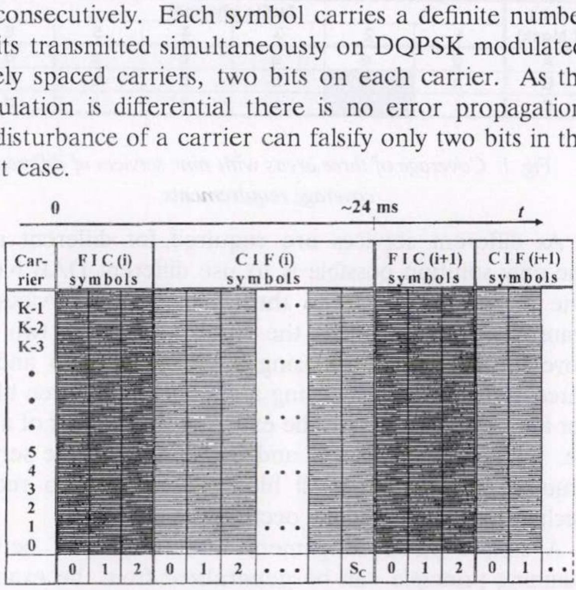


Fig. 2. Mode II transmission frames (without Null and Phase reference symbols)

The data are modulated on the carriers interleaved i.e. consecutive bits of an interrelated group of bits are dispersed within the symbol, possibly intermixed with bits of other groups similarly dispersed.

The symbols in the frame are divided into two groups carrying the data of the Fast Information Channel (FIC) and those of the Main Service Channel (MSC) respectively.

3.1.1. The MSC

The baseband MSC is the time multiplex of the components of the services to be broadcast. To each service component a subchannel of appropriate capacity is assigned. A subchannel is an array of Capacity Units (CU) of 64 bits each, the smallest, undividable unit of the dataflow. The multiplex of the subchannels convolution coded and time interleaved is split into consecutive Common Interleaved Frames (CIF) of 864 CUs and of about 24 ms fixed duration each, independent of the transmission mode. The CIFs in turn are divided into blocks with lengths corresponding to the capacity of the symbols of the transmission mode used (Table 1). The bits are interleaved and OFDM modulation is performed by inverse Fourier transformation and D/A conversion, producing the analog symbols of Fig. 2.

Table 1. Structure of the transmission frames

	Parameter	Mode I	Mode IV	Mode II	Mode III
T_F	Length of frame	96 ms	48 ms	24 ms	24 ms
C_F	CIFs per frame	4	2	1	1
F_F	FICs per frame	4	2	1	1
K	Carriers per symb.	1536	768	384	192
2K	Bits per symbol	3072	1536	768	384
K/32	CUs per symbol	48	24	12	6
S_C	Symbols per CIF	18	36	72	144
S_{Fi}	Symbols per FIC	3	3	3	8

Two important points have to be stressed here.

a) The location of a subchannel carrying one or several but not necessarily complete symbols within a transmission frame, does not change from frame to frame while there is no multiplex reconfiguration.

b) The start addresses and sizes of a subchannel can be selected such that it will be carried by an integer number of complete symbols. Note that this is not only a mathematical possibility. From Table: Sub-channel size for audio service components as a function of the audio bit rate and the protection level (short-form application) of the Standard [2], several bit rate — protection level combinations for audio services can be selected for each transmission mode, meeting this requirement alone or with a spare capacity of a few CUs, which can be used for data service components. Obviously the degree of freedom in the multiplex configuration is not as large as without this additional limitation. Configured this way the subchannel becomes an almost completely isolated part of the received data flow, as frequency interleaving is performed within the block corresponding to the symbol, and as the modulation method prevents error propagation between symbols with lengths of more than two bits.

3.1.2. The FIC

The basic role of the FIC is to periodically inform the receiver on the actual and if necessary the coming contents and structure of the CIFs of the MSC, i.e. to carry the Multiplex Control Information (MCI).

FIC data are transported in Fast Information Blocks (FIB) of 256 bits each. In Modes I, II and IV three, in Mode III four FIBs refer to one CIF and the same group of FIBs build the blocks for convolutional coding. Therefore these groups can be considered separate fast information channels. The coded length of such a group is 2304 and 3072 bit respectively. No time interleaving is applied to the FIBs which are thus in advance relative to the CIFs they are assigned to, allowing the receiver to prepare for the selection and processing of the relevant part of the received signal.

The FIBs assigned to the same CIF form a block on which error protection coding is performed. The coded blocks are then concatenated and split again into blocks this time of lengths equal to the capacity of a symbol in the mode used, necessary because the capacity of a symbol differs from that of the coded group of FIBs assigned to a CIF. The procedure afterwards is the same as described in connection with the MSC.

As a consequence of this method of producing the FIC symbols there is no possibility to divide the FIC into isolated parts to be assigned to the regional and local parts of the MSC.

On the other hand, the FIC is the assembly of various informations partly closely related and partly unrelated to the data of the MSC. Distinct characteristic of most of these informations is that they have to be transmitted repeatedly in order to enable the receiver just switched on, or tuned to another DAB block or having lost some informations due to some disturbances to catch up with the necessary informations within a reasonable time. As the importance and rate of change of the informations needed is different, their repetition rate has to be different either. This necessitates a data structure of individually addressable elements of variable length and predefined data structure realized in the six Fast Information Group (FIG) types and their numerous subtypes (extensions).

From the local window point of view those data are of interest which have to be different from one local area to the other. The indispensable of these are the data of the services, on their components and on the data of the subchannels assigned to them. These data are listed in two different extensions of FIG type 0 (extension 1 and 2). An important point here is that the order of data is free in both lists and that both lists may be split into parts carried by separate FIGs of the same type and extension. This means that one can split both the subchannel definition and the service component — subchannel assignment list — and other data needed to be separated — into regional and local parts and transmit them in different frames.

3.2. Expected operation of the local window configuration

Based upon the above one would expect that

- within a local coverage area where the field strength level of the local transmitter(s) is considerable higher than that of a neighbouring area of a different multiplex, the integrity of the local signal is upheld. The reception is undisturbed, all features transmitted are available, the possibility of reconfiguration included;
- in the interference zone the local symbols of the MSC and the local part of the FIC are more or less corrupted, the regional parts of both the MSC and the FIC, however, remain intact. Thus the reception of the regional services are undisturbed. As for the local services the results depend on the behaviour of the receiver in the given disturbance situation;
- when a moving receiver after having crossed the interference zone arrives in another local coverage zone it has to perform an implicit reconfiguration [4] not upon being informed by the Ensemble Identification of the MCI, but e.g. by recognizing the change in the local FIC data, and checking the CRC of the audio frame header, a new feature necessary in the receiver.

3.3. The Dresden tests

The local window concept has been tested in the laboratory and in an L-Band single frequency network in Dresden using transmission mode II, with very promising results proving that under the test conditions the expectations are fulfilled. The details of the test condition, test methods and results can be found in detail in [4]. Here the quotation of the conclusions would suffice:

- "the services uniform in the whole single frequency network are not disturbed by the symbol swapping for the realization of the local window;
- under certain conditions the symbol swapping can be extended to the control and signalling channel, the Fast Information Channel of DAB;
- the dimensions of the interference zone, the physically uneliminable zone of non-constructive superposition between the transmitters of different local programmes can be controlled and influenced as needed,
- as a consequence of the fading structure of the mobile reception the interference zone expands. Problems are expected with L-Band reception in mode II and first of all in the new mode IV in fast moving (> 150 km per hour) vehicles, both with and without local window."

3.4. Significance of the local window solution

The solution of the same network planning situation as in Fig. 1 using the local window concept is shown in Table 2. It has to be noted that based on the principle above the nesting of local windows should be possible either. This possibility is supposed here.

Because of the three different multiplexes three different local FICs would be needed, the nesting increases the number by one as the window encompassing the areas B and C realizing the 'regional behaviour' of programme 6 between them has to have a separate FIC. In Table 3 possible frame sequences for the local areas are shown, the five FICs fit nicely in the four-frame sequence suggested in [4] for mode II.

The fundamental difference between the conservative and the local window solution is in the number of blocks needed, four for the former one for the latter.

Table 2. Local window solution of the coverage requirements of Fig. 1. (a – d: Regional; g, e: Local 1; f: Local 2; h: Local 3; i: Local 4

Audio Channel	Area A	Area B	Area C
1	g	h	i
2	e	f	
3	a		
4	b		
5	c		
6	d		

Table 3. Frame sequences of FICs of Table 2

Area	Frame 1	Frame 2	Frame 3	Frame 4
A	G	L1	L1	L1
B	G	L2	L2	L3
C	G	L2	L2	L4

The network planning philosophy behind the local window concept can be to start with the area to be served by the regional services and place the transmitters of the SNF covering this area in view of the location, size, shape, etc. of the local windows needed. In other words, the SFN coverage, a unique feature of DAB, should be the basis in which the local windows are inserted.

4. CONCLUSION

It is clear that there are a lot of open questions concerning the local window concept and there are a lot of problems to be solved to make it an everyday solution. It is obvious either that the local window concept is not an universal solution to be applied mechanically to every situation encountered. It is, however a concept which allows to exploit fully the spectrum economy of DAB and which therefore has to be developed and applied whenever possible.

The answer to the question raised at the beginning of this article is clear. Although there are certain difficulties, whether because the lack of spectrum availability or the lack of readily applicable solutions, it seems really to be fallacy that DAB is not appropriate for local broadcasting.

ACKNOWLEDGMENT

Sincere thanks are due to the Technical Director of the Hungarian Radio, for the support of the author's studies of DAB and to Mr. Rainer Vogt, Engineer of the German DAB Platform e.V., for the permanent flow of otherwise unknown and/or unavailable sources of information about this technology.

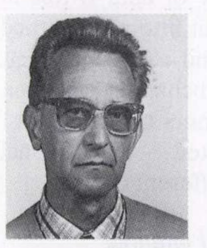
REFERENCES

[1] Thomas, M. N.: DAB for local services. Paper # 13 of the In Depth Seminar on the Technical Aspects of Digital Audio Broadcasting. Montreux. 1995.

[2] ETS 300 401: Radio broadcasting systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers. 1995. European Telecommunications Standards Institute.

[3] Draft prETS 300 401: Radio broadcasting systems; Digital Audio Broadcasting (DAB) to mobile, portable and fixed receivers. 1996. European Telecommunications Standards Institute.

[4] Strey, M., Nowottne, H-J. and Albinus, Ch.: Lokale Programmfenster in DAB-Gleichwellennetzen. Der DAB-Feldversuch Dresden. Sächsische Landesanstalt für privaten Rundfunk und neue Medien, Dresden. ISBN 3-89158-177-7.



Zoltán Vajda received his diploma as a physicist in 1956. From 1955 till his retirement in 1963 he was employed by Hungarian Radio, working first in the Development Laboratory as a engineer, later as the head of a development group and finally as deputy head of the Department for Investment, Building and Development. In the laboratory his main fields of interest were magnetic sound recording, amplifiers, audio measuring instruments and measuring methods. As deputy department head he was in charge of several large projects as new continuity studios, central switching equipment, and program associated communication system of the Radio. He has been active in national and international standardization both as member of national standardization groups and as member and later chairman of the Study Group II of the OIRT, an international organization of broadcasters. He is author of three books and series of articles published in Hungary and abroad.

SHORT-WAVE BROADCASTING IN HUNGARY

GY. DÓSA

ANTENNA HUNGÁRIA CO.
H-1119 BUDAPEST, PETZVÁL J. U. 31/33.

This paper after introducing the character of SW broadcasting deals with international trends and the present situation in Hungary. Discusses in detail the transmitter and antenna networks used at various broadcasting stations of Hungary.

1. PRELIMINARIES

In the field of radio broadcasting, the short-wave (SW) broadcasting plays a distinguished part to some extent; in fact, it is not based on surface propagation. By using appropriate technical means and using antennas of suitable design, the spatial waves are capable of covering very large areas.

The fundamental purpose of SW radio programs directed towards foreign countries is to popularize the specialities of the given country in respect of music, culture, technique, tourism and sport directly and make them known to the listeners in the target area.

As the significance of SW broadcasting in Hungary has been realized, the SW broadcasting of programs started at the Radio Transmitter Station Székesfehérvár in 1933, directed towards North-America and West-Europe and, from 1937 onwards, towards South-America as well.

It is the ionosphere that plays role in the propagation of short waves. Taking the reflection of waves on the layers of ionosphere depending on the season, hour of the day and the geographical latitude as well as the actual sunspot number into consideration, the transmitter frequency being most suitable to cover the target area during various hours of the day can be estimated.

The use of the optimum frequency selected properly is indispensable to create the maximum field intensity at the target area, with the transmitter power remaining unchanged.

Use of improper frequency may make the reception impossible at the target coverage area even with high-power transmitter and high-gain antenna used.

Within the sunspot cycle and during a given season, it is very important in the SW radio broadcasting that the transmitter frequencies to be used at specified times and in various directions will be predetermined.

Therefore, the optimum frequencies for the given target area and period shall always be determined (Frequency plan).

1.1. New trends in the development polity of SW broadcasting

During the past decades, the SW broadcasting has shown further development and increase at a world scale.

In order to increase the efficiency of SW broadcasting, the broadcasting organizations have ensured and ensure, respectively, the coverage of target areas by increasing the transmitter power, by using several broadcasting frequencies to the same coverage area and high-gain antenna systems and, in addition, by means of flexible use of various frequencies.

In the era of digital systems, many experts presume that the amplitude modulated SW broadcasting lost its importance and more efficient new means are available (satellite-based digital broadcasting).

Nevertheless, the SW broadcasting will continue to be the sole medium for long term — in spite of its obvious limitations and the variations in the quality of reception —, that is capable of reaching the listeners far away from the transmitter station. Its important advantage is, the low cost for the user and the availability to millions of radio listeners worldwide.

With the need for more frequency bands increasing progressively, the SW band may be an important means worthy of being used for the digital technology showing rapid development, due to the features as follows:

- very good coverage;
- 530 signal transmission channels in total, with the possibility of multiple allocation;
- possibility of receiving more than 2000 high-capacity transmitters;
- more than 800 million potential radio listeners.

The advantages and disadvantages (positive and negative factors) of the SW broadcasting can be summarized as follow:

Advantages:

- Cheap receivers, allowing new techniques to be used.
- Total loss of reception is very infrequent.
- Reception possibility even within buildings.
- The reception quality deteriorates slowly with the decrease in signal level.

Disadvantages:

- The frequent changes make it difficult to find the optimum frequency at the receiver side.
- The quality of reception may be dependent on the state of ionosphere.
- The crowded frequency band deteriorates the reception quality.

a.) Any interference in a broadcasting system appears for the radiolistener through the receiver.

The SW bands of radio broadcasting are crowded: a very large number of SW transmitter stations in various countries use these bands.

The transmitter stations often change their operating frequencies (summer- and winter-periods), with their programs broadcasted usually in several languages. This makes the tuning for the occasional listeners rather difficult. Earlier, the users had — and has even at present — to tune their receivers to a given SW transmitter manually (provided that the transmitter frequency was known) or to search for the desired transmitter through a wide frequency band.

Although fully synthesized receivers provided with alphanumeric keypad and LCDs (liquid crystal display) to facilitate the selection of a given frequency appeared in the recent past, still, it could not be considered to be a complete new solution.

In order to facilitate the selection of a given (desired) SW broadcasting station for the radio listeners within short time, the new ID LOGIC SW system and the SW digital service system SKYWAVE 2000 were developed, which use the database built in the receiver. These systems involved fundamental changes and opened perspectives in the SW broadcasting.

The requirements set to the new receivers were as follows:

- the desired broadcasting station should be found immediately by its name;
- the desired broadcasting station should be found immediately by the language of program;
- the broadcasting station and the character of service should be automatically identified;
- the frequency plan should be automatically updated for each programmed broadcasting station through transmission.

The user of a receiver thus optimized should be able to find the desired broadcasting station and program in the desired language without investigating the lengthy SW frequency information sheet or waste time on searching through the crowded SW bands.

The facilities listed above can be ensured if the proposed system is built in the radio receiver.

The new receivers developed by the PRS Corporation are provided with built-in databases and are capable of updating the memory within short time automatically or with only a minimum intervention on the part of the radio listener.

In the case of the intelligent receivers of this kind, the information stored in the memory can be updated by means of digital transmission. The updating information consists of either an audible signal between programs or at the end of programs (FSK system) or it is transmitted continuously by using the inaudible phase modulation of the main carrier (AMDS – Amplitude Modulated Data System). The memory capacity represents a significant factor in the price of receiver.

