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Ericsson BCS 10 is a complete telecommunications system which can be bought at a counter and brougt home

The LM Ericsson Prize Winners 1985

This article is an excerpt from the speech given by Professor Gunnar Hambræus, chairman of the LM Ericsson Prize Committee, at the award ceremony of the 1985 LM Ericsson Prize, May 6, 1985, at the Royal Swedish Academy of Engineering Sciences, Stockholm. The prize, which consists of a gold medal, a diploma and at present the sum of 250 000 Swedish Kronor, was first awarded at the LM Ericsson centennial in 1976. According to the conditions governing the award of the LM Ericsson Prize it shall "be awarded to recognize an especially important scientific or technological contribution to telecommunications engineering made during the previous three-year period, or an earlier contribution whose significance has been acknowledged during the period".

The Prize Committee, which has three members, is wholly independent of the Board of Directors and President of LM Ericsson. It has sole responsibility for selecting the prize winner among the candidates, who have to be nominated before October 1 the year before the prize is to be awarded. If an achievement has been shared by two or three persons, these persons may be awarded the prize jointly.

The prize winners' presentations will be found on the following pages of this issue.

The 1985 LM Ericsson Prize is being shared by two men who have made important contributions concerning the analysis and transmission of human speech, *Professor Gunnar Fant* and *Dr. James L. Flanagan*.

Gunnar Fant, born in 1919, is Professor and Head of the Institute for Speech Transmission and Music Acoustics at the Royal Institute of Technology, Stockholm, Sweden.

James L. Flanagan, born in 1925, is the Director of the Acoustical and Behavioral Research Center at AT&T Bell Laboratories, Murray Hill, New Jersey, US.

Research into the nature of speech signals requires a range of knowledge that comprises much more than just engineering sciences. This need for lateral scientific knowledge makes research into human speech very different from most other scientific work. The branches of science concerned include most areas of telecommunications as well as other fields such as physiology, acoustics, phonetics and linguistics. Many of the new concepts in the theories and models propounded by the 1985 prize winners consist of the bringing together of results from different, apparently unrelated, fields of science into uniform and consistent descriptions and systems.

Both Professor Fant and Dr. Flanagan have been engaged in research on speech signals for more than thirty years, partly in collaboration, at the Royal Institute of Technology in Stockholm and at the Massachusetts Institute of Technology and AT&T Bell Laboratories in the US. Their pioneering work on the development of models for the production and recognition of human speech and the analysis of such models forms the basis for the rapidly ex-

Fig. 1
Gunnar Hambræus speaking at the awards ceremony of the 1985 LM Ericsson Prize

panding field of computerized speech analysis and speech synthesis.

Professor Fant's achievements cover a wide range. He has carried out theoretical and experimental studies of the mechanisms of speech production, for example by X-raying the speech organs in movement and by developing analogue acoustic models for speech production. He has applied quantitative experimental methods to a number of problems in phonetics and linguistics. Professor Fant has also developed methods and instruments for analyzing speech signals and for synthesizing speech by means of rule-based control of electrical, and lately computerized, models of the human speech organs.

Dr. Flanagan's achievements include investigations of properties of human hearing and the development of models for speech recognition and for the signal processing that takes place in the human ear. He has carried out theoretical studies of the capacity requirements for systems that are to transmit human speech and developed systems for digital encoding and transmission of speech. Dr. Flanagan has also made theoretical and experimental studies of the speech organs and developed models and systems for speaker identification, speech recognition and speech synthesis.

Professor Fant's and Dr. Flanagan's work forms the basis for the many systems for speech transmission, speech recognition and text-to-speech synthesis that have been developed during recent years. A number of these systems have found particularly urgent applications as aids for the handicapped. For example, children with a speech motory handicap can be taught to use text-to-speech synthesis as a means of communicating with the environment, and blind people can be provided with a way of reading newspapers.

However, such limited applications are not the main effects of the work of the 1985 prize winners. They have laid the foundation for all types of speech-based communication that includes both humans and machines, in the systems we know and use today as well as the systems that will be designed in the future.

On behalf of the Prize Committee, LM Ericsson and all those present I congratulate Professor Fant and Dr. Flanagan on this prize – a prize which is primarily a recognition and a proof of appreciation of many years of creative research into the analysis and transmission of human speech. I now ask you both to come forward and receive the prize for your extraordinary achievements from the hand of His Majesty King Carl XVI Gustaf.

Fig. 2
Gunnar Fant and James L. Flanagan with Gunnar Hambræus, in conversation with His Majesty King Carl XVI Gustaf after the ceremony

Speech Technology — Research and Development

Gunnar Fant

GUNNAR FANT

Gunnar Fant was born in Nyköping, Sweden, in 1919. In 1958 he defended his doctoral thesis in electrical engineering at the Royal Institute of Technology in Stockholm. Previously he had carried out speech research at Telefonaktiebolaget LM Ericsson in Stockholm during the years 1945–49 and at Massachusetts Institute of Technology in Cambridge, USA, from 1949 to 1951. During the period 1951–66 he was in charge of research at the Speech Transmission Laboratory of the Royal Institute of Technology, Stockholm. In 1966 Gunnar Fant received a personal professorship at the newly established Institute for Speech Transmission at the Royal Institute of Technology. The Institute was renamed in 1979 and is now the Institute for Speech Transmission and Music Acoustics.

Professor Fant is the author or co-author of over a hundred scientific books and papers, including some pioneering books on the nature of speech, such as Acoustic Theory of Speech Production, published in 1960, and Speech Sounds and Features, in 1973. Professor Fant is a member of the US National Academy of Engineering and the Acoustical Society of America. He is also a member of the Royal Swedish Academy of Sciences and the Royal Swedish Academy of Engineering Sciences and a honorary doctor at the University of Grenoble, France.

Speech technology comprises two basic fields. One concerns performance, or in other words system evaluation. How well does a transmission system, for example a telephone line, function when certain assumptions are made regarding the technical characteristics of the system and the type of messages and also imperfections in the hearing of the recipient and the speech of the caller? Such system aspects prompted my speech analysis work at Ericsson. The Swedish Telecommunications Administration is strongly represented in this field by Norman Gleiss.

Now, in the age of digital technology, the current problem is what data capacity in bits/s is required to provide satisfactory telephone transmission quality? Sophisticated coding systems that reduce the data requirement from a nominal value of 64 kbit/s to less than 10 kbit/s have been developed. The further we go in reducing the data requirement the more we have to know about the nature of speech. James Flanagan can tell us something about this.

The other main field is speech as a code with applications in speech-based information systems. Here the word code has a wider meaning than optimum signal processing. It concerns the relation between language and speech, how the building blocks of a message, i.e. sounds, syllables, words, intonation elements etc., appear in speech as sound waves. Our reference is the sound spectrogram, speech in pictures, fig. 1. In the spectrogram we can follow the whole sequence of an utterance with the time along the horizontal axis and the frequency along the vertical axis. The intensity of the spectral components is given by the degree of blackening. The picture shows characteristics bands and amplification areas, called formants. The formant patterns mirror the changes between voiced and unvoiced sounds and the movements of the speech organs, which cause a variation in time of the internal resonances in the throat, mouth and nasal cavities and thereby give the speech sounds their individual character. In the spectrogram we see a detailed structure of vertical lines which give the chronological order of the opening and closing of the vocal cords when pronouncing vowels and other voiced sounds. The periodicity of the vocal cord pulses gives the fundamental frequency of the voice. The closer the pulses, the higher the fundamental. It is the variation of the fundamental with time which we percieve as intonation.

I cannot now describe spectrogram interpretation in detail, but it is obvious that we have an analogy with picture analysis. Speech as pictures is more complicated for several reasons, and the available analysis techniques do not give an unambiguous result. The utterance shown in fig. 1 has also been analyzed using narrow-band technique, lower part of the figure, which reveals the harmonics pattern. The detailed structure then appears in the frequency domain and can be used directly for intonation studies. The appearance of the picture is much more dependent on the speaker and the linguistic framework than one would think.