Time schedule (program times) for the major SW broadcasting companies of the world as well as for those of major interest on the part of listeners are also included in the data stored in the receiver memory.

The time schedule shall include:

- name of broadcasting company,
- CIRAF zones (the target area of transmission),
- time,
- frequency,
- day of week,
- language.

In the ID LOGIC system, it is the transmitter equipment that transmits the necessary data in addition to the programs and the listener in possession of a new receiver of this kind is able to read the parameters that appear on the LC display of the receiver — i.e. the frequencies, program times, transmitter power and the optimum frequency(ies) for the given geographic location (zone).

b.) In the SW broadcasting, the HFBC-87 Decree on the use of Single Side Band (SSB) system in the SW broadcasting involved significant step forward.

Most of the transmitter equipment used worldwide is unsuitable to be used for SSB operation and only few receivers are capable of receiving SSB transmission. Therefore, a period for the conversion from DSB into SSB was specified from 1987 till 2015.

Use of the SSB system will offer the following advantages as compared to DSB broadcasting towards the end of the transition period:

- more efficient use of the frequency spectrum,
- higher quality of reception under selective fading conditions,
- significant improvement in signal-power,
- significant saving in energy costs of transmitter equipment.

c.) In the improvement in respect of SW broadcasting, the reduction of crowdedness in the 6-MHz band and the necessity of expanding the spectrum below 6 MHz are considered to be additional new trends.

One reason of crowdedness is, that this band is used not only for services of high- and medium distance but also for those of maximum distance below about 1000 km.

However, the wave propagation forecast shows that for these SW services the frequency band far below 6 MHz could be the optimum, depending on the number of sunspots and the hour of day. Considering that the 6 MHz band is that of the lowest frequency available, the broadcasting companies are forced to use it, in spite of the fact that the quality of service is insufficient and the interferences caused to other broadcasting companies are inevitable.

Based on analyses and monitoring studies, the 4-MHz band is suitable to ensure very reliable coverage for short distances, especially during the periods of low sunspot number; therefore, the use of this frequency band is recommended in order to improve the quality of service and, at the same time, to reduce the crowdedness in he 6-MHz band.

d.) Recently, a new trend appeared in the building and design of SW transmitter stations that have a fine future.

In order to replace the "traditional" SW transmitter station, the so-called integrated high-power SW module station of modular design was already developed all over the world, which represents a system of very high efficiency and flexibility that can be used very economically.

The essence of the system is that the transmitter equipment and the rotatable wideband planar antenna system are accommodated in a basic module.

The high-gain antenna system consists of a rotatable wideband planar antenna system provided with two radiating planes, one of which is operating in the 6 to 11 MHz bands while the other row is used in the 11 to 26 MHz bands. Between the two rows of planes, a passive reflector is mounted.

This remote controlled radiator system is capable of making a full revolution within three minutes.

The antennas (radiator elements) can be used in various radiating configuration; that is, several types of planar antennas can be configured (by means of remote control) which, by changing the vertical characteristic, enable the coverage of areas at medium- and long distances to be optimized.

A very important advantage of the system is that, that with the needs increased (the services expanded), it can be expanded in a modular system, its space requirement is low, does not require large amount of cost intensive supporting structures and long feeder lines.

The system can be used everywhere; it provides very efficient radiation and can be installed within very short time.

In the case of transmitter equipment — used for digital SW services — only semiconductor based transmitter equipment of high efficiency and high reliability as well as low space requirement, provided with automatic (computer controlled) tuning and self test facility can be used with AMDS digital modulator (ADM Data System) and for SSB mode.

2. TRANSMITTER AND ANTENNA SYSTEMS AT THE HUNGARIAN BROADCASTING STATIONS

Antenna Hungária Co. broadcast its SW programs (both in Hungarian and foreign languages) directed abroad from three transmitter stations located in Jászberény, Diósd and Székesfehérvár to the specified target areas, by using various directional and omni-directional antennas.

At present, the target areas listed below can be covered:

- Europe, Middle-East, North-Africa,
- North-America, Canada,
- South-America.

Currently, the SW programs are broadcasted in eight frequency bands in 9 languages.

2.1. Broadcasting station Jászberény

Two SW transmitters made in Hungary (EMV) of 250 kW capacity each are installed here. The transmitters are provided with automatic tuning and operate in the frequency range of 3.9 to 26.1 MHz. The low-frequency- and high-frequency circuits of the transmitter equipment are integrated in a completely closed so-called monoblock unit.

The vacuum tubes of the driver- and power stages of transmitters are cooled by evaporation, while the other units are built with solid state components.

The frequency changes can be pre-programmed; the setting of each tuning component is implemented automatically.

The HF output of the transmitter is asymmetric, that a symmetrizing system of automatic tuning is connected to (300 Ohm balanced output).

For the purpose of measuring and testing the transmitter, an artificial antenna using soda solution is used.

The transmitter is capable of operating in both DCC mode and the so-called trapezoidal modulation mode. In the DCC mode, the carrier level decreases by 2 to 3 dB in modulation-free moments, which enables significant energy to be saved.

The transmitter station uses 28 planar antennas radiating in various directions and 2 short-distance radiators. The antenna system of transmitter station includes planar antennas suitable to be used for both large distances (above 4000 km) and medium distances (above 500 km) in a star configuration.

The selection of antenna will be performed by remote control from the operator room, with the possibility of pre-selection.

The 2/7 symmetric 300 Ohm antenna combiner installed at the central building of the transmitter station ensures the connection of the transmitter equipment either to the feed trunk line leading to the antenna star branches or to artificial antennas by means of remote control; while the remote controlled switches of feed trunk lines enable the antennas to be selected individually.

2.2. Broadcasting station Diósd

The transmitter station includes 2 up-to-date SW transmitters made by BBC, commissioned in 1983 to replace the outdated Standard-made transmitters of 100 kW capacity each.

The transmitter equipment is of low space requirement and easily serviceable, provided with automatic tuning. The units operate in the frequency range of 3.9 to 26.1 MHz. A symmetrizing system between the two unbalanced output feed or lines ensures the 300-Ohm balanced output.

The transmitter station is provided with 5 omni-directional wideband (two or three bands) antennas. The antennas were commissioned in 1990 and they are still considered to be up-to-date. They ensure the coverage of the target areas in Europe, Middle-East and North-Africa in the 4-5-6-7-9-11-13-16 MHz bands. The station is also provided with an up-to-date logper antenna system suitable for remote control, that can be both rotated and tilted. Depending on the tilt angle (+30 to −40°), the antenna system can be used from 300 km up to 12000 km in a wideband (6 to 30 MHz) to cover any target area.

The omni-directional wideband antennas are fed through 300-Ohm balanced overhead feed or cables, by using a remotely controlled antenna combiner of type 2/14, made by BBC.

2.3. Broadcasting station Székesfehérvár

Two up-to-date SW broadcasting transmitters of 100 kW capacity each made by BBC and four dual band omnidirectional antennas (4 to 5 MHz; 6 to 7 MHz; 9 to 11 MHz and 11 to 13 MHz) as well as a 100 kW logper antenna installed recently that can be both rotated and tilted ensure the coverage of the target areas in Europe, Middle-East as well as North Africa, using the necessary frequencies.

One of the 100-kW transmitters at the transmitter station Székesfehérvár broadcasts the program of Radio Kossuth, by using a 6 MHz omni-directional antenna, at 6025 kHz for distances between 250 and 700 km. For the coverage of near-by areas (Carpathian Hollow and Croatia), to SW transmitters of 20 kW each and 4 omni-directional antenna systems are available.

The two SW transmitters of 100 kW each made by BBC are relatively up-to-date, while the 250-kW transmitters installed at the station Jászberény have been in service for 22 years and do not fulfil the current requirements.

At the transmitter station Székesfehérvár, the SW transmitters of 20 kW each have been in operation for 20 to 24 years; the are used up and are far from being up-to-date at present (power stage built with vacuum tubes etc.).

The broadcasting antenna systems at the three transmitter stations are mostly up-to-date.

The frequencies necessary for broadcasting to cover the specific target areas (zones) according to the seasons (spring/summer period and autumn/winter period), together with the modes of radio link including the angle of elevation for the given period will be optimized by using proper software. Demand for the SW programs broadcasted in both foreign language and Hungarian has shown a significant increase for the recent years, especially on the part of USA–Canada, West-Europe; South-America and, to the largest extent, New-Zealand.

3. PROPOSALS AND POSSIBILITIES TO IMPROVE SW BROADCASTING

Considering the need of Hungarian Radio for covering the target areas and for simultaneous broadcasting as well as taking the wave propagation aspects and the proposals and recommendations made by WARC '92 into account, it is reasonable to take the measures listed below, aimed at improving the SW broadcasting in the near future.

- In order to cover the areas in Australia, New-Zealand and the Far-East, it would be absolutely justifiable to commission a wideband, high-gain special purpose antenna system at the Transmitter Station Jászberény, to offer the additional possibility of broadcasting in the 13 MHz and 18 MHz bands as well. Should, for example, a rotatable planar antenna system with integrated modules be used, the possibility of fulfilling any new needs in a flexible manner would also be done.
- In order to increase the efficiency of covering the near-by areas, it would be reasonable to commission at least one new omni-directional antenna at the transmitter station of either Jászberény or Diósd.
- It would be absolutely reasonable to commission a 250-kW up-to-date transmitter equipment at the Transmitter Station Jászberény, instead of operating low-capacity (20 kW) transmitter units (simultaneous broadcasting to several target areas, at various optimum frequencies).
- For Central-Europe, the broadcasting in the 4 MHz band shall be intensified within the next 8 to 10 years (WARC '92). The relocation of the 4 MHz omni-directional antenna system of Diósd would be absolutely necessary.
- Based on the resolution made by WARC '92, the use of frequencies in the expanded bands in order to increase the efficiency of broadcasting shall be ensured, according to the sunspot cycles.
- Based on WARC '92, the SW transmitters shall be prepared for SSB broadcasting.
- When converting from DSB to SSB, it shall be taken into account that the two transmitter units of 250 kW each in Jászberény — that were installed in 1974 and have been in service for 22 years — became outdated, and are unsuitable for SSB mode and the up-to-date AMDS (ID LOGIC "SW") mode.

  The replacement of these two transmitters would be absolutely justifiable within 3 to 4 years.

  The BBC-made SW transmitters of 100 kW capacity operated in Diósd and Székesfehérvár are prepared for SSB mode; however, the use of appropriate auxiliary units will be necessary.

György Dósa graduated from the Technical University of Budapest, Faculty of Electrical Engineering in 1995, and specialized in Radio Broadcasting and Communication in 1970. He joined the Office for Radio Techniques and the Technical Department of Technical Directorate for Radio and Television of the Hungarian Post (PRTMIG) in 1955. The scope of his activities includes the operation of medium- and short-wave broadcasting equipment as well as long- and short-wave communication equipment and antenna systems, their modernizing as well as installation of new equipment and antenna systems. From 1962 to June 1996, he functioned as Deputy Manager of Long-term Development Department of PRTMIG (of Antenna Hungária Co. from June 1992 onwards). During the past years, he was busy in the development and reconstruction of the Hungarian medium- and short-wave transmitters and antenna network as well as in wave propagation experiments. He is the author and co-author, respectively, of several professional papers.

RADIO TELECOMMUNICATIONS BY THE MILLENNIUM

L. PAP and GY. SIMON

ELECTRICAL ENGINEERING AND INFORMATICS DEPARTMENT
TECHNICAL UNIVERSITY OF BUDAPEST
H-1111 BUDAPEST, SZTOCZEK U. 2.

An overview is given of the main trends governing the development of radio telecommunications by the millennium. The summary of the most typical features is followed by the description of two generations of radio systems. The possibilities of improving the radio channel error parameters are given a special treatment. Further the increase of the spectral efficiency is dealt with by data compression methods and/or by proper modulation schemes. Extended radio systems and the growth rate of cellular systems are considered next. The requirements for the European Universal Mobile Telecommunications System are discussed. The aspects of global access (i.e. satellites) is also mentioned.

1. IMPORTANT TRENDS IN THE DEVELOPMENT OF WIRED AND WIRELESS TELECOMMUNICATIONS

Recent trends in radio telecommunication can be summarized as follows:

- Development of global services and networks resulting in global coverage.
- Replacement of the analog transmission by digital communication processing information in traditional "data" form.
- Mobility resulting in the renaissance of wireless technologies. New mobile telecommunication services are complementing the traditional broadcasting systems. Services are to be available for anybody, any time and anywhere.
- Integration in two senses. First the availability of various telecommunications media at the same terminal or by the same person. Secondly a transmission of various types of information: speech, audio, image, data; in a common channel; extended use of multimedia services.

2. TWO GENERATIONS OF WIRELESS DATA TRANSMISSION SYSTEMS

In wireless data transmission systems two generations can be distinguished.

Frequency Division Multiple Access (FDMA) analog modulation systems have been constructed for a direct transmission of analog signals (speech or audio). A secondary or indirect way for data transmission was used by modems with a second audio carrier frequency (telephone modems). The signalling of the various systems are either proprietary or fall within some de facto standards (e.g. the British standard MPT1327). The multiplexing scheme is FDMA. In case of duplex communication Frequency Division Duplexing (FDD) is to be used with a duplex spacing of some MHz.

In the next generation systems digital modulation of the carrier is applied for both the information transmission and signalling purposes. Requirements for authentication as well as privacy (ciphering and deciphering) can easily be performed. The trends involved include the direct transmission of data, digital coding of analog signals (efficient source coding algorithms i.e. data compression has to be performed on the analog signal samples) for a secondary (indirect) coded transmission of analog speech, video and image in the digital channel. For toll quality speech however a considerable reduction in the speed of digital speech is not affordable.

3. METHODS FOR IMPROVING THE PERFORMANCE OF DIGITAL CHANNELS

The parameters of most radio channels with at least one of the related stations moving (mobile site, e.g. a mobile set of a land mobile system, or a Wireless Local Area Network (WLAN), and/or a base station of a satellite system) are time-varying due to multipath propagation, Doppler shift and Doppler spread (delay spread, fading). For the efficient use of the radio channel several methods have been developed

The first possibility is applying some kind of diversity, i.e. transferring the same information through more channels (microdiversity), or the possibility of choosing a transmission channel from a multiple of base stations (macrodiversity) . The joint probability of more channels of poor quality (i.e. high bit error rate) at the same time is rather low, so theoretically combining the channel outputs or selecting a proper base station a more optimal result can be achieved.

In the case of microdiversity two or even more received signals are available at one site. The components can be combined, or simply the best one (with the highest carrier-to-interference ratio) can be selected. The basic possibilities include spatial (e.g. by using more antennas), frequency (more carriers for the same signal at the same time; or slow frequency hopping), polarization diversity and time diversity (packets repeatedly transmitted) and field component diversity by using receiver antennas sensitive to either the electric or the magnetic component of the electromagnetic wave. There are different approaches for selecting or combining the signals. The optimal structure would use an optimally delayed and weighted sum of the received components. Thus each component is contributing to the result (RAKE receiver).

The idea of macrodiversity can be derived by recognizing that the cell area of any system with wide area coverage is mostly limited by interference. In the basic case the co-channel interference (from stations using the same frequency) is the most significant. The second source of interference is caused by the adjacent channel components. By considering e.g. the fading there is a given probability of having poor channel quality within a cell while a much better situation for the interfering signals may be valid in the neighbouring cell.

In the second method of channel performance improvement redundancy is added to the bitstream to be transferred. This results in an increased speed, too. The additional redundancy by proper coding however makes some extent of error detection and error correction possible (Forward Error Correction, FEC), thus resulting in a reduced error rate. Some types of such codes will be briefly listed here:

Block codes are operating on source information words mapping them into codewords consisting of more bits.

Cyclic Redundancy Checking Codes are linear block codes. The information bits are multiplied by a special generator polynomial with a linear feedback shift register. Checking is to be performed by dividing the received codeword by the generator polynomial. In an error-free case the remainder is an all-zero polynomial.

A perfect code is able to correct all words with a maximal number of errors t, but is unable to do it in the case of $(t+1)$ or more errors.

The Bose-Chaudhuri-Hocquenghem (BCH) codes are error-correcting cyclic (block) codes The choice in the number of source bits and codeword bits ($2n-1$, n integer) as well as maximal number of single errors (t) in a word which can be corrected is limited and can be found in tables.

Hamming codes are perfect single error correcting binary BCH codes, i.e. if there is one error in a word it can be corrected.