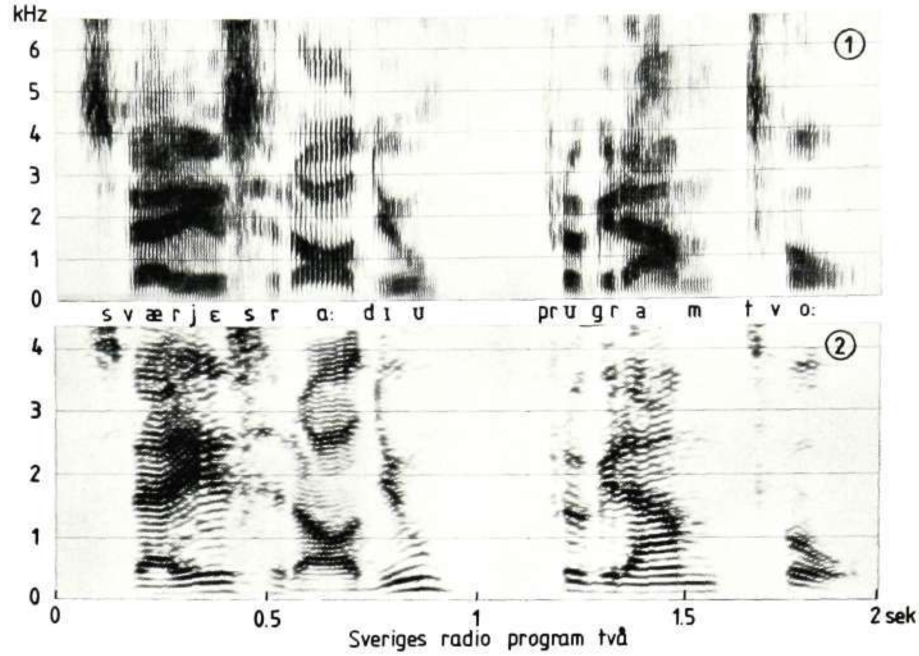
If we could break the code we would be able to convert writing into speech and vice versa, analyze the sound picture in order to find out what has been said. If we are to be able to do this, i.e. explore the language of visible speech, we must learn more about how speech is formed and how speech patterns are interpreted. A large part of my scientific work has concerned the relationship between the physiological stages in the production of speech and the formant patterns of the speech sounds.

The technical applications – speech-based information systems – mean that information is made available in the form of speech instead of, or as a complement to, text on a visual display unit. But also the converse, that information is fed into the computer by speech, instead of entering it from or as a complement to a keyboard. Together these two possibilities, speech out and speech in, mean that we can speak to the computer and receive a spoken answer.

The computer speaks – synthetic speech

What is meant by computer speech? There are two possibilities. One means that the computer reproduces a record-

Fig. 1
A sound spectrogram. The text is "Sveriges radio program två" (Sweden Radio program two). (1) Wide-band analysis, the formant pattern is apparent. (2) Narrow-band analysis. The harmonics structure appears



ed message in the same way we use tape recorders. Signal processing that reduces the amount of data may be carried out prior to the recording in order to save computer storage space. The technical term for this process is speech coding. In the case of large-scale data reduction, down to 1 or 2 kbit/s, an automatized analysis-synthesis is carried out with, for example, linear prediction. In order to obtain satisfactory sound quality it is then necessary to make a number of adjustments and to edit the analyzed speech before storing it in the computer.

This direct method is the one used in preprogrammed chips for speaking watches, dolls, computer games, lifts, cookers, cameras, dictionaries, and various warning systems in cars, process supervision etc. The technology has not been as widely used as expected, partly because the speech quality has been too poor. This disadvantage could be overcome by means of new generations of cheap, large digital memories. However, let us hope that it will not go too far. We might have environmental problems with wallpaper speech as a counterpart to wallpaper music.

However, in the long term the most important technique for speech-out from a computer is general text-to-speech translation. The computer must then be able to receive digital information consisting of ordinary writing, with normal spelling. It reads the text, which means that it first considers how the words should be pronounced. This is done by consulting a pronunciation dictionary supplemented by general pronunciation rules. The system then initiates a speech synthesis, which basically works in the same way as human speech organs.

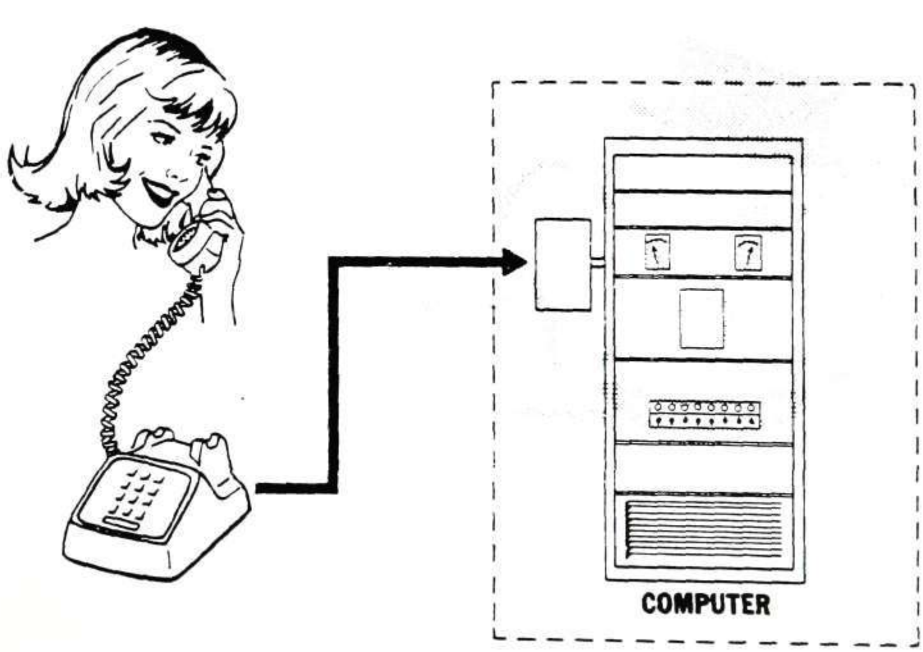
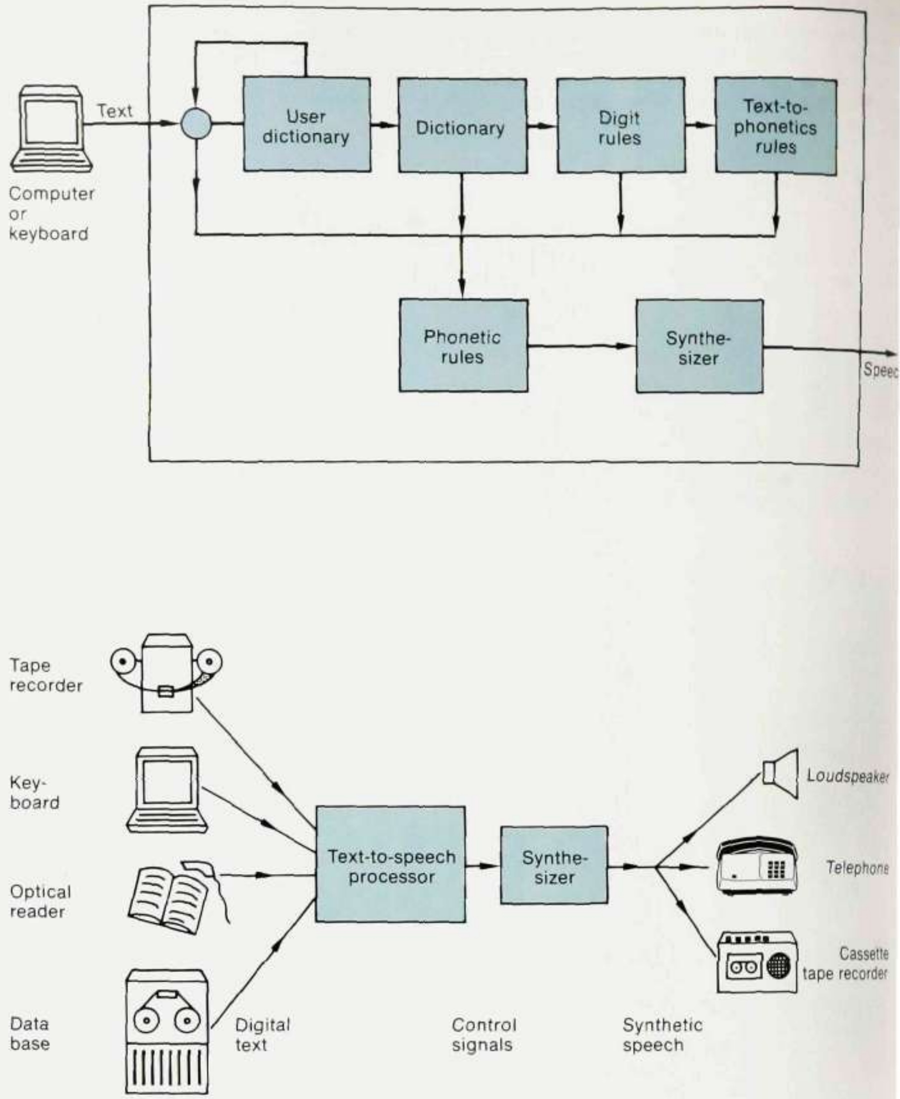


Fig. 2
To speak to a computer and receive an answer

Fig. 3
Block diagram of a text-to-speech translation system developed by Rolf Carlson and Björn Granström, with possible peripheral equipment



The computer can be provided with extremely detailed pronunciation rules, but it does not understand what it is saying and therefore it has a neutral and unengaged intonation. However, the most important requirement is that we understand what the computer says. Current systems have certain deficiencies, but one soon learns the specific dialect. The quality may not be good enough for public use, but this technique has meant a great advance in aids for the handicapped, for example work terminals and text reading systems for the blind, personal speech devices for dumb people and training equipment for children with reading and writing difficulties.

My colleagues Rolf Carlson and Björn Granström have developed a text-to-speech translation system that operates in six different languages: Swedish, English, German, French, Italian and Spanish. It is now available commercially, for example in the form of a printed board assembly for personal computers. We might not have the world's best English synthesis, but the multiplicity of languages is unique. Writing in the digits 1 2 3 4 5 6 7 gives the synthesis "one million two hundred thirtyfour thousand five hundred sixtyseven".

There are interesting analogies between synthetic speech and the visual arts. A spectrogram of a bit of speech is analogous to a photograph. Digital recording, storing and reproduction of speech is equivalent to video technique. A synthesis produced to copy a specific human utterance is analogous to a painting or drawing produced with access to the subject. Synthesis by rule as used in general text-to-speech conversion is analogous to producing a work of art without access to the subject, on the basis of memory and general artistic ability. In extreme cases it can become a caricature.

I will now demonstrate these alternatives, fig. 4. The first picture is the original utterance, the next a synthetic copy. Note that the synthesis was carried out at the Royal Institute of Technology as long ago as 1962, using the old analog OVE II. The quality of the fine synthesis demonstrates that our model of the peripheral speech organs was quite good already 23 years ago.

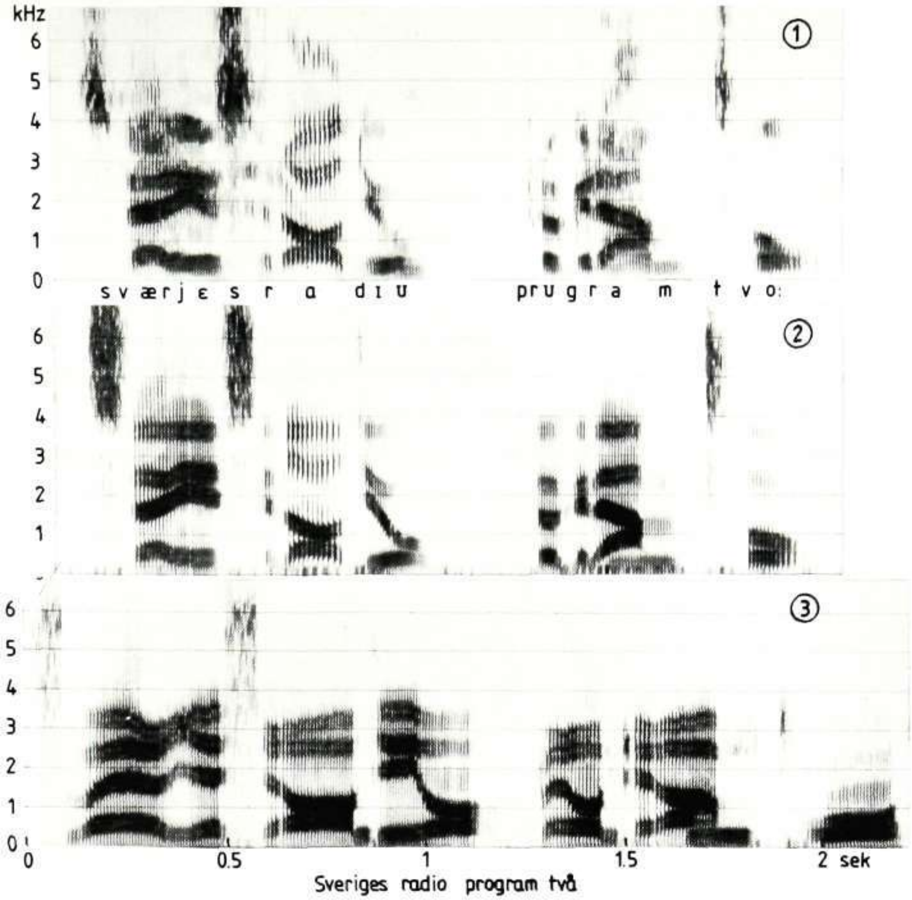


Fig. 4
(1) The original speech. "Sveriges radio program 2"
(2) Synthetic copy of 1. produced by OVE II in 1962
(3) Synthesis by rule of the same text, produced from the text-to-speech synthesis system in 1985

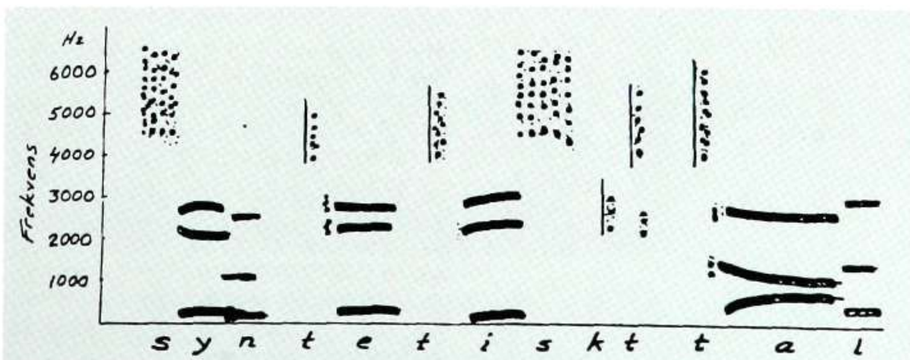
Next comes the synthesis by rule of 1985 of the same utterance. Why does it not sound as good? The answer is because the synthesis rules are not general enough. It is an immense task to define in a computer program our whole competence as readers. Much knowledge is still lacking, and we have not yet been able to incorporate all the knowledge we do have. The personal experience of the designer can be compared to the ability of an artist. It is there, but it is difficult to define after all the adjustments and modifications by instinct. We need more science and less art in the synthesis engineering. The same problem has concerned electronics firms, which have tried to improve the quality of speech with large-scale data reduction, stored in speaking chips with a limited vocabulary. They have engaged phonetics experts, who have busied themselves with the encoding, but with poor results. The speech production models in these speaking chips have not been adequate.

Historically speech synthesis have been an art form. The first speech machine was entirely mechanical. It was developed by Wolfgang von Kempelen, an Austrian, in the 1780s. The first electronic speech synthesizer was Homer Dudley's voder of 1939. It was controlled by a keyboard like that of a piano. Our first speech synthesizer at the Institute, OVE I, was completed in 1953. It is controlled by continuous hand movements, which simulate the movements of the tongue. The most important historical speech synthesis is probably Haskin Laboratories' pattern playback, in which drawn spectrographic patterns are converted into monotonous, stereotype speech. When I came to the US in 1949 I could, on the basis of my experience of speech analysis at Ericsson, create synthetic speech by painting a stylized formant pattern on a plastic sheet and then having it played back, fig. 5. This is an intriguing form of art, which might become popular with the computer buffs of tomorrow. If you can draw, you can make the computer speak. It might become a sport to try different combinations and simulate different types of voices.

The aims of the current development of speech synthesis are to improve the quality of the speech, to make it possible to simulate different types of voices and ways of speaking and to develop a feeling for the meaning of the text and thereby better intonation. The technique should then be applicable for public distribution of information.

Talking to a computer

The converse technique, speech-to-text translation, i.e. to talk to a computer and make it recognize what has been said, is



ig. 5
stylized sound spectrum

Fig. 6
Mr Speech Technology and the knowledge barrier

much more difficult. Present technique is mainly directed towards the recognition of individual isolated words. The systems can handle a large vocabulary, but they must be calibrated for each individual speaker. The basic principle is recognition of the pattern of a whole word at a time.

My colleagues Mats Blomberg and Kjell Elenius have developed a recognition system for individual words, or groups of words, which is now one of the best systems available in the market. It has been incorporated as an option in an intercom system developed by Ericsson. Instead of pressing buttons you simply say the name of the person you wish to be connected to.

The greatest efforts in speech technology are now being concentrated on more general systems which must be speaker-independent and be able to handle natural, continuous speech. The automatic speech-to-text printer will remain a utopian device, however, if we demand a human feeling for language of the computer. With certain restrictions as regards vocabulary and clarity of pronunciation it could become an important part of the automated office as a complement to word processing systems. We have already tested the technique, using individual commands for choice of format and other editing functions. Proofreading would be simplified if text-to-speech translation was available. The computer is never dyslexic and faithfully reads out what has been written.