The Golay codes are the only perfect codes with the possibility of correcting more errors in a word.

Convolutional coding has a redundancy ratio of k/n, where k is the number of bits in an elementary source word while n is that in an elementary codeword, but the mapping is performed by using the last K source words. As a consequence the Maximum Likelihood Sequence Estimation (MLSE) algorithm by Viterbi can be applied for choosing the best values based on a limited number of possible processes including the effect of the past.

Against the effect of error bursts (more bit errors in a codeword) interleaving can be used. In this case the source bits of the individual codewords are mixed, coded and transmitted in different channel bursts, thus each transmitted burst will contain only a fraction of the error loaded bits and after error correction the original row of bits can be reconstructed (in case of relatively short bursts of error).

The third possibility of quality improvement is related to the adaptive channel equalization. In case of radio channels most frequently the components of various delays resulting from multipath propagation with the exception of the strongest wanted signal should be eliminated or at least suppressed.

The above three groups of improvement strategies refer to real-time systems. In case of packet type communication detected errors may result in an Automatic Request for Repeat (ARQ) through the backward channel.

4. METHODS IMPROVING THE SPECTRAL EFFICIENCY OF WIRELESS DATA SYSTEMS

4.1. Efficient source coding procedures

The redundancy inherent in analog speech, pictures and music can be reduced by efficient source coding algorithms.

In case of speech quality can be maintained by waveform coding, which is used at present for the short-range applications like digital cordless telephone and wireless local loop (32 kbit/s Adaptive Differential Pulse Code Modulation, ADPCM). For cellular digital radiotelephone, however, the speed of digital information should meet the channel bandwidth requirements and therefore vocoders are used to shape the original speech samples and after forwarding the extracted parameters the receiver recovers the speech by synthesis. For example the net full-rate speed of the GSM Regular Pulse Excited coder is 13 kbit/s, while the net half rate speed using a Code Excited Linear Prediction (CELP) algorithm is 6.5 kbit/s.

The video data to be transmitted yield two-dimensional information. There are already standards for data compression, i.e. decreasing the redundancy and reducing the required speed of transmission in cases of low-speed images (Joint Photographic Experts Group, JPEG) and video (Moving Pictures Experts Group, MPEG) standard coding.

For audio transmission (including music or Compact Disc/CD quality) the Digital Audio Broadcasting (DAB) standard is established with a high flexibility for a possible choice of various speed and quality parameters for parallel transmission providing subchannels within the system channel.

The above coding approaches result in lower speed requirements for analog signals. From another viewpoint the efficient use of the limited resource of available frequency bands can also be supported by spectrally efficient modulation methods.

4.2. Some important modern modulation systems

Linear modulation structures are listed as follows:

- Quadrature Amplitude Modulation (QAM, multi-level)
- Quadrature Phase Shift Keying (QPSK, multi level)
- DQPSK (Differential QPSK with spectrum shaping coding and coherent or non-coherent demodulation).

The main advantage of these schemes is that the multi-level modulation results in lower symbol rate compared to the source bit rate (narrower frequency band for the channel), but because of the smaller distances between the symbols higher carrier to interference ratio is required.

Non-linear modulation schemes are the following:

- Frequency Shift Keying (FSK, multi-level)
- Continuous Phase Modulation (CPM): Minimum Shift Keying (MSK); Fast Frequency Keying (FFSK); Gaussian Minimum Shift Keying (GMSK). The elementary signal is chosen for a small bandwidth in order to get good adjacent channel rejection; the detection, demodulation is either coherent or non-coherent.

These modulation systems reduce the required bandwidth even at the price of introducing intersymbol interference even in the case of an ideal radio channel.

Coded modulation systems are especially interesting:

- Trellis-Coded Systems, TCS (elementary signal suited for shaping the spectrum, coherent receiver, enhancing the efficiency by algorithmic — i.e. maximum likelihood — methods).

Spread spectrum transmission systems have a wide range of special applications:

- Direct Sequential systems (DS-SS, spreading the spectrum by high rate pseudorandom chip waveforms)
- Frequency Hopping systems (FH, pseudorandom elementary orthogonal harmonic pulses within a bit from a multi-level FSK symbol set)
- Time Hopping systems (TH, pseudorandom short pulse position modulation)

The spread spectrum systems have processing gain. With a higher value of spreading the interference effects due to jamming, or other interfering signals are decreased. Therefore low spectral densities are required for an error-free communications and even Code Division Multiple Access (CDMA) is available.

As an interesting example the renaissance of a classical modulation system is also worth while to mention. Although inherently analog (5 kHz channel spacing for toll quality speech) it can be used for digital data transmission with the help of modems. The modulation concerned is of Amplitude Modulation/Single Side Band – Supressed Carrier (AM/SSB-SC) type. The theoretical advantage is the minimum bandwidth required corresponding to the baseband signal bandwidth. The application has been however limited because of the fading and Doppler effects causing additional Amplitude Modulation (AM), Frequency Modulation (FM) and Phase Modulation (PM) in the radio channel, and the sensitivity to nonlinear distortions. The proposed system uses Transparent Tone In Band (TTIB) equalization for the elimination of channel fading: a pilot tone is placed into the middle of the analog channel and used as representative of the channel fading by feedforward control. The linear transmitter power amplifier of high efficiency is a result of a baseband control loop in the transmitter. The above features of the system are based on digital signal processor algorithms.

5. EXTENDED RADIO DATA TRANSMISSION NETWORKS

In the following a list of some typical important systems is given.

Microwave backbone network elements:

- Pulse Code Modulation (PCM) hierarchy or Synchron Digital Hierarchy (SDH)
- Linear, multi-level modulation, e.g. Quadrature Amplitude Modulation (QAM)
- Point-to-point communication path sections

Digital cellular land mobile radiotelephone systems:

- Global System for Mobile telecommunications (GSM) with GMSK modulation, Frequency Division Multiple Access/Time Division Multiple Access (FDMA/TDMA), Frequency Division Duplexing/Time Division Duplexing (FDD/TDD)
- IS-41 (in the USA, DS-SS, CDMA, FDD)

Cellular mobile data systems:

- RAM Data (Mobitex)
- ARDIS

These are for packet communications without speech transmission capabilities.

Digital wireless telephone: Radio Local Loop (RLL), Wireless Private Branch Exchange (WPABX).

A standardised common access media is the Digital Enhanced (formerly: European) Cordless Telephone (DECT). In addition to general system requirements it is complemented also with special profiles of different specifications for matching the open system features for the various application areas (e.g. RLL, GSM access, etc.).

Wireless Local Area Networks (WLAN): These are intended for high data rates in excess of 1 Mbit/s for short distance applications. Base stations are not included generally in the system architecture. There are some Industrial, Scientific and Medical (ISM) dedicated bands available also for unlicenced radio operations in the ranges of 900 MHz, 2.4 GHz and 5.8 GHz. The trend is a shift to higher frequencies. Such systems are using spread spectrum technologies.

Standards are to be developed in the United States for both direct sequence and fast hopping spread spectrum cases (IEEE 802.11).

In Europe the (HIgh PErformance Radio Local Area Network, HIPERLAN) standard is intended to get use in the 5.2 GHz and 17 GHz frequency bands. In addition to packet communication schemes Distributed Time-Bounded Services (DTBS) will also be possible.

Digital paging system:

- ERMES (simplex, GMSK)

Digital trunked radio systems:

- Trans European Trunked Radio Access (TETRA) system for Private Mobile Radio (PMR) and Public Access Mobile Radio applications ($\pi/4$ Differential Quadrature/Quaternary Phase Shift Keying, $\pi/4$ DQPSK, TDMA/FDMA, FDD/TDD)

Digital radio and television:

- Digital Audio Broadcasting (DAB)
- High Definition TeleVision (HDTV), Enhanced Definition TeleVision (EDTV), Standard Definition TeleVision (SDTV) and Low Definition TeleVision (LDTV).

The number of subscribers for both the fixed (wired) and the mobile telecommunications system is growing. The annual rate of growth in the number of new cellular subscribers has been first equal to the number of new fixed subscribers (in the range of 30 million each) in 1996. The number of new cellular subscribers is growing very fast. The total number of cellular and fixed subscribers is supposed to be equal about 2005 (in the range of 600 million each).

6. EUROPE AN TRENDS IN THE NEAR FUTURE

The Universal Mobile Telecommunications System (UMTS) is under development in the 1885–2025 and 2110–2200 MHz frequency bands. It is representing a third generation system. The standardization is completed under the European Telecommunications Standard Institute (ETSI), with the following time schedule:

- requirements by 1997;
- standards by 1999;
- operation by 2002;
- standards for the second, enhanced version by 2002;
- operation of the enhanced system by 2005.

The services to be covered are summarized in the Table 1.

The platforms to be considered within the RACE II project are:

- Enhanced GSM (General Packet Radio Service: GPRS, HSCSD, GSM/DCS operation in multiplex bands)
- Two projects in the frame of the Research and development in Advanced Communications technologies in Europe (RACE) program are:
  – ATDMA (Advanced TDMA, GSM compatibility),
  – COde DIvision Tested (CODIT, with CDMA)
- MONET (fixed network issues)
- SIANT (integrating satellite components into the system)
- OFDMA (enhancement of DAB in the frame of ETSI)
- Hybrid solutions (e.g. CDMA, TDMA, etc.)

The technical background should be represented by asymmetrical downlink (base station $\longrightarrow$ mobile) and uplink (mobile $\longrightarrow$ base station) channels and an efficient common demodulation scheme.

Basic requirements include domestic, business and public coverage, toll-quality speech, capacity for multimedia, good spectral efficiency, fast handover, data speeds between 144 kbit/s and 2 Mbit/s, direct satellite access.

7. GLOBAL SATELLITE SYSTEMS

In addition to the present GEostationary Orbit Satellite (GEOS; examples include INMARSAT-M, MSAT, etc.) or High Earth Orbit (HEOS) satellites for special purposes, partly with Very Small Aperture Telecommunication (VSAT) land units (of some meters diameter!) Low Earth Orbit Satellite (LEOS), Medium Earth Orbit Satellite (MEOS) and Highly Elliptic Orbit Satellite (HEOS) systems are also considered. Examples of planned systems include Iridium (LEOS; 66 satellites for global coverage), Globalstar (LEOS, 48 satellites) and Odyssey (MEOS, 12 satellites). The low earth orbit satellites at a distance in the range of 1000 km from the earth can be used in conjunction with land mobile sets.

Table 1.

	Speed, kbit/s
Telephone	8 – 32
Data in telephone band	2.4 – 64
Audio	128
High fidelity audio	940
Video telephone	64 – 384
Short messages/paging	1.2 – 9.6
E-mail	1.2 – 64
Fax	64
Data broadcasting	1.2 – 9.6
Public announcement	8 – 32
Digital data	64 – 1920
Database access	2.4 – 768
Teleshopping	2.4 – 768
News	2.4 – 2000
Remote control	2.4 – 9.6
Navigation	64
Telegram	32 – 64

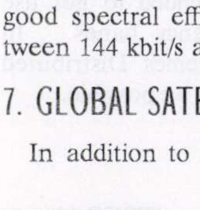
REFERENCES

[1] J. D. Gibbson (Editor-in Chief): The Mobile Communications Handbook. CRC Press in with IEEE Press, 1996.

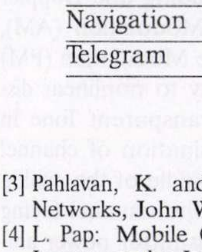
[2] *IEEE Communications Magazine*, January 1995 Vol. 33 No. 1 (special issue on Wireless Personal Communications).

[3] Pahlavan, K. and Levesque, A. H.: Wireless Information Networks, John Wiley, New York, 1995.

[4] L. Pap: Mobile Communications: Development Trends and Impact on the Information Society. *Journal on Communications*, Vol. XLVI, pp. 13-16, September-October 1995.



Laszló Pap graduated in 1987 at the Technical University of Budapest from the Telecommunication Branch of the Electrical Engineering Department. He received the Candidate of Sciences and the Academic Doctor of Sciences degrees in 1980 and 1992, respectively. Since 1992 he is a full university professor and head of the Telecommunications Department, and since 1994, dean of the Electrical Engineering and Informatics Department. His main interests include communication theory, encoding, modulation, synchronization, further mobile communications and the theory of modern telecommunication systems.



Gyula Simon graduated in 1994 at the Technical University of Budapest. He received the Candidate of Sciences degree in 1992 and is an Associate Professor at the Department of Telecommunications. He is interested in research and education on circuits, radio telecommunications and fiber optic systems.

FREQUENCIES FOR RADIO BROADCASTING IN HUNGARY

B. EISELT

COMMUNICATION AUTHORITY, HUNGARY
P.O.BOX 75
H-1525 BUDAPEST

The Media Act came into effect in Hungary on February 1, 1996. Among other things, the Act defined the number of broadcasting networks for national public and commercial services in the medium wave and in the VHF-FM bands. From this the available frequencies for radio broadcasting can be determined. The paper reviews these frequencies in each band.

1. MEDIUM WAVE BROADCASTING

The 1975 Geneva Agreement specified the frequencies that can be used for broadcasting in Hungary in the medium wave frequency band. Two national network were implemented in the band and some low power transmitters for local transmission has been realized.

The Media Act made provisions for the future use of the band. In accordance with the Act a single frequency will be available for broadcasting a national public service program in the MW band The remaining frequencies will be allocated by competitive bidding for either national or local programmes.

In the MW band development can be expected with the introduction of digital broadcasting which will result in significant quality improvement compared to the analog AM. The quality improvement will be accompanied by the possibility of data transmission of 20–25 kbit/s speed. Probably these factors will enhance the competitiveness of this band. Of course the widespread use of the band is not expected before the turn of the millennium.

2. SHORT WAVE BROADCASTING

The importance of short wave broadcasting has been reduced due to recent political changes and introduction of satellite transmission which offers excellent quality. The expedient utilization of the properties of SW band is in the interest of each administration. WARC 1992 resolutions dealt with sound broadcasting providing 790 kHz additional bandwidth for purpose. The most popular band under 10 MHz band has been increased by 200 kHz. These bandwidth increases will be effective after 2007 exclusively for SSB transmission.

It is clear therefore that improvement in SW transmission is not expected in the near future. It is interesting, however, that there was an increase in the use of SW for military purposes. The reason for this is that SW transmission can cover long distance and thus enables direct connection. Only the two end equipment are involved in the communication, thus the transmission is more safe, than in case of satellite communication. In the military purpose SW communication spread spectrum techniques are frequently used.

Two joint European proposals were discussed during the 1995 WARC meeting. The first proposed that the bands originally scheduled to be usable from 2007 should be made available already from 1997. This is supported by the Hungarian administration, since this will reduce the overcrowdedness of the band. The second proposal was that the new band should only be used for SSB from 1997. This is an unfavourable goal for Hungary, since Antenna Hungaria Co. operating SW transmitters will be unable to make the necessary investment on SSB transmitters by 1997, thus we did not support this proposal.

SW transmitters in Hungary can be directed to foreign territories only. The broadcasting aims can be grouped in the following categories:

- Hungarian language transmission to territories inhabited by Hungarians in neighbouring countries.
- transmission to neighbouring countries in their own language;
- Hungarian language transmission for the Hungarian diaspora worldwide.

The reception quality is quite poor in the above cases thus the popularity of this broadcasting possibility is quite small. Therefore it is worthwhile to provide some other way of information transmission for the mentioned directions.

The Communication Authority in Hungary is carrying out frequency planning to perform wave propagation calculations and determine optimum condition and technical parameters in the SW band. But only ITU can calculate interference parameters for the target area, since the problem requires a big and accurate database.

ITU's plan is that it should carry out the full SW planning for the member countries. But unfortunately all attempt in this direction, hitherto, has failed.

3. VHF-FM TRANSMISSION

Pursuant to the 1960 Geneva and the 1961 Stockholm Agreements, Hungary was given the opportunity to implement 4 national broadcasting networks in the 66–73 MHz so-called OIRT band. However, due to intermodulation reasons the fourth network could not be implemented by using the same sites.

Currently, 3 public service transmission networks are operating in OIRT band: broadcasting Kossuth (10 sites), Petőfi (10 sites) and Bartók (11 sites) radio programs.

A paging service is operating on the transmission network for Petőfi radio as a value added service.

In the seventies the need for expanding the VHF band emerged in the countries using the OIRT band and also in the countries using the CCIR band As a result of long preparation work, based on planning specifications previously approved at international level Geneva Regional Agreement and Plan was accepted in 1984. The agreement enables the use of the whole 87.5–108 MHz frequency band for radio broadcasting in Europe.

According to the agreement Hungary can use 127 frequencies at about 36 sites. 53 of the available frequencies are low intensity (50 W – 1 kW ERP).

The 1984 Geneva Agreement created the basis for European frequency harmonization in the area of VHF-FM radio broadcasting.

In the former socialist countries the 87.5–100 MHz band was previously allocated for TV transmission (04 and 05 TV channels). TV transmission on these channels had to be stopped according to the 1984 Geneva Agreement.

The future utilization of the VHF-FM band in Hungary is dealt with in the Media Act . According to the Act two national public service and two national commercial broadcasting networks have to be established in the 87.5–108 MHz band. Fig. 1 shows the coverage maps of the two public service and commercial broadcasting networks to be implemented. The networks presently operating in the 66–73 MHz band may be operated for 3 years and one designated network for 10 years subsequently to the establishment of the new networks.

We do not plan new developments in 66–73 MHz band. With switching off the presently operating transmitters broadcasting in the band will be discontinued gradually. This complies with the European harmonization since the band is used for mobile services in Austria, Croatia, Slovenia, Yugoslavia. The 3 years and 10 years delays in the discontinuation in the program transmission in the OIRT band is reasonable as a significant portion of the presently used receivers can be operated in the 66–73 MHz band only.

The Geneva Agreement specifies also additional frequencies for Hungary which can be used for regional or local broadcasting. According to the Media Act these frequencies, may be used in public service or commercial programs, the frequency allocation is based on a competitive bidding. Before the Media Act was passed, three biddings have been held for studio licenses of local programs. During the bidding procedures more than 200 studio licences were issued and most of them are already in operation.

Broadcasting networks in Europe are overcrowded therefore new transmitters with high power can not be allocated without disturbing the existing 1984 Geneva Plan New frequencies with small output power can be used in the CCIR band in case of suitable geographical conditions.

4. DIGITAL AUDIO BROADCASTING (DAB)

DAB broadcasting will result in revolutionary transformation of radio transmission in the future. DAB transmission will replace traditional VHF-FM transmission after the transition period.

Basic DAB advantages are as follows:

- CD quality sound reproduction even with mobile receivers;
- more efficient utilization of the frequency band than in the current VHF-FM broadcasting;
- possibility of simultaneous high speed data transmission together with the good quality radio program.

DAB was developed by the European Union under Eureka 147 program. The development took about 8 years and costed about DM 125 million. The developed system was standardized by the ETSI. The efficient frequency use is illustrated by the fact that in a 1.5 MHz bandwidth. 6 national CD quality stereo programs can be broadcasted with additional data service. It is worth mentioning that in a single TV channel 4 sets of 1.5 MHz DAB blocks with the necessary protection bands can be implemented simultaneously.

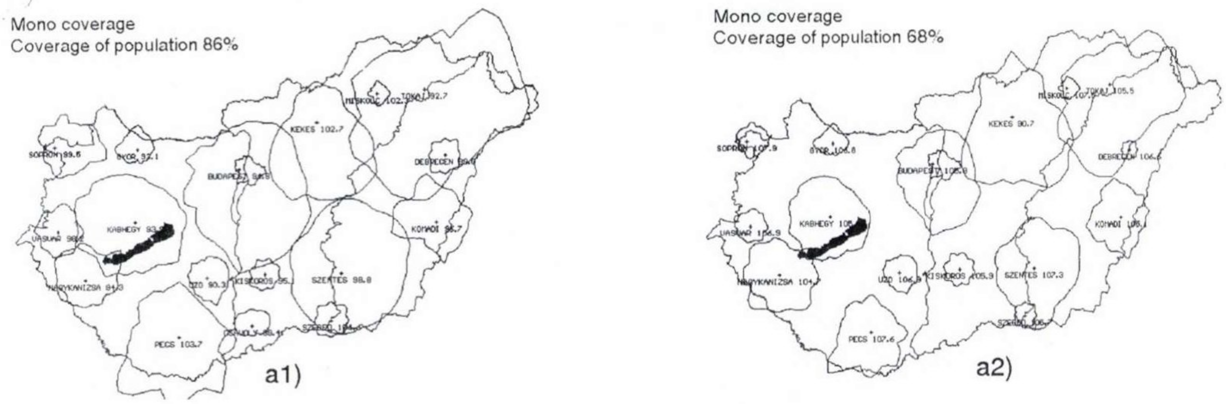
The 1995 T-DAB Planning conference in Wiesbaden allocated frequency bands for each CEPT member country for the possible implementation of two DAB networks of national coverage each network carrying 6 programs To have compatibility in the frequency allocation of neighbouring countries, each country was be divided into zones.

Both DAB networks in Hungary comprise 6 zones as shown in Fig. 2. In each DAB network some of the 6 possible programs may be of national coverage the remaining programs may serve regional purposes.

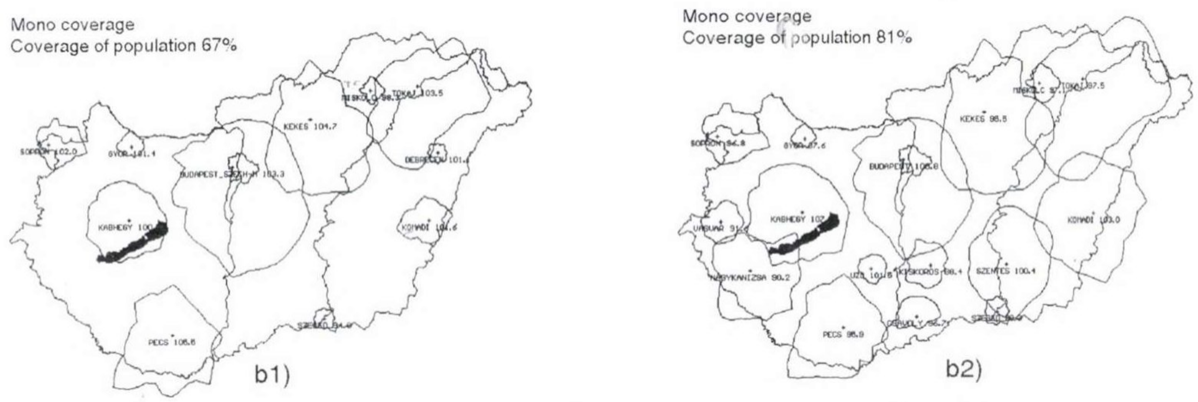
Hungary will implement the first network — with the exception of one zone — in the 223–230 MHz band A zone boundary modification (combination of zones 5 and 6) is planned. with successful international coordination. After this modification the frequencies can be allocated such that the network will operate in the frequency band of TV channel 12. This has the advantage that less transmission stations are necessary than in the 1.5 GHz L band.

The condition for implementing of T-DAB in the 223–230 MHz band is the discontinuation of the operation of TV channel 12 at the location Kab-hegy. This is agreed to happen in 2002 with the simultaneous implementation of a TV transmitter covering the same area as the present transmitter.

The second T-DAB network can be implemented in the 1452–1467.5 MHz band. In Hungary, point-to-multipoint low capacity digital microwave (IRT) systems are operating in this frequency band. These IRT systems may be operated until 2002.



a) Public service radio transmitter networks



b) Commercial radio transmitter networks

Fig. 1. Plans of the national broadcasting networks referring to the Media Act
a1) Broadcasting network of Hungarian Radio 1; a2) Broadcasting network of Hungarian Radio 2; b1) Broadcasting network called Danubius; b2) 2nd commercial broadcasting network

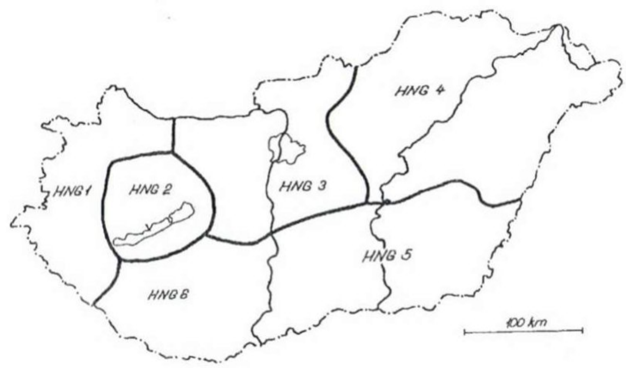


Fig. 2. T-DAB allotment areas

Béla Eiselt graduated in electrical engineering at the Technical University of Budapest in 1970, he received doctor's degree in 1975. From 1975 he has been working in the field of frequency management first at the Hungarian Broadcasting Co. and presently at the Communication Authority, Hungary, where he is Vice-President for Frequency Management. He was managing the development of the broadcasting network for the South-Yemen Television Co. for six years. In 1993 he worked at the Directorate XIII of the EU Committee on Communications and Frequency Management. He is representing Hungarian administration at various international forums of ITU and CEPT.

Student Work

INTRODUCTION

Research activity of outstanding students is an old tradition at the Technical University of Budapest (TUB). The Faculty of Electrical Engineering and Informatics (FEEI) has followed this tradition since its foundation. The so-called Scientific Student Circle (SSC) constitutes the basic framework for this activity.

Talented and interested students join some university staff member who supervises their work: introduces them into the specific field chosen, gives them the necessary professional and methodological ideas how to reach targets. Both students and teachers benefit from SSCs. Students get a deeper insight, and better knowledge of, the given area, learn the techniques of research and development, get acquainted with team work. Teachers also benefit from the student's work in their own research activity.

TUB, as other universities, organizes a conference each year where students present their work in the form of a competition. They write a report on the work performed which is evaluated by two reviewers as well as they make an oral presentation. The jury gives awards to the best ones recognizing outstanding activities both morally and financially. Those who have won award are nominated for biannual domestic SSC competition. The Hungarian Academy of Sciences established the "Pro Scientia Award" which is available for students who have proven to carry out the best scientific research activity.

Another manner to measure students' ability for individual engineering productivity is offered by the course "Project Laboratory". This two-semester course forces students to perform real project activity where they can prove their independence and creativity. Besides the usual end-of-term evaluation the best results, proposed by the supervisors, are presented and published at the "Final-Year Student Conference" of our Faculty where experts from the industry are also invited. Some conference papers often are of high scientific value.

The topic of the project lab goes usually over to the thesis. Scientific Societies such as for Telecommunications, ministries and other institutions/foundations provide awards for the best theses each year, thereby stimulating students to achieve perfection.

Real individual research of higher quality is being done by the PhD students guided by a professor. Completing their graduate studies and passing an entrance exam where their motivation and preparation for research are tested, they devote most of their time to the chosen research area. Up to the end of the three-year PhD study they are assumed to be able to compile already in a short time period their PhD thesis.

Under the STUDENT WORK heading a new column is started in this journal presenting student's research activity. In the following you can find some representative samples with the first results of PhD students from our Faculty. In accordance with the journal's profile, the topics have been chosen from the field of communications.

We recommend these papers to your kind attention.

ÉVA GÖDÖR
Editor of
Student Work Section

TIBOR TRÓN
President of the
Scientific Student Circle Committee

Éva Gödör graduated in telecommunications engineering in 1961 and received the specialized microwave engineer degree in 1966. She got her Dr. Univ. degree in 1971. She is a senior assistant of the Department of Microwave Telecommunications at the Technical University of Budapest. Her current education and research activities are related to microwave communications systems and microwave remote sensing systems.

Tibor Trón received the M.Sc. and Dr. Univ. degrees in electrical engineering from the Technical University of Budapest, in 1965 and 1974, respectively, and the Ph.D. degree in network theory from the Hungarian Academy of Sciences in 1987. He is presently an associate professor at the Department of Telecommunications and Telematics (TUB), and teaches network theory and computer aided design. His research interests are in neural computing and communications systems simulation. He is author and co-author of several lecture notes, technical reports and publications.

ADAPTIVE ANTENNA ARRAYS IN MOBILE COMMUNICATIONS

J. BARTA

TECHNICAL UNIVERSITY OF BUDAPEST
DEPARTMENT OF MICROWAVE TELECOMMUNICATIONS
H-1111 BUDAPEST, GOLDMANN GY. TÉR 3., HUNGARY

This paper summarises research activities on the usage of adaptive antenna arrays at cellular base stations. After describing the concept of adaptive antenna arrays at base sites a comparison is made between microcells and macrocells describing the feasible adaptive algorithms. A ray-tracing based microcellular propagation method is presented to determine the propagation characteristics of the radio channel. Finally, preliminary results are reported when an adaptive antenna array is implemented at base sites of microcells.

1. THE INTELLIGENT CELL CONCEPT

In conventional omni cells, when a mobile enters the cell, the transmitted power is delivered into the whole coverage area of the cell. Sectorization is a solution when the number of co-channel cells seen is reduced. The definition of *intelligent cell* is that the cell can intelligently monitor the location of the subscriber within the cell and finds a suitable way to deliver the power right into the direction where the mobile is located resulting in improved carrier-to-interference (C/I) ratio and as a consequence, spectral efficiency is increased as well.

2. ADAPTIVE ANTENNAS AT BASE STATIONS

The main advantages offered by adaptive antennas are the followings:

- Using the adaptive array as a spatial filter, C/I ratio can be improved both uplink and downlink by positioning the main lobe to the wanted signal and rejecting interferer sources. In downlink less interference is spread out.
- Cancelling the unwanted delayed waves, caused by multipath propagation, can reduce intersymbol interference.
- Ability to form the coverage area of each base station to match local propagation conditions and increased antenna gain.
- Possibility of complete frequency reuse when adjacent channels are co-channels.

There are two groups of algorithms which can be applied for calculation of the weights of an adaptive antenna array. The first group of methods estimate the direction of arrival of the received signals with the help of high resolution algorithms. When the directions are known accurately, digital beamforming can direct the antenna main lobes to the wanted sources. These methods are suitable in *large cells*.

In case of *small cells* the nature of multipath propagation results in a wide angular spread of the received signals. Thus, in urban areas different methods should be applied, which are known as optimum combining methods. In small cells uplink and downlink communications are in complete uncorrelation due to the different propagation situation and the frequency duplex used. The information collected in uplink direction cannot be used directly in downlink, however, the applications of adaptive arrays in microcells can compensate the asymmetrical power budget.

3. PROPAGATION MODEL

As a simulation tool a site-specific two-dimensional 'image-based' ray-tracing algorithm is implemented with the addition of one ground-reflection for each ray. In the calculations to determine the signal strength and phase of all wanted and interfering signals direct, single, multiple wall-reflected and single diffracted rays (plus their corresponding ground reflected pair) are considered with the following assumptions. Signals are narrowband and vertically polarized. The antenna patterns are omnidirectional, all surfaces are smooth and buildings are infinite high. In a microcellular situation the base sites and the mobile stations are assumed to remain below roof level. This leads to the neglection of those diffracted rays leaving the roofs. Diffraction is only considered for vertical edges (street corners) with the use of uniform theory of diffraction (UTD). An extra ground-reflection is added to each diffracted ray. Wall transmission is neglected. The short-term signal variations are to be considered in future simulations.

4. SIMULATIONS

A PC based simulation tool is presented to examine the performance of an adaptive array in microcellular environment in the uplink direction.

The Sample Matrix Inversion (SMI) adaptive algorithm is implemented. The signal arriving from the MS is taken as the reference signal. The thermal noise is represented by its power in the diagonal elements of the covariance matrix. In the future, hardware tolerances will also be considered.

The presented results are the C/I ratio and RMS delay spread of the MS-BS path. The second factor indicates how adaptive arrays can support high data rate systems. Calculations were performed in case of a single interferer resulting in C/I of 5.23 dB when omnidirectional receiver is assumed at the base station. The base site was situated at a street corner and the MS and the interferer were in perpendicular 15 m wide streets. Fig. 1 shows the dependency of C/I as a function of the number of array elements of a circular array with element spacing (ka) of 3.

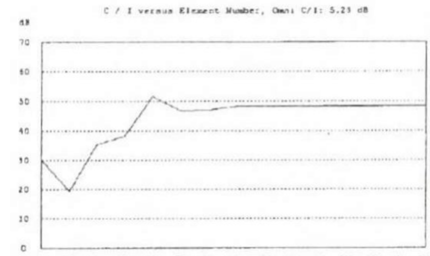


Fig. 1.

Result of reduced RMS Delay Spread is presented in Fig. 2 as a function of the number of elements. As the element number is greater than eight, no further improvement can be noticed in both parameters examined.