There are many applications for speech-based information systems: telecommunication services, process control, supervision and inspection, teaching and handicap aids, including aids that enable people with physical handicaps to control their environment.

A question presents itself. Apart from applications for the handicapped, can we not be satisfied with the technology we have now? Is it not enough to use the computer keyboard and visual display screen? The counter argument is that for human beings speech is the natural way of communicating. When the computer speaks your eyes are free. When you speak to the computer your hands are free for other tasks.

International efforts

This is the argument that sells speech technology. Expectations are great. The international competition is increasing. Japan has announced its program for the fifth generation of computers, which are to be able to speak, understand and translate from one language to another. The United States has a large speech technology program in the DARPA project, mainly concentrating on recognition. Large computer and electronics companies have their own speech technology projects. IBM has long-term plans that concentrate on office automation. In England there is the Alvey project, which has, among other things, given more support to university research. Speech research is well advanced in France, and there collaboration is sought from researchers in artificial intelligence (AI).

The explosive development of electronics and computer technology has led to a great but often uncritical optimism. Now we have finally got the computer resources needed to solve complicated problems and to handle large amounts of data. There is talk of computer memories that can compete with the human brain as regards capacity, but this does not mean that we know how the brain works. Innumerable projects are being started by electronics firms, computer companies and telecommunications institutions with the purpose of solving all problems associated with recognition techniques within a few years. On the surface it is so simple. It is only a question of utilizing advanced signal processing and pattern recognition techniques in order to learn to recognize the speech sounds of the language and that is it.

This hubris is illustrated in fig. 6. Mr Speech Technology is happily and with high expectations wind-surfing towards the envisaged goal – the information society, where we can speak to computers as easily as we can to people. But he is not aware of the danger. He will soon founder on a lurking reef. It is the knowledge barrier. What we need is perhaps not a fifth generation of computers but a fifth generation of speech researchers with an interdisciplinary approach and long-term backing.

From an international point of view the Swedish work has not been great in quantity, but its quality has held its own. We have had continuous support from the Swedish Board for Technical Development and before that from the Swedish Technical Research Council. Altogether this support covers a period of 40 years. We have also had good collaboration with research in the arts and medical faculties and with Swedish industry, primarily Ericsson, and the Swedish Telecommunications Administration. ELLEMTEL now has a very active section working on speech recognition.

General speech recognition

Speech coding, and to a certain extent speech synthesis, are now established techniques, but more general speech recognition still constitutes a major scientific and technical challenge.

The spectrogram pictures of fig. 7 show the same utterance, "Hur Sverige danats" (How Sweden was formed), pronounced by two male subjects and by a young girl. Let us consider some of the problems. The sound picture made by the girl is difficult to interpret because of the high fundamental, which means that

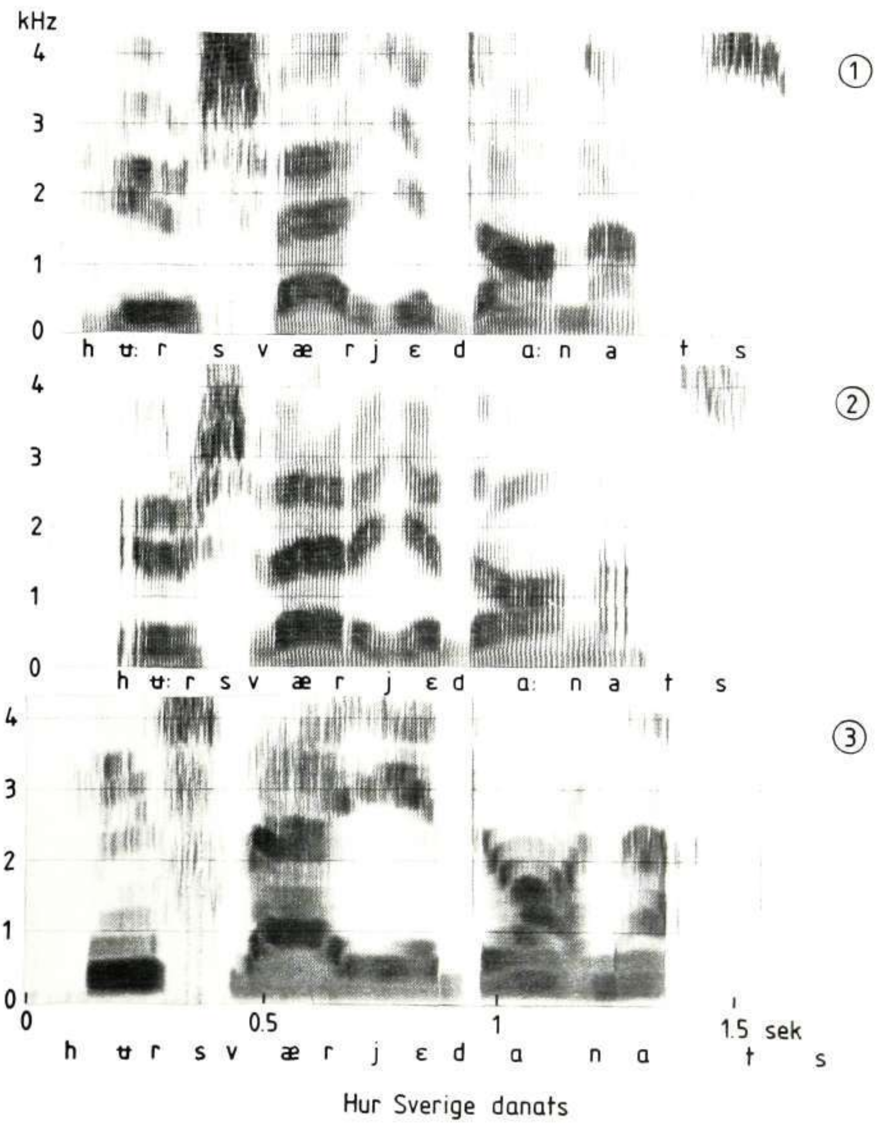


Fig. 7
(1) Male voice, (2) male voice (south-eastern dialect), (3) 10-year old girl

harmonics and formants are mixed up. The scaling factors for the formants are different, and the third formant is much weaker than for the male voices.

Acoustic segmentation is never unambiguous. It is impossible to find a break between væ and r in the word Sverige for speaker no. 1, but the segmentation is much clearer for speaker no. 2. There are obviously discrepancies between our abstract model of a message as a sequence of discrete speech sounds and the physical segments we can find in the spectrograms. *A single linguistic* segment can colour several successive sound segments, but the converse is also true: a single sound segment can carry information concerning several successive phonemes. The language is discrete but speech is continuous. We look for invariance but we end up in a morass of variability as regards context, specific speaker patterns etc.

The path out of this dilemma lies only to a limited extent in trusting to absolute invariance and attempting to structure the variability in rules. The rules comprise general knowledge of the structure of languages and pronunciation, for example the probability of sound sequences and modifications. We hear what we expect to be said.

Phonetically structured recognition, fig. 8, is then organized as interaction between a top-down prediction, *i.e.* linguistic expectation, and a bottom-up acoustic analysis, which gives a preliminary identification and basic data for verification. A model of the auditory functions is built into the primary analysis, and the linguistic prediction includes knowledge of speech production, which is realized in rules for text-to-speech conversion. General speech recognition and general speech synthesis are complementary and dependent on the same fundamental store of knowledge.

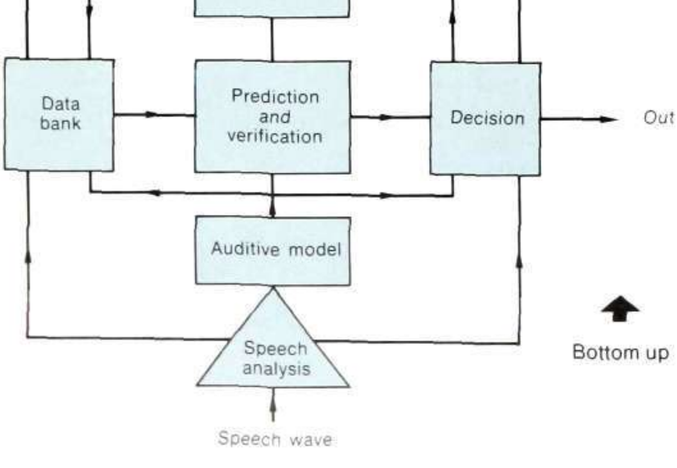
Speech research – an interdisciplinary science

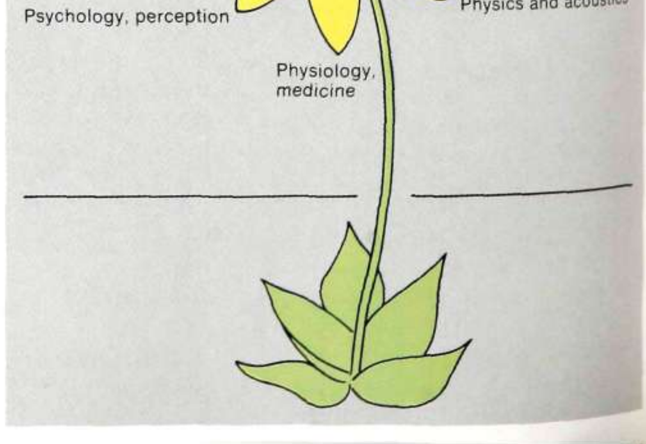
Speech research is an interdisciplinary assignment. Can we really let Mr Speech Technology and his engineers guide the development on their own? Of course not, but the problem requires large resources and interdisciplinary collaboration.