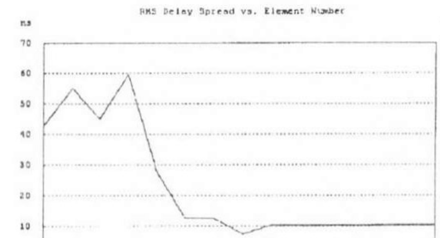


Fig. 2.

The simulations were performed in only a single receiver position neglecting the effects of statistical variations, but these initial results show the applicability of adaptive arrays in this environment.

MEASUREMENT RESULTS ON THE SATELLITE-TO-INDOOR RADIO CHANNEL

Z. BODNÁR and Z. HERCZKU

DEPARTMENT OF MICROWAVE TELECOMMUNICATIONS
TECHNICAL UNIVERSITY OF BUDAPEST
H-1111 BUDAPEST, GOLDMANN GY. TÉR 3., HUNGARY

The rapidly developing communication is planning to implement Satellite Personal Communication (SPC) systems. It means that there is a need for both narrow- and wideband characterization to get a deeper, more comprehensive understanding of the satellite-to-indoor propagation channel. In this article we will describe a wideband sounder, present a visual result of the wideband measurements and we mention some conclusions we have drawn from narrowband measurements as well.

1. MEASUREMENT EQUIPMENT, WIDEBAND RESULTS

The wideband measurement set-up can be partitioned into three parts: the RF sounder containing the transmitter and the receiver; the control and processor instruments; the software and firmware elements. Basically the same equipment can be used for narrowband measurements also.

In case of wideband measurement the transmitter radiates a BPSK modulated signal at 1620 MHz. We modulate the carrier with a Pseudo Random Binary Sequence (PRBS) generator. In the receiver (Fig. 1) the incoming RF signal is demodulated with a replica of the transmitter's PRBS signal. We can apply a general purpose narrowband receiver because the bandwidth of the remodulated signal is much less than the bandwidth of the original transmitted signal.

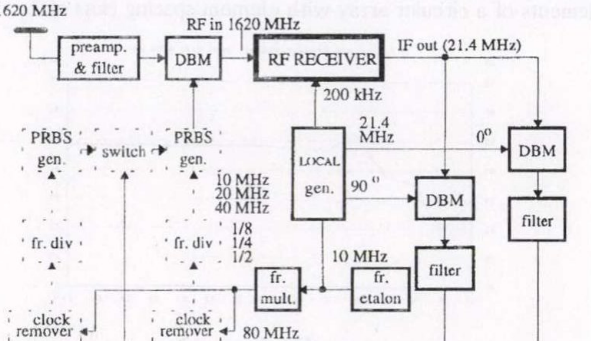


Fig. 1. The receiver of the sounder. The functions implemented by XILINX module are shown with dotted border.

Using a slightly different clock frequency in the receiver's PRBS generator is the well-known conventional method to shift PRBS signal in time for correlation measurement. In our receiver the PRBS generator operates at the nominal frequency without any frequency shift. On the other hand we have to perform different delays between the transmitted and received PRBS signal to characterise channel at different delays. Different delay means different phase shift between the PRBS signals. We applied a clock pulse remove circuit to perform the appropriate phase control. The minimum available time shift is 12.5 ns because this circuit operates at 80 MHz. Additionally we can apply an external clock with different frequency.

The IF signal is processed in the IQ processor formed by two Double Balanced Mixers (DBM), filters and A/D converters. The IQ signal is stored in the control computer.

This wideband channel sounder system yields the time variant impulse response of the channel. First, the computer calculates the delay profile of the channel by averaging the square of impulse responses in time (a typical result can be seen on Fig. 2). The frequency correlation function of the time variant transfer function is the Fourier-transform of the profile. The coherence bandwidth and the rms delay spread can be calculated from the frequency correlation function and the delay profile.

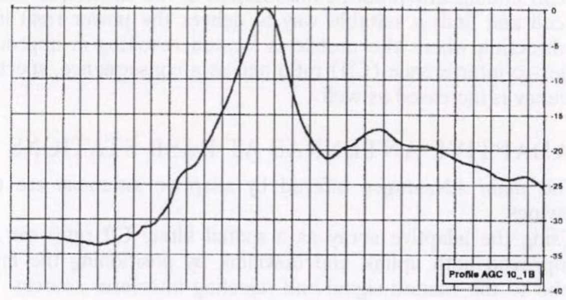


Fig. 2. Delay profile. Elevation angle 60°, full circle. Average delay spread = 57.83 ns.

Delay spread was close to 60 ns, i.e. coherence bandwidth is in the order of 10-20 MHz. This is high enough to make voice-band communication without difficulties possible.

2. NARROWBAND RESULTS

In narrowband measurements only a simlpe CW unmodulated signal is transmitted through the channel but the same receiver hardware and IQ detection can be applied as in the wideband measurements. In these measurements we determined the average penetration loss of the propagating signal. We prepared elevation and azimuth dependence diagrams as well as first order statistics of the results. The analysis yielded empirical probability density functions of the received signal amplitudes. It was possible to determine the amplitude distributions, e.g. to prove that the distributions can be approximated to be Rayleigh — or more precisely, Ricean with a small Rice factor. We were able to determine the Rice factors in cases of sectorized measurements. (In case of the overall data the distribution can not be determined because of the different mean power levels the signal has in different sectors). Unlike in the outdoor measurements, where the higher elevations mean generally smaller losses, in the indoor case in many cases higher elevation meant bigger losses. This is probably because the floors had bigger impact on the signal strength than the sidewalls. This assumption is supported by the results that the deviation of the losses turned out to be sometimes 2-3 times bigger at high elevation angles than at smaller ones in the investigated building.

CONSIDERATIONS ON THE WAVEFORM IMPROVING IN SWITCHING-MODE POWER AMPLIFIERS

A. TELEGDY

DEPARTMENT OF MICROWAVE TELECOMMUNICATIONS, TECHNICAL UNIVERSITY OF BUDAPEST
H-1111 BUDAPEST, GOLDMANN GY. TÉR 3., HUNGARY

This paper presents the basic ideas of a work presented also on the Fourth Japan–Hungary Joint Seminar held in Fukuyama, Japan in July, 1996.

The Class E switching-mode amplifier achieves high efficiency even if the switching times of the switching device are up to 10–15 % of the output waveform's period. At higher frequencies the appreciable switching times yield considerable increases of the losses. That dissipation could be minimized if the nonideal voltage and current waveforms could be made to have zero values at the times when the switch is turned on and off. It is known from an earlier publication, that no configuration of linear circuit elements supplied from DC source can provide this if nonzero output power is to be delivered to a load. In the presented work a new general equation was derived to characterize the operation of the lossless switch from the energetical point of view. A waveform improving method is presented which can make the amplifier have high efficiency and can provide zero voltage and zero current at both transitions of the switch. The simulation results confirm that the method suggested for achieving continuous waveforms on the switch can increase the efficiency of an amplifier stage even if the switch doesn't work as an ideal one.

The original concept of the basic Class E amplifier contained the requirement for a jumpless switch-voltage waveform (jumping from V_0 to zero when the switch turns on, where V_0 is desired to be small) to avoid the power loss of $CV_0^2 f/2$ in discharging the capacitance C in parallel with the switch, from a nonzero voltage V_0 to zero, at switching frequency f.

It is considered desirable to find a way to make the circuit work well without requiring a jump in the switch-current waveform at turn-off, to allow efficient operation at frequencies high enough that the switch turn-off would occupy an appreciable fraction of the waveform period.

The first part of the paper presented the derivation of a new power equation.

We are interested in the powers at harmonic frequencies flowing into the switch. Let the real power of the switch at the k-th harmonic be denoted by P_k. Let us assume that one of the waveforms is continuous and the other waveform may have a jump of Δ. S is the slope of the waveform that does not have a jump at the switching instant. S is measured in V/rad or A/rad, depending on whether the voltage waveform or the current waveform is the one that is continuous. The equation of power conservation must be satisfied because the ideal switch cannot generate or dissipate any power.

By calculating the average value of the voltage derivative times the current derivative over the period a new equation can be written in two forms as follows:

$$\sum_{k=1}^{\infty} k^2 P_k = \frac{1}{4\pi} S \cdot \Delta$$

$$\frac{1}{T} \int_0^T v'i'dt = \omega_0^2 \sum_{k=1}^{\infty} k^2 P_k$$

For P_{OUT} to be greater than zero, $v'i'$ must be nonzero at some time(s) during the cycle. Except at the instants of switching, v' is zero when the switch is "on" and i' is zero when the switch is "off".

As a direct consequence of the power equation it can be concluded that in order to satisfy both of the requirements of (a) jumpless voltage and current waveforms on the switch and (b) nearly pure sinusoidal output waveform (nonzero output power) we have to inject active power into the amplifier stage so as the power equation to be satisfied.

In the particular case when the resultant current of the switch and its parallel capacitor contains DC, fundamental, and n-th harmonic terms only.

$$i = I_0 + I_{1A} \cdot \cos\vartheta + I_{1B} \cdot \sin\vartheta + I_{nA} \cdot \cos n\vartheta + I_{nB} \cdot \sin n\vartheta.$$

In this biharmonic case the n-th order harmonic frequency power $P_n \neq 0$ that is to be injected into the amplifier stage can be calculated as

$$P_n = -\frac{1}{n^2} P_1.$$

Taking into account the requirement for jumpless waveforms and the condition for unipolar voltage on the switch the current components are the solution of a linear equation system which consists of 4 equations.

The MicroSim Design Center, a PSPICE based program package was used for simulation purposes.

The simulated waveforms of a biharmonic amplifier were in good agreement with the theoretically calculated ones.

In general, the practical switching devices working on high frequencies cannot be considered ideal switches. For this reason the theory presented in this paper should be applied carefully in the RF domain. The simulations have shown that the power injection can improve the efficiency even when the switch can't be considered as an ideal one. The simulation was made on a circuit operated on 1 GHz, containing an RF power MOSFET capable to deliver 5W output power with 50 % nominal efficiency. The PSPICE parameters of the basic MOSFET and the parameters for the rest of the corresponding subcircuit, which altogether models correctly the high frequency behaviour of the MOSFET device, were available on the Internet. The manufacturer provides the recommended generator impedances the optimum load impedances (resistance and reactance) on the specified operating frequencies (e.g. 1 GHz). These were realized by series RLC circuits. Successive simulations were accomplished at first, with zero second harmonic injected power in order to tune the rest of the parameters to have maximum efficiency. In the case of optimum tuning, the second harmonic power injection yielded an additional increase of the efficiency by 2.5 %. (The injected harmonic power was considered as supply power together with the DC supply power.) The current waveform on the MOSFET was nearly sinusoidal and the voltage waveform was smooth, too. This is the reason why the efficiency increase was so small. (In the case of a Class A amplifier the whole injected harmonic power is dissipated on the active device.) Expectedly, the harmonic power injection can improve considerably the efficiency when the active device operates close to the switching-mode.

The main results:

- A new equation was derived, to characterize the electrical operation of a lossless switch.
- The possibility of jumpless waveforms in a switching-mode tuned power amplifier was investigated. It was shown that the harmonic power injection is necessary to fulfil the jumpless waveform requirement.
- It was verified by simulation means that the harmonic power injection into the stage can improve the energetical balance of the amplifier even when the active device of the circuit is far from being an ideal switch.

ACKNOWLEDGEMENTS

The presented work was supported by the OTKA project no. 11595. I wish also to express my best gratitude to my scientific advisor Dr. Béla Molnár without whose help the research couldn't have been carried out.

APPLICATION OF NEURAL NETWORKS TO CALL ADMISSION CONTROL FOR ATM NETWORKS

S. IMRE

DEPARTMENT OF TELECOMMUNICATIONS
TECHNICAL UNIVERSITY OF BUDAPEST
H-1111 BUDAPEST, GOLDMANN GY. TÉR 3., HUNGARY

The bandwidth gain based on statistical multiplexing is one of the main advantage of Asynchronous Transfer Mode (ATM). Therefore Call Admission Control (CAC), which allows efficient use of network resources (bandwidth, switches, etc.), plays an important role among the management functions of ATM Networks.

Using an appropriate source model, CAC can be regarded as a constrained optimization problem, formulated as follows: *How to provide the most accurate tail estimation of the aggregate traffic (in order to maximize the number of accepted calls still not exceeding the network capacity), based on simple statistical parameters declared by the users?*

A traffic state can be accepted by the CAC if

$$P(Y = \sum_{j=1}^{J} X_j > C) < e^{-\gamma}$$

where Y denotes the aggregate traffic generated by J user, X_j is the traffic of a single user, C represents the link capacity and γ is the Quality Of Service (QOS) parameter. The fast evaluation of the above inequality (known as the tail estimation of the aggregate traffic) proves to be a hard problem. One way to slve this problem is to interpret CAC as dichotomy.

The idea is the following: let us group the users into traffic classes with regard to their traffic parameters. Each user from the same class presents homogenous load to the network. The state of the system is therefore characterized by a vector $\mathbf{n} = (n_1, n_2, \ldots, n_J)$ whose ith component tells how many users want to be admitted from the ith class. Thus, CAC is a set-separation (see Fig. 1) defined on the traffic-state space **V** expanded by all possible **n** traffic vectors.

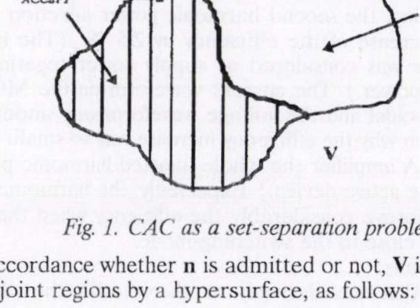


Fig. 1. CAC as a set-separation problem

In accordance whether **n** is admitted or not, **V** is separated into two disjoint regions by a hypersurface, as follows:

$$\mathbf{V}_{ACCEPT} := \{\mathbf{n}: \quad P(\sum_{i=1}^{M} Y_i > C) < e^{-\gamma}\}$$

$$\mathbf{V}_{REJECT} := \{\mathbf{n}: \quad P(\sum_{i=1}^{M} Y_i > C) \geq^{-\gamma}\}$$

$$\mathbf{V}_{ACCEPT} \cup \mathbf{V}_{REJECT} = \mathbf{V} \,;\; \mathbf{V}_{ACCEPT} \cap \mathbf{V}_{REJECT} = \mathbf{0}\,,$$

where M denotes the number of the traffic classes.

Based on the set separation capabilities of neural networks have long been established, a two-layer neural network can perform the dichotomy (depicted in Fig. 2) corresponding to the CAC.

The two-layer neural network carries out the set separation by implementing the following decision function:

$$y = \operatorname{sgn}\left[\sum_{i=1}^{N} a_i \cdot \operatorname{sgn}\left(\sum_{j=1}^{M} w_{ij} \cdot n)j - w_{i0}\right) - a_0\right].$$

Each neuron in the first layer represents a hyperplane. The single neuron in the second layer performs a logical OR function uniting the individual separation hyperplanes.

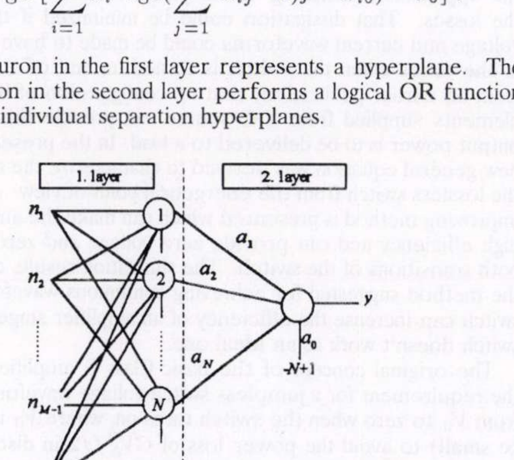


Fig. 2. Two-layer neural network

The only question left is how to determine the weights of the individual neurons that the deviation of the union of the hyperplanes from the theoretical separation hypersurface be minimized (optimal network utilization)?

Implementing the OR function the weights are set as $a_i = 1$; $i = 1, 2, \ldots, M$ and $a_0 = -N + 1$.

The real challenge lies in determining the weights of the first layer. Instead of applying a learning algorithm, the weights are calculated by forcing the best polynomial approximation to the theoretically calculated hypersurface. In this way better and faster approximation can be obtained than by learning.

The use of neural networks described above allows fast and accurate CAC in high speed networks like ATM. The results are shown for two traffic classes in Fig. 3, where the separation surfaces based on neural network and statistical inequalities are compared.

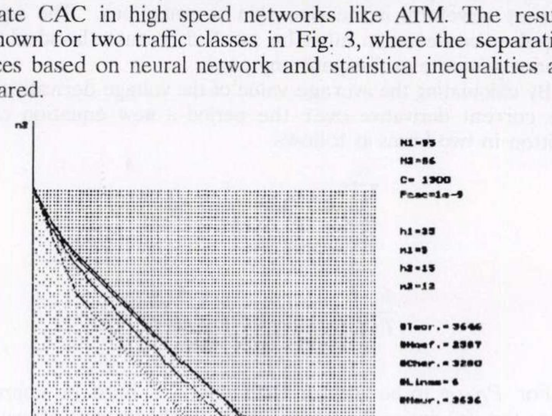


Fig. 3. Simulation results

SIMULATION IN THE DESIGN OF VSAT NETWORKS

A. VARGA

DEPARTMENT OF TELECOMMUNICATIONS
TECHNICAL UNIVERSITY OF BUDAPEST
E-MAIL: ANDRAS.VARGA@HIT.BME.HU

VSAT networks play an important role in business data communications in many countries, including Hungary. With VSAT networks, careful network design and application can improve performance significantly and spare network resources. In this article, simulation was used to investigate an important case when the application worked with terminal emulation.

1. INTRODUCTION

A VSAT network is a star topology network that consists of VSATs and a central hub connected via satellite links. The outbound direction uses a single TDMA stream (*the outroute*) that is received by all VSATs. On the inbound direction, VSATs are divided in groups and each group is assigned an independent channel (*inroute*) on a separate frequency. Medium-access protocol is assigned on a per-inroute basis: Aloha, slotted Aloha, or a form of TDMA can be used (fixed assignment [F/TDMA], a variety of demand-controlled TDMA etc.).

The satellite links introduce long delays (about 560 ms on one hop) and the data rates are slow (between 9.6 kbit/s and 64 kbit/s). In general, it is not an easy task to predict how much bandwidth a customer application needs for satisfactory performance and how the VSAT network's internal protocols need to be configured (especially, which multi-access protocol should be used and how). Also, very often legacy applications (originally not designed to work with long delays) are ported to VSAT networks. It is important to predict how these applications behave on VSAT links and determine how the applications can be tuned to be more effective. Because of the complexity of the model, simulation is very often the only feasible method to investigate them.

2. VSAT MODELLING

Today's VSAT systems have internal protocol structures that were designed with the OSI reference model in mind. Wherever it was possible, OSI protocols were used in the VSAT simulation models. We built the models to be as close to the original system as possible. The modelling was done using *OPNET Modeler*, a leading software for the simulation of communication networks [2].

The characteristics of the *physical layer* (the microwave satellite link) were modelled by a simple collision model and by quantities like delay, bit rate and bit error rate. At the *data-link layer/MAC sublayer*, the most common multi-access protocols were modelled on the inroutes (Aloha, and different TDMA schemes). For the *LLC sublayer*, a LAPB-like protocol with selective retransmission was incorporated in the model. As *network layer* and *transport layer* protocols, the OSI-standard CLNP and TP4 were used, respectively. There was also an *application layer* that was used to bridge the gap between customer protocols and the transport layer of the VSAT network.

3. A SIMULATION CASE STUDY

The network consisted of a host computer and several terminals placed to LAN sites (5 to 35 terminals per site). Each LAN was connected to a VSAT. The host machine was connected to the hub of the satellite network via a terrestrial link. The terminals accessed the host with terminal emulation, using TCP/IP. It was assumed that originally, terminals had used remote-echo character-mode terminal emulation which performed bad on satellite links. Therefore, the introduction of other terminal emulation methods was considered. In this study, two kinds of block-mode terminal emulation were examined: one supported local editing of database fields (*field-based*), the other supported local editing of a whole database form (*form-based*).

The subject of the study was to find out the following: how the performances of the two types of *terminal emulation* compare to one another; which *multi-access protocol* would be the optimal choice; whether terminals at a *small* site behave significantly different than those at a *large* site.

4. SIMULATION RESULTS

Four series of simulation runs were performed for two multi-access protocols (slotted Aloha and fixed assigned TDMA) and the two types of terminal emulation. Within each series, the inroute load was gradually increased by assigning more and more terminals on average to one slot.

The main statistics that were looked at were the *response times* at a small (6 terminals), medium-size (12 terminals) and a big (35 terminals) site.

The following observations were made about the simulation results. First, response time and channel load for the two types of terminal emulation did not differ significantly. Second, the distribution of response time showed a very strong dependence on the size of the site. The difference was the largest between the small and medium-sized sites; medium-sized and large sites behaved almost identical. Third, small sites performed significantly better with slotted Aloha than with F/TDMA at the same bandwidth allocated to them; or, they needed about 50 % more bandwidth with F/TDMA to deliver the same response times as with slotted Aloha. Medium and large sites behaved good with F/TDMA (most values being under 1 sec) but virtually choked with slotted Aloha.

5. CONCLUSION

A number of suggestions can be derived from the simulation results. No single multiple access protocol is appropriate for all site sizes. Instead, a mixture of F/TDMA (for medium and large sites) and slotted Aloha (for small sites) should be used. That results in significant savings of inroute bandwidth and a gain in application response time compared to an entirely F/TDMA solution. The bandwidth requirements that are a good compromise between application performance and bandwidth consumption are also determined by simulation. The study showed that using simulation in network design can result in both better application performance and savings in network resources.

6. ACKNOWLEDGEMENT

The author wishes to acknowledge the support of Hungaro DigiTel Ltd, a major VSAT service provider in Hungary that made it possible to carry out this study.

REFERENCES

[1] OPNET Modeler manuals, version 2.5C, MIL 3, Inc.

[2] K. Budai, A. Czifra, G. Dallos, P. Giese, D. Jobbágy, C. Szabó, L. Szabó and A. Varga: "Traffic simulation and cost analysis system – Hungarian contribution to COST-226" *COST-226* 10-12 May, 1995.

LOSS SPEECH ENCODERS BASED ON LINE SPECTRUM PAIRS

HAI LE VU

DEPARTMENT OF TELECOMMUNICATIONS
TECHNICAL UNIVERSITY OF BUDAPEST
H-1111 BUDAPEST, GOLDMANN GY. TÉR 3., HUNGARY
E-MAIL: HAI@HIT.BME.HU

Line Spectrum Pairs (LSPs) have been gaining interest as alternative parameters because of their intrinsic properties which permit more efficient encoding than the more often used reflection coefficients. In this paper a new method of quantization over the LSPs using the Karhunen-Loeve (KL) transform is presented. It is shown that the optimal KL transform is an alternative method to spread the high-frequency spectral distortions when not all characteristic parameters of speech representation are transmitted (loss encoders).

1. INTRODUCTION

A low-bit-rate speech encoder must employ bit-saving measures to achieve intelligible and natural sounding synthesized speech. Some important measures are: quantization of parameters based on their spectral error sensitivities, and quantization of parameters in accordance with properties of auditory perception (i.e., coarser quantization of the higher frequency components of the speech spectral envelope, and finer representation of spectral peaks than valleys). The use of LSPs [1] makes it possible to employ these measures more readily than the better known reflection coefficients. As a result, a new loss encoder-scheme for encoding the LSP parameters are developed using the KL transformation which reduces the high-frequency distortions.

2. STATISTICAL PROPERTIES OF LSP AND CODING SCHEME

We studies the statistical property of LSP by using a different speech data base of male and female speech data, each frame is 20 ms long and 10th order LPC analysis is employed. Our investigation indicates that the LSPs within frame, and from frame to frame, are correlated [2].

In this paper we propose a new method derived from KL transform based upon this nonuniform property of LSPs. The first 4 LSPs are encoded directly into 4 bits for each parameter. The remaining coefficients then are transformated with KL matrix, but only three dominant KL coefficients are quantized (into 3 bits for each parameter) and are transmitted.

At the receiver three parameters, which are have been chosen by us from our experiment of studying of different speech data base, and other dominant KL coefficients produce the high-frequency components by inverse KL transform. It can be seen that with a 30 % reduction in bit rate requirements (from 36 bits/frame to 25 bits/frame), the spectral of reconstructed speech from LPC coders is similar to the spectrum of LSP coders.

3. EXPERIMENTS AND RESULTS

We have compared the performance of the LSP-based loss encoder-scheme at 25 bits/frame rate using optimal nonuniform scalar quantization with partial correlation (PARCOR) coefficients based speech coding system at 36 bits/frame [3]. The spectral distortion measure (SD) is known to have a good correspondence with subjective measures, that is defined as follows:

$$SD = \frac{1}{N}\sum_{n=1}^{N}\left(\frac{1}{\pi}\int_{0}^{\pi}\left(\log S_n(\omega) - \log \hat{S}_n(\omega)\right)^2 d\omega\right) \quad (\mathrm{dB}^2)$$

where $S_n(\omega)$ and $\hat{S}_n(\omega)$ are the spectral of the nth speech frame of original- and distorted speech, respectively. N is the total number of frames.

The speech signals were digitized at an 8 kHz sampling rate. At 10-th order LPC analysis, based upon the autocorrelation method, was performed on the data using a 20 ms Hamming window. About 5840 frames of LPC vectors were used in the experiments. Results are shown in Table 1.

Table 1. Spectral distortion for LPC and LSP loss scheme

$0-2$ dB^2	$2-4$ dB^2	> 4 dB^2	Average
LPC (36 bits, 10 PARCORs are transmitted)			
93.458 %	2.054 %	3.715 %	0.809 dB^2
LSP (25 bits, 4 LSPs and 3 KLs are transmitted)			
94.366 %	1.935 %	3.698 %	0.806 dB^2

4. CONCLUSION AND FURTHER STUDIES

This paper presents the use of Karhunen-Loeve transform for the LSP coefficients in low bit-rate speech coding. At 25 bits/frame, the new technique achieves the same level of spectral distortion as LPC representations of speech at 36 bits/frame (with approximately 1 dB^2 average spectral distortion). Replacing the scalar- by vector quantization to quantize KL coefficients are among the interesting ideas for further research on this subject.

REFERENCES

[1] F. Itakura: "Line Spectrum Representation of Linear Predictive Coefficients of Speech Signals", *J. Acoust. Soc. Amer.* Vol. 57, S35(A), 1975.

[2] H. L. Vu and K. Elek: "A Low Bit Rate LPC Vocoder Model Based on Line Spectrum Frequency", *7th Television and Audio Conference*, Budapest, May, 1996.

[3] H. L. Vu: "Speech Coding by the Optimal Transformation of LSP Parameters", *IEEE NORSIG'96 conference*, Finland, September, 1996.

PERFORMANCE OF FUTURE EVENT SET IMPLEMENTATIONS

G. LENCSE

DEPARTMENT OF TELECOMMUNICATIONS
TECHNICAL UNIVERSITY OF BUDAPEST
H-1111 BUDAPEST, GOLDMANN GY. TÉR 3., HUNGARY
E-MAIL: GABOR.LENCSE@HIT.BME.HU

Seven data structures were examined with simulation and compared to find out which one should be used to implement the Future Event Set of an event driven discrete event simulator. The number of key comparisons and pointer references as well as the CPU time were measured. The effect of the different CPU architectures was examined. Various parameter settings were used to determine the characteristics of the studied event set implementations.

1. INTRODUCTION

In an event-driven discrete event simulator the events (the state changes of the system) are stored in the Future Event Set (FES). The data structure and the algorithms used for storing the elements of the FES influence the speed of the simulator to a great extent.

The most common data structures have already been studied and their general performance characteristics have been determined. Most investigations assume that the keys are random (with uniform distribution), the number of search operations is much higher than the number of insertions or deletions and the element to be searched or deleted is also randomly chosen. Nevertheless, in the case of the FES, the keys are not uniformly distributed, but show a rising tendency; in the vast majority of cases, the first element is deleted and a new one is inserted. Sometimes randomly chosen elements are deleted, but no other search operations are used.

Reeves [1] examined variants of lists and heaps, but the different kinds of tree structures and the skip list [2] required further study.

The examined data structures were: ordered single-linked list, binary tree, AVL-tree, B-tree, 2-3-tree, heap and skip list.

2. THE INVESTIGATION

The behavior of a discrete-event simulator (from the viewpoint of the Future Event Set) was simulated in the following way: new events were inserted into the FES, then the *first* event (the one with the smallest time stamp) was taken out or a randomly chosen event was deleted. This procedure was repeated many times and the time stamp of the *new* event was the sum of the time stamp of the most recently removed "*first*" event and a random delay computed according to different distributions. The parameters were: the number of the events in the FES, the state of the FES (transient or steady), the proportion of the randomly deleted events and the distribution of the delay (exponential, uniform, and two normal distributions with different deviations).

To determine the performance characteristics of the studied event set algorithms, the number of key comparisons and of pointer references as well as the CPU time were measured. By changing the parameters of the simulation model orthogonally, simulations were run with all possible parameter combinations. The simulation was performed on the following processors: Intel 486DLC, Intel 486DX, DEC ALPHA.

3. RESULTS

By examining the number of key comparisons, it was found that the balanced trees and heap produce the lowest values among all the data structures. For this reason, if we use a processor where the CPU time of the event set operations is dominated by the key comparisons (that is, floating point operations are not supported by hardware), balanced trees and heap are the best choice. Binary tree produced good performance characteristics in the case of exponential distribution, though in the case of other distributions binary tree showed worse results (Fig. 1).

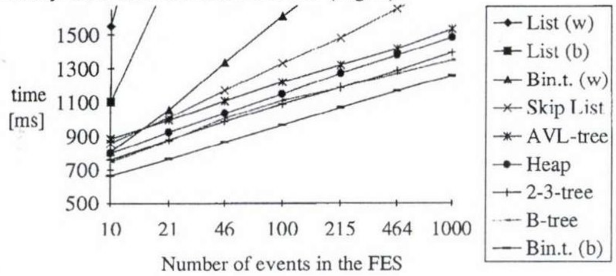


Fig. 1. Execution time of 1000 simulation steps, i486 DLC proc. (b = best; w = worst case)

On the other hand, skip list produced much better time characteristics than the balanced trees when advanced hardware floating point processing was used (Fig. 2).

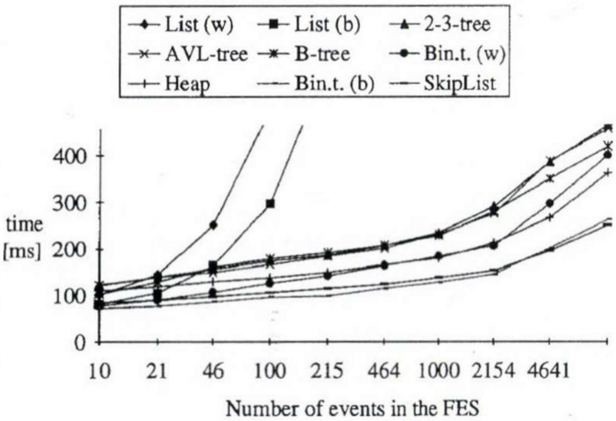


Fig. 2. Execution time of 10000 simulation steps, DEC ALPHA proc.

With strong floating point support, it is worth using skip list rather than balanced trees, especially because its algorithms are even simpler than that of the balanced trees.

4. CONCLUSION

We concluded that both the processor type and the distribution of the delay of the new events influence which data structure produces the best time properties. Though balanced trees proved to be quite good, heap and especially skip list was found to be significantly better on modern processor architectures.

REFERENCES

[1] Reeves, C. M.: "Complexity Analyses of Event Set Algorithms", *The computer journal* 27. No. 1, pp. 72-79, 1984.

[2] Pugh, V.: "Skip Lists: A Probabilistic Alternative to Balanced Trees" *Commun. of the ACM* 33, No. 6, pp. 668-676, 1990.

TRACE ANALYSIS AND THE ISO-9646 FRAMEWORK

G. ZIEGLER

DEPARTMENT OF TELECOMMUNICATIONS AND TELEMATICS, TECHNICAL UNIVERSITY OF BUDAPEST
KFKI-MSZKI RESEARCH INSTITUTE OF MEASUREMENT AND COMPUTING TECHNIQUES, DEPARTMENT OF INFORMATICS
E-MAIL: ZIEGLER@TTT-ATM.TTT.BME.HU, ZIEGLER@SUNSERV.KFKI.HU

This paper is a summary and overview of the work published earlier in [4], [5]. The main focus is on the applicability of the trace analysis method to the standardised protocol conformance test methodology [3]. We found that the idea of integration of trace analysis into the ISO 9646 framework seems to be useful and desirable. A preliminary study about a trace analyzer under design is included.