The flower of interdisciplinary science of fig. 9 shows the fields concerned in speech communication research: telecommunication engineering, computer technology with AI, phonetics, linguistics, pedagogy, psychology, physiology

Fig. 8
Sophisticated model for automatic recognition

Fig. 9
The flower of interdisciplinary science





and physics. However, we will not get far by just delegating problems to the various fields and then compiling the results into a finished product. Neither is it sufficient to share our knowledge. We must cross over and work actively in adjoining fields.

In technical speech research it is seldom *possible to obtain ready-made software* from linguistics and phonetics. Hitherto we have had to prepare it ourselves or at least adapt available knowledge to our system solutions.

There is a strong reciprocal trend in phonetics. Linguistics and phonetics have acquired good computer resources, and posts for engineers now exist in most phonetics institutions of any consequence. Phoneticians have also started to consider technical problems, such as synthesis by rule and automatic recognition, partly in order to tie in with real requirements. Another incentive is that the objectives of synthesis by rule and automatic recognition create new opportunities for academic research concerning language and speech.

We engineers not only have the motivation to design systems that work. We are also curious about language, speech and hearing and find it very worthwhile to contribute to the fundamental knowledge and to convert our findings into aids for the handicapped and other applications. The pendulum has swung over. Increased specialization is being compensated by a trend towards interdisciplinary integration.

However, such integration is not free from problems. It is easy for technicians to put product development before broader knowledge and to choose technically smart solutions that are economical but which limit the possibilities of development as regards research. A tool or an algorithm can become an end in itself to a technician, whereas a linguist might be more motivated by a philosophical problem that has to be solved, than by a contribution to a real project.

We have innumerable examples of such diverging objectives, but both technologists and humanists must have the freedom of self-realization. The effect is often positive, whether it is a matter of new technical tools or stimulating theories.

Finally some thoughts on computer technology and speech research. Demands are made for advanced equipment and high-level languages, which require not only money but also time for adjustment. The fundamental question remains. Should we first learn all about speech and then teach the computer, or should we let the computer create its own knowledge and intelligence for the recognition of speech? The latter model, in a simplified form, has dominated the development of systems that recognize patterns. The designers have let go their grip on the visual reference, speech as pictures and the phonetic code.

There is now an obvious reaction in favour of phonetic principles. For a few years MIT has been arranging summer courses in spectrogram reading. A skilled spectrogram reader can interpret a short phrase in, at best, a few minutes. But it can also take hours, and sometimes the reader gets it wrong. How can we then expect a computer to manage what we experts with our linguistic sense and phonetic knowledge cannot do? This is a challenge that has to be met with patience and new, more efficient strategies for integrating knowledge and computer technology. What we learn about language and speech will also benefit the text-to-speech conversion.

It is so easy to speak. A small child masters what all wise men in the world do not quite understand. That, Mr Speech Technology, is the problem!

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New Benefits from Information and Communication Technologies

James L. Flanagan

JAMES L. FLANAGAN
James L. Flanagan received the Sc.D. degree in Electrical Engineering from Massachusetts Institute of Technology, Cambridge, USA, in 1955. He joined Bell Laboratories in 1957 where he worked extensively in voice communication and digital signal techniques. From 1967 to 1985 he was Head of the Acoustics Research Department at AT&T Bell Laboratories. He is presently Director of the Acoustical and Behavioral Research Center at AT&T Bell Laboratories. Dr. Flanagan holds over 40 patents in speech processing and digital techniques and is the author of a landmark book, *Speech Analysis, Synthesis, and Perception*, published in 1965. He is a member of the US National Academy of Engineering and the US National Academy of Sciences, past chairman of the IEEE Acoustics, Speech, and Signal Processing Society and a fellow of the Acoustical Society of America and of the IEEE.

Dr. Flanagan's contributions have included the study of the auditory characteristics and the development of models for speech perception and for the ear's signal processing; theoretic treatment of the requirements on the information transmission capacity in speech transmission and the development of digital coding systems. He has also performed theoretical and experimental studies of speech production mechanisms and developed models and systems for speaker identification, speech recognition and speech synthesis.

It is with great pleasure, and I must say humility, that I join with you for this 4th presentation of the LM Ericsson Prize. It is a singular honor to be named for this recognition, and I am most grateful to LM Ericsson and to the Prize Committee.

The honor is made still greater by sharing it with my long-time friend and research contemporary – Professor Gunnar Fant. My admiration and esteem for Gunnar's many contributions to speech communications are well known, and reach back over 25 years – even to his doctoral thesis defense at the Royal Institute of Technology. It is, therefore, a unique pleasure to accompany Professor Fant in this honor.

In response to presentation of the LM Ericsson Prize, I would like to direct some remarks toward the outlook and evolution of speech communications technology. What new benefits can we anticipate? What are the promises of fundamental research now in progress?

Human speech communication

In drawing this perspective it is useful to recall the ingredients of speech communication between humans. Fig. 1 shows that speech generation begins with a message formulation – an idea at a central, cortical level. The idea is cast into a sound sequence representing the distinctive and meaningful sounds of the language – the phonemes. Neural signals activate articulatory muscles that control the shape of the vocal tract (and hence its acoustic resonances), and turn on and off the source of sound, primarily the vocal cords. The result is the acoustic speech wave whose spectral variations bear the spoken intelligence.

At the ear of the human listener the acoustic wave is analyzed into its characteristic frequencies, and neural processes extract features representative of the sound sequence. Message comprehension follows.

Through experiments we can estimate the information rates (in binary digits/second, or bits/s) needed to describe the speech information as it is represented at each internal level. The description is more compact and efficient at the higher, more central levels – being of the order of 50 bits/s. The rate increases to the order of 2000 bits/s at the articulatory and feature levels (and includes additional personal characteristics), and to a still higher rate at the external acoustic-waveform level.

In face-to-face conversation the speech wave is transmitted acoustically, and this works fine over short distances. But for communication at a distance, electrical transmission is needed, and the digital techniques for this represent the technology of speech coding. Our traditional methods for digitally coding the speech wave require about 60 000 bits/s – a rate much higher than the 50 to 2000 bits/s required in the internal representations – indicating that we are employing much more transmission capacity than is really needed. In other words, we need to know better how to extract and encode the intrinsic information in the acoustic wave. We will return to this subject presently.

Human/machine speech communication

As our telecommunication networks increase in sophistication, we find computational capability and digital memory distributed at many levels. This capability for storage of large amounts of information, and for high-speed logical decisions, makes possible new information and communication services for humans. But humans must be able to communicate with the machines. Speech is the most natural and convenient means of communication for humans. Hence, we wish to give machines the functional abilities of speech and hearing. The techniques of speech synthesis allow machines to assemble messages and speak them to people. The techniques of speech recognition allow humans to control machines by spoken commands. The interest in providing these human-like abilities to communication networks and to information systems is illustrated in fig. 2.