1. INTRODUCTION

The modern life in the "Information Age" requires huge quantity of information transfer every day, such as e-mails, bank transfers, telephone calls, etc. Every time a non-human entity is involved in the communication process, some kind of communication rule-set is required (i.e., a communication protocol). Widely accepted, respected, and vendor-independent protocol standards are required to make possible the communication with any machine regardless their particular vendor. Systems conforming to such requirements are called *Open Systems*.

This kind of systems has to be tested in order to check whether they really conform to the standards. This is a preliminary requirement before a new entity can safely be connected to a network of Open Systems. From this point of view, standardised test methods are very important for providing reliable, repeatable, and comparable results [2]. However, in certain situations they cannot be the best solution. A promising alternative method is the trace analysis. In this paper we try to enlighten some of the benefits of this method together with its suitability for the standard ISO conformance testing framework [2].

In Section 2 a summary of the method is given together with the discussion of its advantages and disadvantages. In Section 3 an outline of a planned trace analyzer is shown. Finally, in Section 4 a short conclusion is written down.

2. THEORETICAL BACKGROUND

The base idea of trace analysis is to inspect a sequence of observed input-output events (called a *trace*) to determine whether it is acceptable with respect to the *reference specification* of the protocol being investigated. During ISO 9646 conformance testing the process is controlled by a tester entity that obeys the instructions of the test suite. This suite consists of test cases, which in turn consist of *test sequences*. The test sequence specifies the prescribed sequence of *test events* that are to be observed to qualify the outcome of the *test campaign* as *PASS*. It also contains all the possible alternative sequences of events together with the appropriate qualification of the test outcomes as *INCONCLUSIVE* or *FAIL*.

In other words in the ISO 9646 testing method the designing of a test is done simultaneously with the classification of the possible test outcomes. It causes the test suites to be infeasibly long, which in turn results in low reliability (since several standard test suites were generated manually). The approach of the trace analysis separates the test selection process and the decision making regarding the behaviour of the Implementation Under Test (IUT) [1]. The separation of the test generation/selection and the verdict assignment makes possible the automation of the judgement about the response of the IUT. However, it requires a formal specification of the protocol. Not every current protocol possesses such a specification. The analyzer should be able to internally model the protocol; in other words it should contain the reference model of the specification [1], [3]. If the formal specification of the protocol is missing, then the designer of the particular analyzer module has to create the machine analysable reference model from the informal specification.

3. THE TRACE ANALYSER AND THE TEST DRIVER

The ISO 9646 methodology is widely accepted and used despite its problems described above. Therefore, when we want to improve the standard we should keep the necessary changes at a minimum level to save others' work that is already done. We want to reuse the already published standard test suites with the least necessary changes and preserve the most possible from the methodology. Hence, our goals are the following: to replace somehow the ISO 9646 tester with a trace analyzer making possible to leave the handling of unexpected events out of the standard test suites. This will decrease the size of the test suites and also increase their reliability. The rest of ISO 9646 should be left intact.

A trace analyzer — by its nature — is a passive, observing entity. Before we can replace with it the ISO 9646 tester we have to solve the problem of controlling the test campaign. A possible solution is the introduction of a new entity — a so called *test driver* [5]. It should be able to conduct the test campaign and to recognise the unexpected test events. The classifications of these events are no more included in the test suite, but rather based on the error indication of the trace analyzer. This requires co-operation between the test driver and the trace analyzer. The rule of operation is shown in Table 1.

Table 1. Evaluation of the results

During the test campaign the trace analyzer...	The opinion of the test driver about the test purpose at the end of the test campaign...	Then the verdict (according to ISO-9646) is the following:
indicates faulty behaviour	*(irrelevant)*	FAIL
indicates faultless behaviour	fulfilled	PASS
indicates faultless behaviour	not fulfilled	INCONCLUSIVE

4. CONCLUSION

As shown in details in [4], [5] the test driver and a trace analyzer together can replace the traditional ISO 9646 tester. They can be functionally equivalent replacement and the explicit handling of unexpected events in the standard test suites can be omitted. This can result in shorter and more reliable test suites.

REFERENCES

[1] G. von Bochmann, R. Dssouli and J. R. Zhao: "Trace Analysis for Conformance and Arbitration Testing", *IEEE Trans. on Software Eng.*, Vol. 15. No. II, pp. 1347-1356. November 1989.

[2] International Standard ISO/IEC 9646-1: 1991(E): *Information Technology – Open systems Interconnection – Conformance testing methodology and framework* Part 1: General Concepts.

[3] R. Wvong: "LAPB Conformance Testing Using Trace Analysis", *11th int'l Symposium on Protocol Specification, Testing, and Verification*, pp. 248-261. June 1991.

[4] Ziegler G. and Miskolczi J.: "Új módszerek a protokoll alkalmasságvizsgálatokban: eseménysorozat-elemzés", *Híradástechnika (Journal on Communication)* XLVII. évf. 1996. June-July. pp. 47-52.

[5] Ziegler G.: *Protokollok konformancia tesztelése eseményelemző módszerrel*, (A dr.univ thesis submitted for acceptance to the Technical University of Budapest, Hungary, 1996.)

BLOCK-BASED DISPARITY ESTIMATION

P. DEMETER and A. GAZDAG

DEPARTMENT OF TELECOMMUNICATION AND TELEMATICS, TECHNICAL UNIVERSITY OF BUDAPEST
E-MAIL: DEMETER@TTT-202.TTT.BME.HU, GAZDAG@TTT-202.TTT.BME.HU

For many applications, a stereoscopic video communication has been found useful. A stereoscopic imaging system provides two images from two cameras. A system like this generates a viewpoint fixed by the positions of the two cameras. But a true 3D system provides unlimited number of viewpoints. To create different viewpoints we need to know the depth information of the objects on the image. This depth information can be estimated by several methods. In the following sections we are going to introduce one of these methods.

1. INTRODUCTION

In the ideal case there are only horizontal differences between the objects on the left and right images, depending on their depth. This displacement is called disparity. The disparity information of two images is called disparity field. The multi viewpoint capability can be achieved by interpolating intermediate viewpoints using stereo image sequences and their disparity field. We investigated several stereoscopic sequences with the estimation method called full search.

2. THE BLOCK-BASED FULL SEARCH METHOD FOR DISPARITY ESTIMATION

In this method we divided one of the images (for example the left one) into blocks of given size. We tried to find the block in the right image that is the most similar (similarity was measured based on the Euclidean distance) to the chosen block of the left image. This method is called full search because all possible blocks are considered in a given range. This method is the simplest, but due to its simplicity the results are generally not as good as that of other methods. We tried to improve the results by applying pre- and post processing algorithms.

3. PREPROCESSING

Because the two cameras are not identical, there will be statistical differences in the luminance characteristics, which cause problems in recognition of the most similar blocks. We used an algorithm to equalise mean and variance of both images by modelling the differences between the two cameras as a linear relationship. We improved this algorithm because the original formula treated the occlusion areas as if they were camera differences. In our algorithm the images are divided into adjustable number of identical size partitions that are treated separately. In this way, the large occlusion areas (occluded regions are spatially coherent groups of pixels that can be seen in one image but not in the other) in one part of the image cannot interfere with the transformation of the other parts.

4. POSTPROCESSING

Because the errors are generally present in isolated blocks, we adopt various filters on the disparity data. In the following we will describe the effects of filters on the AQUA stereoscopic sequence.

When a median filter was used we experienced a significant improvement on the interior surfaces of the objects (for example on the striped fish). On the left side of the big rock there were less discontinuity. The same improvement can be experienced at the meeting of the background and on the objects in the foreground. Without applying a median filter, at the borders of the objects we experienced an unambiguous blocking effect, while with applying the filtering the borders were dizzy (improved the subjective quality).

The smoothing filter also improved the objects' interior region. The edges were better than with the median filter, but after thorough examination we can discover other kind of errors. The edges looked blurred and the regions close to the edges were also misty. At the borders of the objects the blocking effects disappeared, the filter smoothed them away (it looked like that the images were fuzzy).

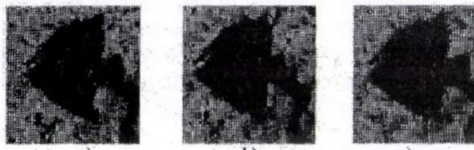


Fig. 1. A part of the interpolated image of the AQUA sequence, a) with no filtering, b) with median filter, c) with smoothing filter

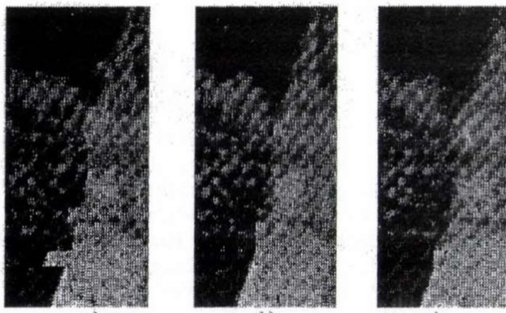


Fig. 2. A part of the interpolated image of the AQUA sequence, a) with no filtering, b) with median filter, c) with smoothing filter

Comparing the effects of the two filters: by using the smoothing filter we generally got better results, mainly at the borders of the objects. Fig. 1 and Fig. 2 show parts of the interpolated image of the AQUA sequence. We asked a few non-professional persons about their opinion on the two interpolated images; everyone preferred the images with the smoothing filter to the images with the median filter. On the other hand, inside the objects the usage of the median filter seemed to be useful.

5. COMPARING REAL AND ARTIFICIAL IMAGE PAIRS

If we used artificial image pairs, the results were quite fascinating (of course we used images free from problematic regions, like large occluded areas and repetitive patterns). The luminance equalisation and the filtering did not cause any noticeable improvement.

6. CONCLUSIONS

After several examinations, we can conclude that the luminance equalisation is necessary to achieve acceptable results. Our modifications on the equalisation algorithm also looked useful and the required computing time is not increased. We can also say that the filtering of the disparity data can improve the result, but it requires thorough considerations to decide what kind of filtering should be adopted.

The applied block size is also variable and this can cause significant changes in the results but unfortunately there is no general block size for all possible stereo image pairs.

ACKNOWLEDGMENT

We would like to thank our mentor, Gyula Marosi for his assistance.

TESTING MESSAGE TRANSFER PART (MTP) LEVEL 2 FOR SIGNALLING SYSTEM NO. 7

V. V. PHAN
FACULTY OF ELECTRICAL ENGINEERING AND INFORMATICS
TECHNICAL UNIVERSITY OF BUDAPEST
E-MAIL: PHAN@BME-TEL.TTT.BME.HU

The Message Transfer Part (MTP) provides a correct delivery of User Part signalling information for Signalling System No. 7 in both reliable and unreliable network environments. To handle the wide range of applications, a number of new features were developed and incorporated in the MTP level 2 and 3 Recommendations. This paper reviews the current status of the MTP level 2 Recommendation with its own major features and presents some results on the protocol conformance test for the MTP level 2 Signalling Link.

1. INTRODUCTION

Since the first appearance of the Store Programming Control Exchanges a number of telecommunications services (e.g. abbreviated dialling, wake-up service, multiple subscriber number, call forwarding on busy, on no answer or unconditional, malicious call identification, call completion to busy subscriber, etc. have become possible. It is widely realized that the efficent provision of telecommunications service requires an advanced signalling protocol in which maintenance and control information can be conveyed. SS No 7 is the first internationally accepted standard for Common Channel Signalling (CCS) in which signalling information is transfered along a common signalling path in the form of labelled message. The most basic building block element of the SS No 7 system is the Message Tranfer Part for a correct delivery of User Part signalling information. Therefore, the conformance testing of the MTP protocols plays a very important role in the correct implementation of the SS No 7 standard.

The conformance testing procedure comprises three phases:

- the first one is the test case preparing phase,
- the second one is the conformance testing phase,
- the third one is the result analysis phase.

This paper presents the method to specify the abstract test suite for the MTP level 2 with concrete suppelementary examples carried out by the author. The rest of this paper is organized as follows. In the next Section, MTP level 2 is briefly described, then the test specification phase in the standardized Tree and Tabular Combined Notation is explained. Finally, there are some remarks concluding the paper.

2. MTP LEVEL 2

Recommendation Q.703 defines the specification of Level 2 Signalling Link whose functions include: signal unit delimitation, signal unit alignment, error detection, error correction initial alignment, signalling link error monitoring, flow control. The signalling link functions with the coordination of the link state control provide the reliable transfer of signalling messages between two directly connected signalling points in the data link.

Signalling messages delivered by superior hierarchical levels are transferred over the signalling link in signal units of variable length. The signal units include transfer control information for proper operation of the signalling link in addition to the signalling information.

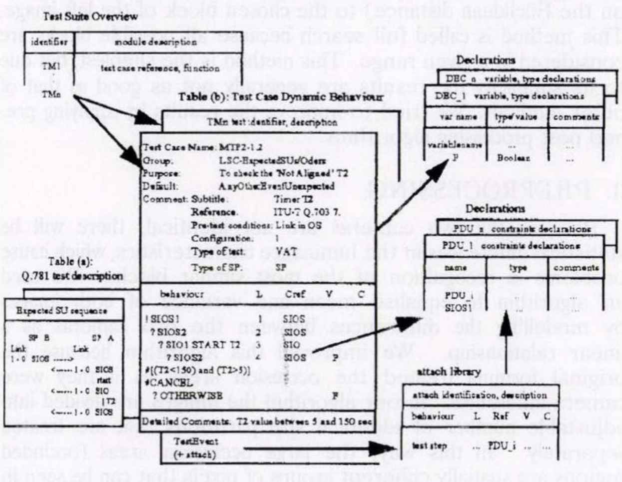
3. TEST SPECIFICATIONS

The basic concept of test methods is defined in recommendations. However, the conformance test suits have not been specified yet in the abstract form using the Tree and Tabular Combined Notation (TTCN). Therefore, we would like giving a method to rewrite the description of the test suits to the TTCN form in accordance with Recommendation Q.781.

The TTCN specifies a test suite in four part: test suite overview, declarations part, constraints part, and dynamic behaviour part. Because the abstract form is not in executable form, it is necessary to translate it into executable one before testing by some test suit generator program.

The test suits have a hierarchical structure: every test case has a defined test purpose, which is shown in the overview part of TTCN test suite. The test cases are grouped with the others to form a test group or test subgroups according to basic function of the protocol generally. The test case may be decomposed to test steps and the test step into test events. The test step comprises at least one test event; the test step may be attached statement, which used for modularize test tree (test case) by acting as call procedure. These attach statement symbolized by '+' is used to specify that a particular test step is to be attached into the test case (into the particular point of the test tree). One attach could contain other attach call may be defined locally or globally. The sequence of test steps (attaches and series of test steps) can be grouped too, and forms a test sequence to put the Implementation Under Test (IUT) from an actual state to a required state from which the test body starts (preamble) or to put the IUT to starting state from the end of the test body.

The process of the specification of test suits in the TTCN form is illustrated in Fig. 1. In Table (a) the description of the test case are given in a human-readable form, but not in the TTCN form. In order to carry out the conformance testing phase this a human-readable form is converted to the TTCN form in Table (b).



C:	Comments	SIO:	Status Indication Out of alignment
L:	Label	SIOS:	Status Indication Out of Service
OS:	Out of Service state	SP:	Signalling Point
PDU:	Protocol Data Unit	SP A:	Signalling Point A under test
Ref:	Reference	SP B:	Signalling Point B tester
V:	Verdict	VAT:	Validation

Fig. 1. Illustration of abstract test suite in TTCN

4. CONCLUSIONS

The contribution of the work presented in this paper is the description of the conformance test suits in the TTCN form in accordance with Recommendation Q. 781. Some test results are also presented. In the future some works will be carried in order to efficiently facilite the protocol conformance testing procedures.

ACKNOWLEDGEMENT

The author would like to thank Prof. K. Tarnay, T. V. Go, and J. Miskolczi.

PERFORMANCE ANALYSIS OF B-ISDN SIGNALLING MESSAGES

P. ABDOLRASOUL

DEPARTMENT OF TELECOMMUNICATIONS AND TELEMATICS
TECHNICAL UNIVERSITY OF BUDAPEST
E-MAIL: RASOUL@TTT-ATM.TTT.BME.HU

This paper examines the performance analysis of Broadband-ISDN signalling messages in comparison with Narrowband-ISDN messages. It analyses the bandwidth utilization and the cell distribution of signalling messages in Asynchronous Transfer Mode (ATM) networks, the accepted technology for B-ISDN.