Speech coding, synthesis and recognition

The generic field of digital speech processing therefore embraces three technologies which typically are dis-

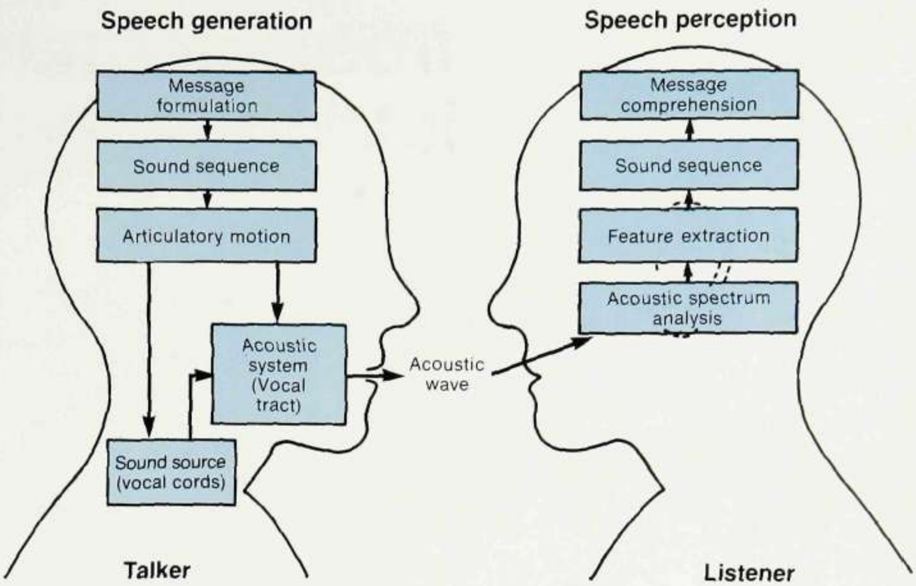


Fig. 1
Elements of speech communication between humans

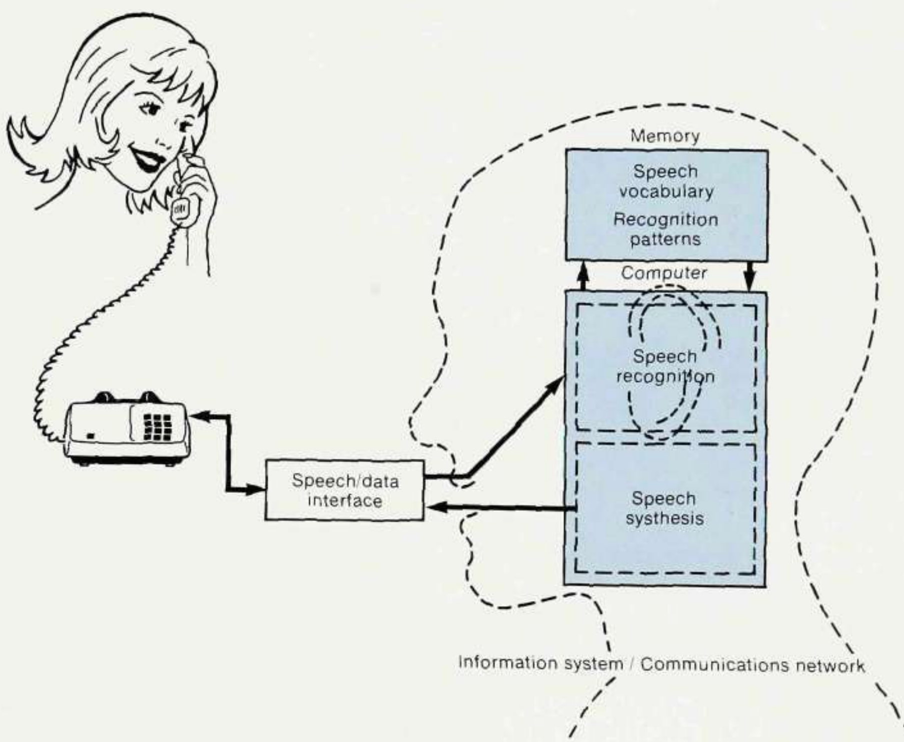
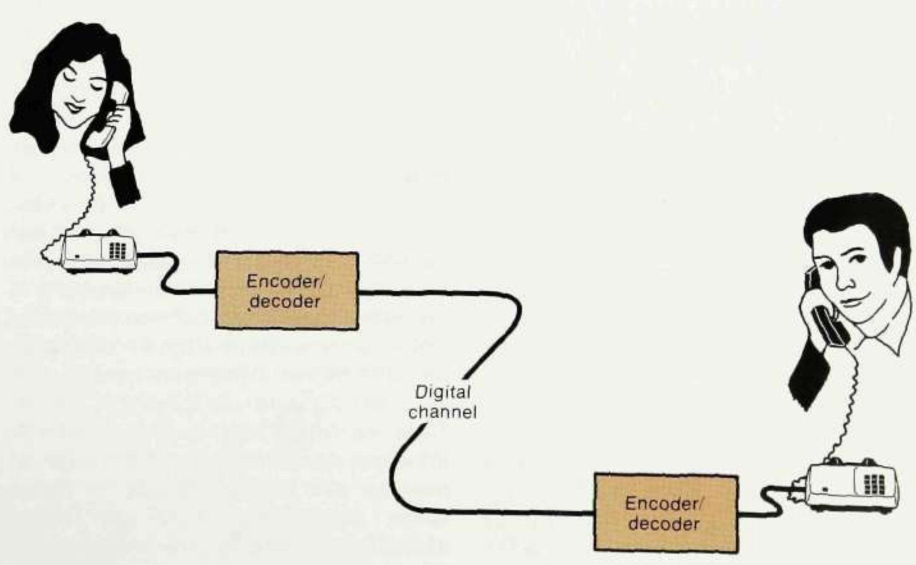


Fig. 2
The techniques of speech synthesis and speech recognition provide human-like abilities for machines



ig. 3
'he technology of speech *coding* provides fficient digital transmission over switched hannels

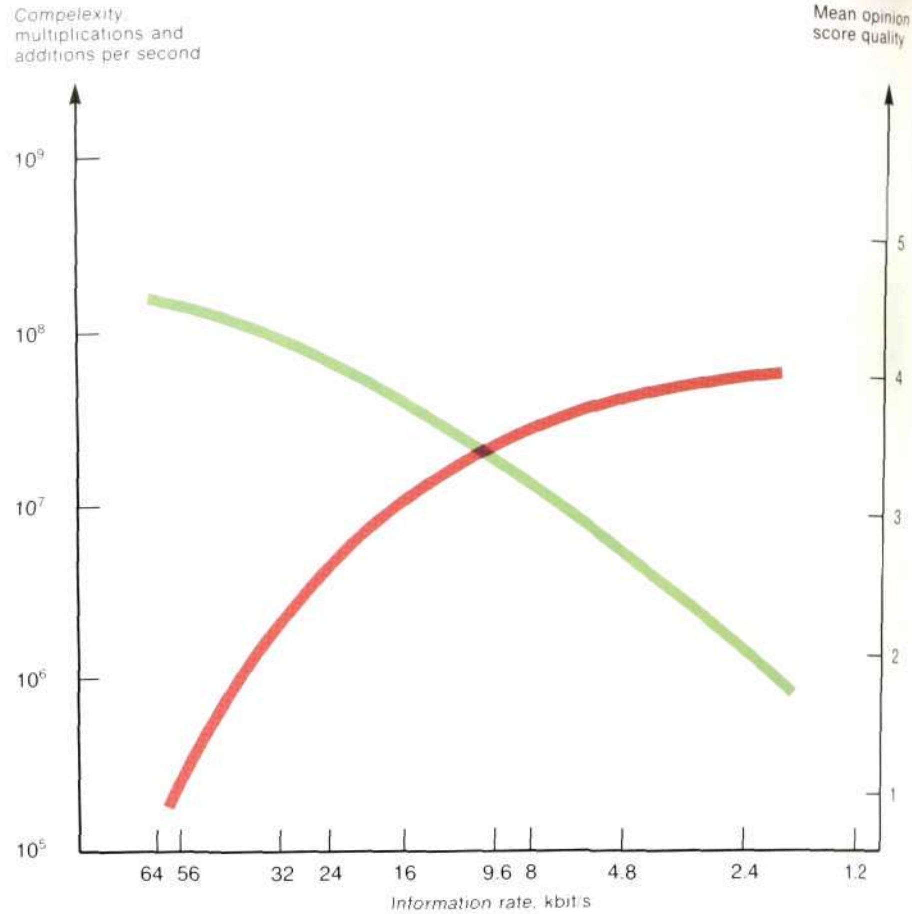


Fig. 4
Relations among transmission information rate, computational complexity and auditory quality of speech coding methods
Signal quality
Processor complexity

tinguished by the terms coding (people talking to people), synthesis (machines talking to people), and recognition (people talking to machines).

In each of these sectors, fundamental research in speech generation and perception has been key in establishing technical capabilities useful for humans. Let us briefly examine the state of progress and the future outlook for each.

Speech coding

The objective in speech coding is to analyze and represent (encode) speech as a digital signal which requires the smallest amount of transmission capacity necessary to recreate (decode) the speech at the receiver. Ideally, the resulting auditory quality should be comparable to the original. The procedure is illustrated in fig. 3. There are theoretical reasons to believe that a minimum transmission capacity of the order of 2000 bits/s should be adequate to transmit speech with natural quality, including personal characteristics of the talker. But we are far from this capability at present. Fig. 4 shows representative relations among information transmission rate, the transmitted signal quality, and the computational complexity of the encoder/decoder. High-quality transmission can be achieved in the range of 64 kbit/s (the traditional rate for Pulse Code Modulation – PCM) to about 32 kbit/s (the rate for the more recent Adaptive Differential PCM). Relatively modest complexity is required.

At the lower transmission rates quality diminishes significantly, and the computational complexity to implement the coding algorithm increases. The research frontier in this technology centers on methods for achieving high quality transmission at rates of 9.6 kbit/s and below. Very likely, substantially increased computational complexity will be required to elevate the quality of low bit-rate coders.

A broader perspective in speech coding is given in fig. 5, showing some technical milestones and the fundamental knowledge responsible for the advances. In the current sphere of research activity, fundamental studies on parametric models of speech generation and perception figure prominently.

Fig. 5
Milestones in coding

Technology	Knowledge base	Status
Analog	Replication of continuous waveform of speech	Deployed mature technology (1900)
TASI	Burst nature of speech	Deployed (1960)
PCM (64 kbit/s)	Theory of sampling and binary representation	Deployed (1962)
ADPCM (32 kbit/s)	Correlation of samples of speech waveform	In development (1984)
Adaptive coding (16–9.6 kbit/s)	Perpetual properties of coding noise	In explanatory development, research (1985)
Articulary coding (9.6–2.4 kbit/s)	Parametric models of speech generation and perception	In research

Speech synthesis

The ingredients of speech synthesis are illustrated in fig. 6a and b. Ideally, the machine should be able to convert any message text into spoken form, fig. 6b. This unrestricted "text synthesis" is an ambitious objective and is the focus of intensive research. Some initial successes have been achieved and first products, somewhat limited in capabilities, are emerging.

Broader applications exist for techniques more limited in versatility, fig. 6a.

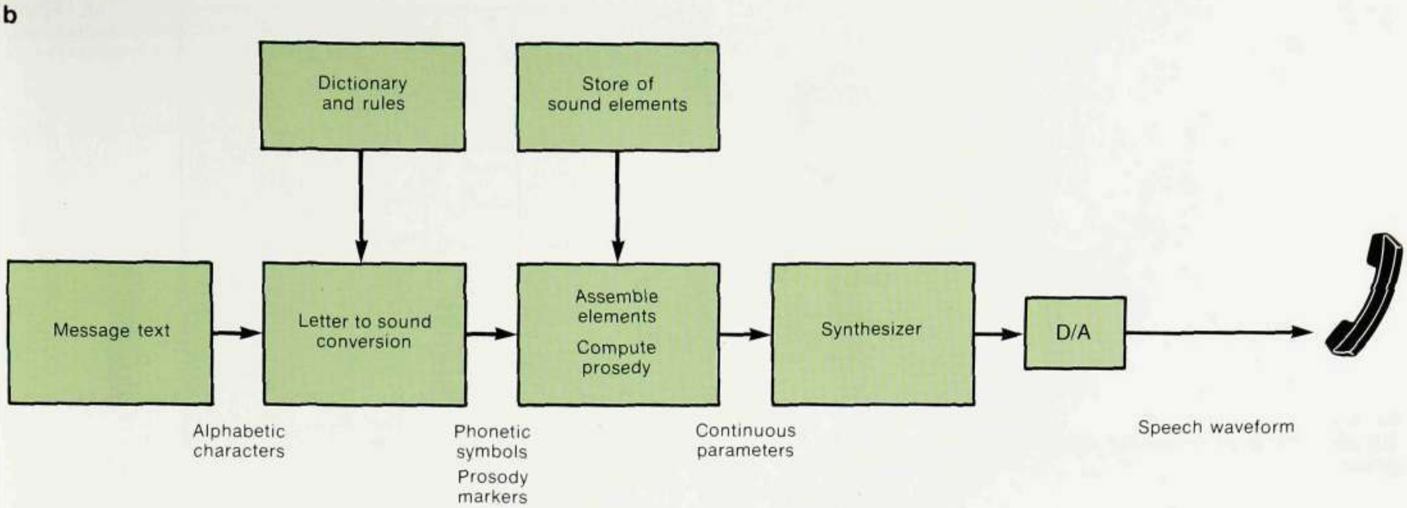
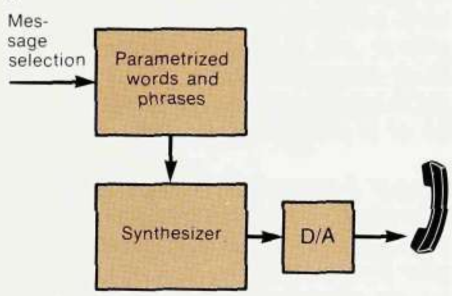


Fig. 6
Ingredients of speech synthesis.
(a) synthesis from low bit-rate parameters
(b) synthesis from text



Here human-spoken words and phrases are analyzed and stored efficiently in low bit-rate form in digital memories. Messages can then be assembled from this restricted vocabulary. Many announcement machines of this type are in use today.

In both techniques a synthesizer component based upon fundamental knowledge of the vocal tract and human speech mechanism is required.

Fig. 7 depicts three important dimensions in speech synthesis – namely, speech quality (or naturalness), message versatility (or vocabulary size), and complexity (or cost). The desired performance, of course, is high quality and versatility and low complexity. In actuality, simple systems that store recorded waveforms have high quality and low complexity, but also low versatility (i.e., the messages can essentially be used only in the form recorded). At the other extreme, text-to-speech systems based upon computational synthesis have great versatility, but also high complexity and, as yet, limited quality.

Some perspective in speech synthesis is drawn by the milestones listed in fig. 8. Again, in current research, computational models of speech generation and articulation have a central role in the required knowledge base.

Speech recognition

There are a variety of ways to approach the problem of automatic recognition of spoken commands. One of the more successful to date is based upon pattern

Technology	Knowledge base	Status
Analog recording	Magnetic and optical storage; message composition	Deployed, mature technology (1960)
Digital waveform storage (64–24 kbit/s)	Digital speech coding	Deployed (1978)
Parametric synthesis (16–4.8 kbit/s)	Acoustic model of speech generation	In development (1984)
Synthesis from text (0.05 kbit/s)	Letter-to-sound rules; model of speech articulation	In research

Fig. 8
Milestones in synthesis

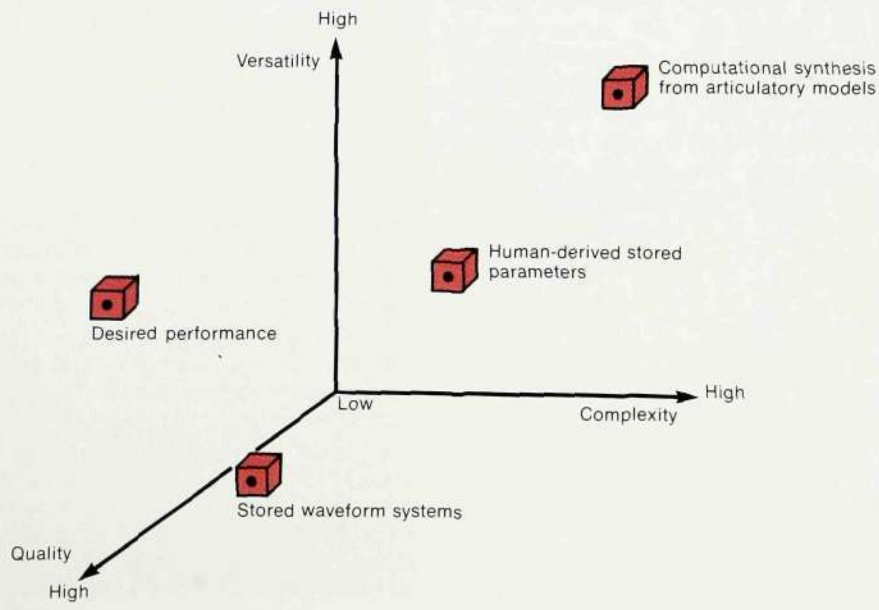


Fig. 7
Relations between auditory quality, message versatility and algorithm complexity in speech synthesis

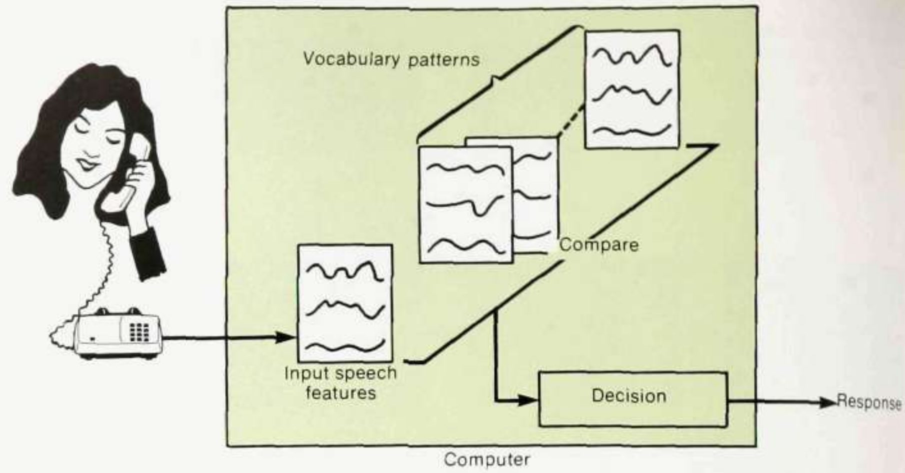


Fig. 9
Speech recognition by matching of stored templates

recognition, using stored templates for word units, as shown in fig. 9. This approach will encounter limitations for large vocabularies, and feature-based techniques may have advantages. But, as yet, the latter are not established.