1. INTRODUCTION

Signalling in an ATM-based integrated services network has an increased complexity, when compared to single-service networks (packet data or telephony). Connections belonging to different services have to be controlled not only during call setup and release, but also for changing service caracteristics within the call duration. This situation becomes more complex in the broadband case with the provision of even more services and a wide range of negotiable transfer rates and Quality of Services (QoS) parameters offered to the user.

A performance analysis for signalling is presented using discrete queueing models in. Also processing and transmission delay encountered by signalling messages at the AAL and ATM layer is presented in that paper.

2. BANDWIDTH UTILIZATION

Due to the growing number of services and applications in ATM networks, the lenght of the signalling messages is becoming longer. So the maximum lenght of the B-ISDN User Part (B-ISUP) messages is becoming much longer than that of N-ISUP messages.

We define the bandwidth utilization for signalling η as the ratio of pure payload size PPS in bytes divided by the total length in bytes of the ATM cells ($n \times 53$ octets) used for its transport, i.e.

$$\eta = \frac{PPS}{n * 53},$$

where n is the number of ATM cells, when using the segmentation of messages.

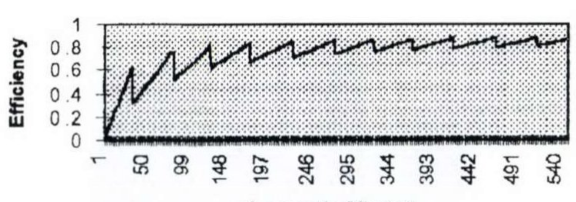


Fig. 1. Bandwidth utilization for segmentation into ATM cells

The maximum points of this curve in Fig. 1 correspond to the case when the PPS fits exactly in a number of cells (no padding bits). Minima correspond to the worst case of padding, when for the last one bit of the PPS we have to use an other cell. The minimum and maximum points converge to the same value, as the lenght of the messages increases. The maximum efficiency of bandwidth utilization is limited to a value of 0.86, because of the presence of ATM cell header.

Appart from the new information elements introduced in B-ISDN signalling messages and especially in the SETUP message, the main difference with N-ISDN is that signalling messages will be issued much more frequently. The length of the corresponding signalling messages at the UNI interface is nearly equal to that at the NNI interface. For e.g. the SETUP message consist of 5 or 6 cells, the corresponding message to IAM at the NNI interface.

Experiences show that the rest of the B-ISUP messages (eg. SUS, RES, SAM, RSM, RAM, etc.) appear with a probability of less than 10 % of that ones from Table 1., when there are no congested links on the network. So we can neglect their effect on our calculation. The lower layers add 20 more bytes to the length of B-ISUP messages when using AAL5 protocol.

Table 1. The lenght of NNI messages (in cells)

Message Type	Min length (byte)	Max. length (byte)	Nr of ells (Min.)	Nr of cells (Max.)	Nr of cells (Default)
IAM	94	528	3	12	5
IAA	32	35	2	2	2
IAR	27	29	1	1	1
ACM	26	286	1	7	3
CPG	27	268	1	6	2
ANM	44	345	2	8	3
REL	22	229	1	6	2
RLC	23	26	1	1	1

The minimum length of messages is calculated only with the mandatory information elements in the message, while the maximum length include all the optional information elements in the corresponding message. Of course, this situation never happens in reality, so we have chosen a default length for every message, according to our experiences (see last col. of Table 1.).

3. PROBABILITY DISTRIBUTION OF MESSAGES

According to minimum and maximum lenght of the messages presented in Table 1. we can assume a probability distribution of the signalling messages. An N-ISDN signalling frame is not longer than 192 octets, while a B-ISDN signalling frame can achieve 300 – 400 bytes.

Assuming a Poissonian distribution to the call attempt generation, we have obtained a cell distribution presented in Fig. 2 (in case of default value). Similar results for data transmission can be seen in, presenting the cell burst histogram for both TCP and UDP protocols.

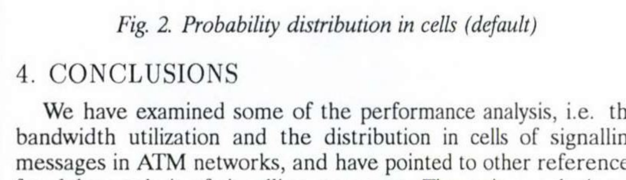


Fig. 2. Probability distribution in cells (default)

4. CONCLUSIONS

We have examined some of the performance analysis, i.e. the bandwidth utilization and the distribution in cells of signalling messages in ATM networks, and have pointed to other references for delay analysis of signalling messages. The main conclusion is that signalling messages in ATM networks have a little bit higher bandwidth utilization than in ISDN networks, and they are short packet sized messages, usually not longer than 5 ATM cells.

FULLY SOLID STATE FAMILY OF VHF-FM TRANSMITTERS

- Wideband design (87 – 108 MHz)
- MOS-FET power amplifiers
- Microprocessor controlled ALC and automatics
- Modular construction
- Active inner power reserve
- Protection against the effect of extreme terminations
- BITE (built in test equipment)
- Low stereo crosstalk and FM noise
- Controllable by PC from serial line (RS232)
- Eight stored frequencies
- N+1 storage system
- Traditional or switching power supply
- Cooling with indoor or putdoor air
- High reability
- Attractive price

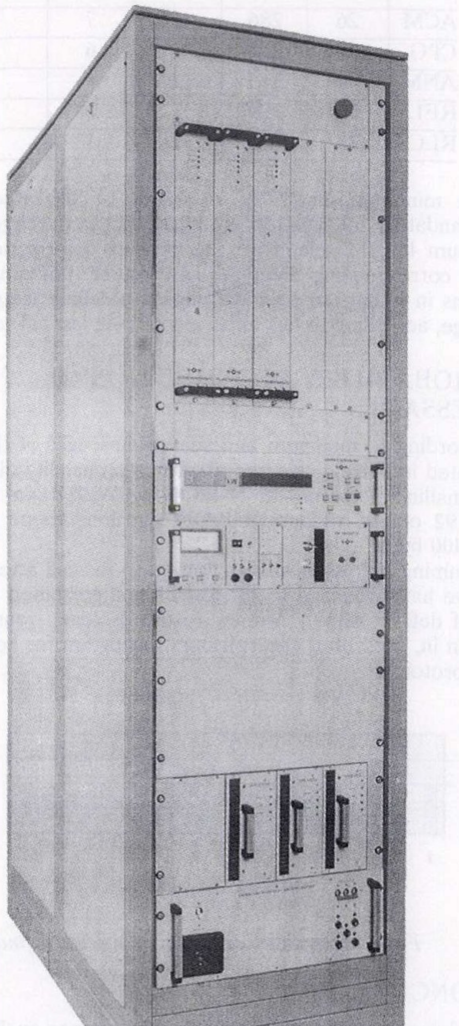


Fig. 1.

POWER OUTPUT: 1, 2, 3, AND 5 kW

Specification:

Frequency range	87.5. . . 108 MHz
Frequency set in	increments of 10 kHz
Operating frequency	can be pre set
Accuracy of frequency	± 300 Hz/year
Modulation mode	FM
RF output power	1, 2, 3, 5 kW
Output loading impedance	50 Ohm
Output VSWR for delivering (for nominal power)	≤ 1.5
RF output	Protected agains extreme terminations
Level of harmonics	≤ 1 mW
FM signal/noise (ratio for 75 kHz)	≥ 66 dB
Sinchronous AM	≥ 46 dB
AM signal/noise	≤ −50 dB
Impedance of modulation input	600 Ohm simm./ /asymm.
Modulation input level	−5 . . . + 10 dBm
Frequency band of modulation	30 Hz. . . 15 kHz
Frequency Response	±0.5 dB
Preemphasis	50 μs
MPX input frequencyband	30 Hz. . . 91 kHz
Distortion factor	0,3 %
Stereo crosstalk	
– 100 Hz. . . 5 kHz	≤ 50 dB
– 30 Hz. . . 15 kHz	≤ 45 dB
Mains voltage and frequency	3x230/400 V, 50 Hz
Mains voltage fluctuation	
– with traditional supply unit	−10 . . . + 5 %
– with switching type supply unit	−15 . . . + 10 %
Operating temperature range	+5 . . . + 45°C
Functional temperature range	−10 . . . + 45°C
Cooling	Forced outdoor or indoor air cooling

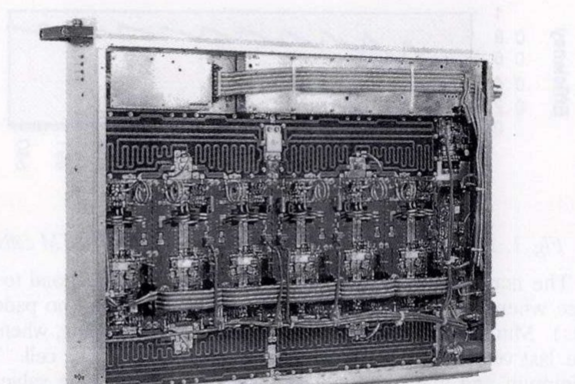


Fig. 2.

Utilizing the developing and manufacturing experiences of decades, the Antenna – BHG Ltd. (ABA Ltd.) has developed a new VHF-FM family of transmitters using most up-to-date semiconductors. The mechanical and electrical construction makes possible the developing of 1, 2, 3 and 5 kW power types generally accepted in the international practice using identical elements such as FM-modulator, power amplifier subrack of central automatics.

Power amplifier stages of the transmitter are wideband types (87. . . 108 MHz). Adequately low harmonic content is provided by low-pass filters.

A modular design of the mechanic structure ensures adequate flexibility for both the construction and operation.

Frequency modulated (FM) signals are produced by a PLL synthesizer FM modulator, in which high linearity, low noise circuit solutions are used. The output signal of the FM modulator is amplified to 20 W by a power amplifier modul containing appropriate protection and a harmonic filter at its output.

Modulator is provided with an internal stereo coder producing standard signals (MPX) having pilot signals. Due to the up-to-date technique of switching, very good damping of crosstalk can be achieved. Connection of the audiofrequency-signal may take place both symmetrically and asymmetrically.

The output signal of the FM modulator arrives to an 0-degree distributor network, where the signals for the power amplifier subracks are distributed.

The power amplifier of the transmitter is made of moduls generating 300 W power typically. The power amplifier moduls are mounted into subracks. One subrack contains six amplifier modules and one exiter amplifier. The amplifier modules are running parallel by means of a Wilkinson type 6/1 distributor-combiner network. Independent harmonic filter microprocessor controlled level control and automatics are belonging to a subrack. The subrack is designed to ensure appropriate cooling for the modules. The larger, 3 and 5 kW output power transmitters are established by parallel running of subracks, providing typically 1.3 kW RF power. The combination of the output powers takes place with 0-degree type Wilkinson combiner. The size of the combiner and also the number of its ports depend on the value of the power to be produced. At the output of the transmitter a direction-coupler is installed.

Operation, remote controllability, error signals, and long term stability of operation parameters are ensured by microprocessor type central automatics.

The control of automatics is of a menu system, and can be managed by push-buttons provided on the front panel or by remote control. The central automatics controls the amplifier subracks and set in the required output power. Central automatics ensure a most optimal utilization of active reserves inherent in the design.

Power supply for the equipment may be traditional, or switching type depending on demand. In case a traditional power supply unit is used smaller tolerance may be permitted on the mains side. The supply unit is placed on the bottom of the supporting structure. The equipment requires a mains connection of three phases. Auxiliary voltages necessary for the operation are produced in a heavy current distributor.

The transmitter is provided with a built-in cooler. Cooling air can be supplied indoors or outdoors as per specification of the order.

The equipment is housed in a welded steel frame.

J. Bereczki
ANTENNA – BHG
Transmitter Technics Co. Ltd.
H-1119 Budapest, Fehérvári út 70.

■ THE SCIENTIFIC DAYS OF PKI

The Scientific Days of PKI were held at the Óbudai Társaskör, in the calm neighbourhood of Óbuda on 19 and 20 November. The organizers have chosen the motto: *New Services – Technical Prerequisites*. In line with the motto the presentations were focused on the very timely topics of revision and development of the innovation process, the different aspects of the information revolution and the formation of the information society, supporting technologies, tarrifing and pricing of the services – and last but not least – quality.

The conference was opened by Ulrich Schaumann, MATÁV's Chief Technical Officer. The technical program started with the presentation of Christine Rogge about the revision and re-engineering of the innovation process at Deutsche Telekom. Gábor Frischmann, Development Director of MATÁV, was talking about the product development process at MATÁV. It seems that the meaning of R&D is changing and services — unlike in the past — are more and more considered to be a product, too. The speakers emphasized that the innovation cycle is becoming shorter, with the quality maintained or even improved.

John Breckenfelder, Chief Strategic Director of MATÁV dealt with the strategic priorities of technical development in his talk. He said that with the disappearance of waiting lists and emerging competition it is in the interest of the strategic investors of MATÁV to get into broadband business, especially into CATV. In connection with that John Anderson, Executive Director of MATÁV was talking about the optimal tarrifing of broadband services.

Several talks have dealt with the different aspects of the information revolution and the formation of the information society. I consider here the presentations which dealt with the convergence of telecommunications, informatics and entertainment, as well as those about the introduction of ATM and multimedia. Among these probably the most interesting one — at least to the author — was by István Lipp, Chief Marketing Officer of MATÁV, who was talking about the social impacts of technical innovations. The talks of Ferenc Farkas, Informatics Director of MATÁV — Telecommunications and informatics —, and Sándor Hegedüs, Deputy Director of Telecommunications Documentation Centre of MATÁV — Possible application of the Set-Top-Box in the access network of MATÁV —, were also within this scope. This general topic was discussed in the presentations on the ATM trial network of MATÁV, on the technical background of the video-on-demand application and on the compression coding for multimedia applications.

The most important supporting technologies were also dealt with at the conference. Umberto Ferrero from CSELT was talking on the results of some EURESCOM projects in the field of multiservice access networks. There were also presentations about the future transport networks — the so-called photonic transport network — and their key elements.

The Scientific Days of PKI reflected the change in the role of PKI within MATÁV. Presentations were held not only by the researchers of PKI, but also by the top managers of MATÁV. Almost every top manager responsible for technical or service development held a presentation. The conference was closed by Elek Straub, President – Chief Executive Officer of MATÁV.

The conference was connected via video-conference links to other locations. Thus the conference could be attended from Pécs, Sopron and Miskolc, as well as from the Zombori street laboratories of PKI. An ATM based video-on-demand demonstration was also in operation at the location of the conference.

Á. KAPOVITS
MATÁV PKI

■ BCN SEMINAR AND DEMONSTRATION

A two day seminar and live demonstrations were held on 15-16 October in Budapest by BCN Ltd. BCN is a network system integrator on the Hungarian market having developed partnerships with Motorola Information Systems Groups (ISG), PictureTel and Ascend.

Topics of the seminar were corporate networks, video conferencing, remote access solutions and ATM networks. In corporate networks greatest interest was focused on the 6560 multiprotocol router family of Motorola. This router can provide voice transmission on frame relay networks. 8 kbit/s compressed voice transmitted through a demo network could clearly be heard, even tone could be recognized. This technology has great opportunities in the future. Another novelty was the V.34 33.6 kbit/s modem from Motorola, offered also for corporate users.

An outstanding event of the video conference session was the world premiere of SwiftSite, a new portable equipment from PictureTel. PictureTel introduced the system in New York, concurrently with BCN Days. Participants of BCN Days could enjoy the event live through video conferencing. SwiftSite weights ca. 5 kg without screen, thus a TV set on the spot has to be used, but the equipment only needs a basic ISDN line, 220 V mains connections and the TV receiver. SwiftSite can easily be transported in the specially designed SwiftPort box.

Great interest was paid to the digital access systems of Ascent. The Pipeline 25-Fx seems to be ideal for SOHO (Small Office/Home Office) applications. This subscriber equipment connected to an ISDN BRI line provides Ethernet LAN and Internet access up to four users and two analogue telephone sets. The MAX family of Ascent is suitable for central systems, thus complete teleworking networks can be built. Even Internet providers can use the biggest element of the family, MAX 4000. This equipment is used by MATÁV, as Internet service provider for ISDN access.

In the ATM session, Csaba Szabó, executive director of BCN held a presentation on ATM launching strategies, MATÁV's plans on this field were introduced by Imre Abos and Ágnes Szente. At the end plans and programs of ITTK, a training centre managed by BCN were presented.

I. BARTOLITS