In fig. 9, the unknown spoken command is analyzed in terms of spectral features (a popular analysis is in terms of linear predictive coefficients, or LPC). The unknown utterance is then compared to the stored patterns for the vocabulary that the machine can recognize. A dynamic programming technique is used for the comparison. The closest match is then identified with the unknown. If a sufficiently close match is not found, the machine can so announce, using its synthetic voice.

The dimensions of the recognition problem are at least three: vocabulary size, speaker identity, and fluency of input speech. Current progress is such that practical systems are being built for recognition of several hundred words. Recognition for any and all speakers (i.e., speaker-independent systems) requires about an order of magnitude more computation than recognition for designated speakers (i.e., persons who have entered their own vocabulary patterns into the machine, or speaker-dependent systems). Recognition of single words or short phrases, spoken in isolation, can be reliably done, even over dialed-up telephone channels. Recognition of connected words is under exploratory development. Recognition of conversational, fluent speech is in fundamental research, and will strongly depend upon viable computational models for syntax and semantics.

Vocabulary	Mode	Recognition accuracy (%)
10 digits	SI	98
37 dialer words	SD	99
39 alpha digits	SI	79
54 computer words	SI	96
129 airline terms	SI	91
200 city names	SD	97
1109 words, basic English	SD	79

Fig. 10
Typical performance for isolated word recognition

SI Speaker independent
SD Speaker dependent

Typical performance data in fig. 10 illustrate how word recognition can depend not only upon the size of the vocabulary, but also upon the nature of the vocabulary. The spoken digits – a fairly distinct, small vocabulary – can be recognized with high accuracy, even in speaker-independent operation. When combined with the spoken alphabet (39 alpha digits) the recognition performance typically deteriorates, because the spoken a, b, c, d, e, etc. have very similar acoustic features. In this case, some appeal to the constraints of context and programmed syntax is valuable.

The applications history in speech recognition is, for the most part, more recent than for coding and synthesis, and some milestones are illustrated in fig. 11. Techniques for speech analysis, models for auditory perception, and algorithms for syntactic and semantic analysis are current issues in research.

Future outlook

Some indications of the focus for future progress have already been suggested by the "bottom lines" of the milestone charts, figs. 5, 8, and 11. But what are the fundamental limitations to this progress?

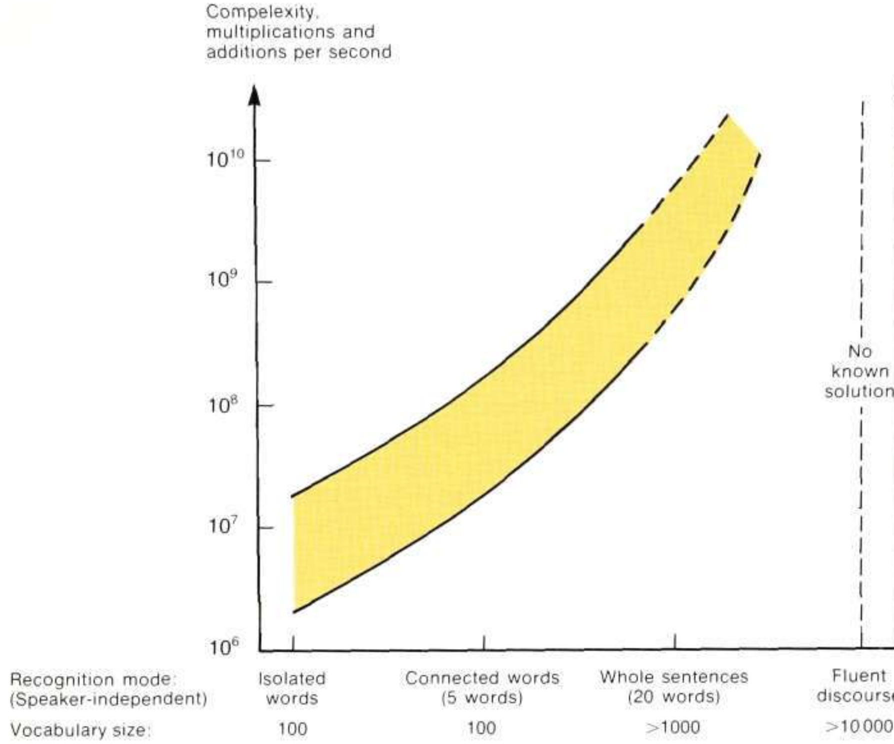
The limitations are not so much in our ability to conceive, in unfettered ways, new techniques and principles. Neither are they in our ability to implement, in sophisticated special-purpose hardware, the new and daring principles. Rather, they lie in the rate at which we can acquire knowledge, through experimenting with new algorithms and estab-

Technology	Knowledge base	Status
Speaker-dependent isolated words	Acoustic spectrum analysis Pattern similarity measurement	In advanced development (1984)
Speaker-independent isolated words	Algorithms for universal patterns	In development (1984)
Connected words, finite vocabulary	Word boundary location; Contextual effects on speach patterns	In exploratory development, Research (1985)
Fluent speech recognition	Models for auditory signal processing, syntax, semantics	In research

Fig. 11
Milestones in recognition

Fig. 12
Computational complexity required to implement known algorithms in speech recognition

Known algorithms



lishing the feasibility and value of new models.

A fundamental limitation to the rate of knowledge acquisition is the general-purpose computational power available in the laboratory. The issues in speech recognition provide a good vehicle to illustrate this point, though the same limitations exist to a lesser extent in coding and synthesis.

Fig. 12 shows the approximate range of computational complexity necessary to support presently-known speech recognition algorithms. Research must be done on general-purpose, programmable and re-configurable computers. Once a proven answer is gained, the result can be implemented in special-purpose hardware designed for high speed on that specific algorithm. Laboratory computational capability is presently typified by the lower end of the complexity axis, i.e. capability for about 10^6 multiplications and additions per second. This is inadequate to perform in real time even the simpler algorithms. The algorithms of current research interest (i.e. for whole, connected sentences) require about two orders of magnitude greater computational speed for real-time operation. Therefore, each small modification in an experimental model or principle is subject to substantial delay in laboratory evaluation.

One may argue that so much computation is required because we must be solving the problem the wrong way. And this may be true. Research is in fact slowly pressing downward the lower boundary of the curve in fig. 12, but to make such progress, computation sufficient for rapid assessment of known techniques and their logical extensions is necessary. Happily, the cost of general-purpose computation continues to diminish, and the outlook for having computational capability of the order of 10^8 multiplications and additions per second, or greater, routinely available in the laboratory is good. This capability will permit realistic evaluation of ideas as complex as the "shadowing" of an unknown real speech signal by an adaptive computer "mimic". If such could be successfully implemented, the related problems of coding, synthesis and recognition coalesce into one ultra-sophisticated model – and are, at once, solved.

TRTS – a System for Verifying the Availability and Grade of Service in Telecommunication Network

Morten Brinch, Per Haldorsen and Ragnar Huslende

TRTS has been developed by EB Telecom, a division of the EB Group in Norway. It is a system for veryfying the availability and grade of service in a telecommunication network. TRTS provides the administration with information about the network as experienced by the subscribers. It is also a fault locating aid and a useful tool for pinpointing bottlenecks in the network. The results of the test calls are processed to provide extensive statistical data, which in clear text inform the administration about the grade of service and availability in the different parts of the network.

The authors describe the theory of such testing and the function and structure of the equipment, concluding with a summary of results and the economical aspects.

UDC 621.395.74.001.42
telephone network
telecommunication traffic
testing
reliability
fault location

Artificially generated test traffic versus monitoring of real traffic has for many years been a topical issue for telecommunications administrations in many countries. Manual test calls have been used on their own or in combination with different types of automatic equipment in order to measure the availability of the telecommunication network.

Towards the end of the 1960s many administrations purchased the traffic route tester TRT m60 for the purpose of obtaining more systematic verification of the availability of the telecommunication networks. The discussion concerning the usefulness of such equipment continued. However, several administrations were firm believers in such equipment, and the Dutch, among others, have made large savings and great improvements through their extensive use of traffic route testers (TRT m70)[1].

The Norwegian Telecommunications Administration collaborated with EB

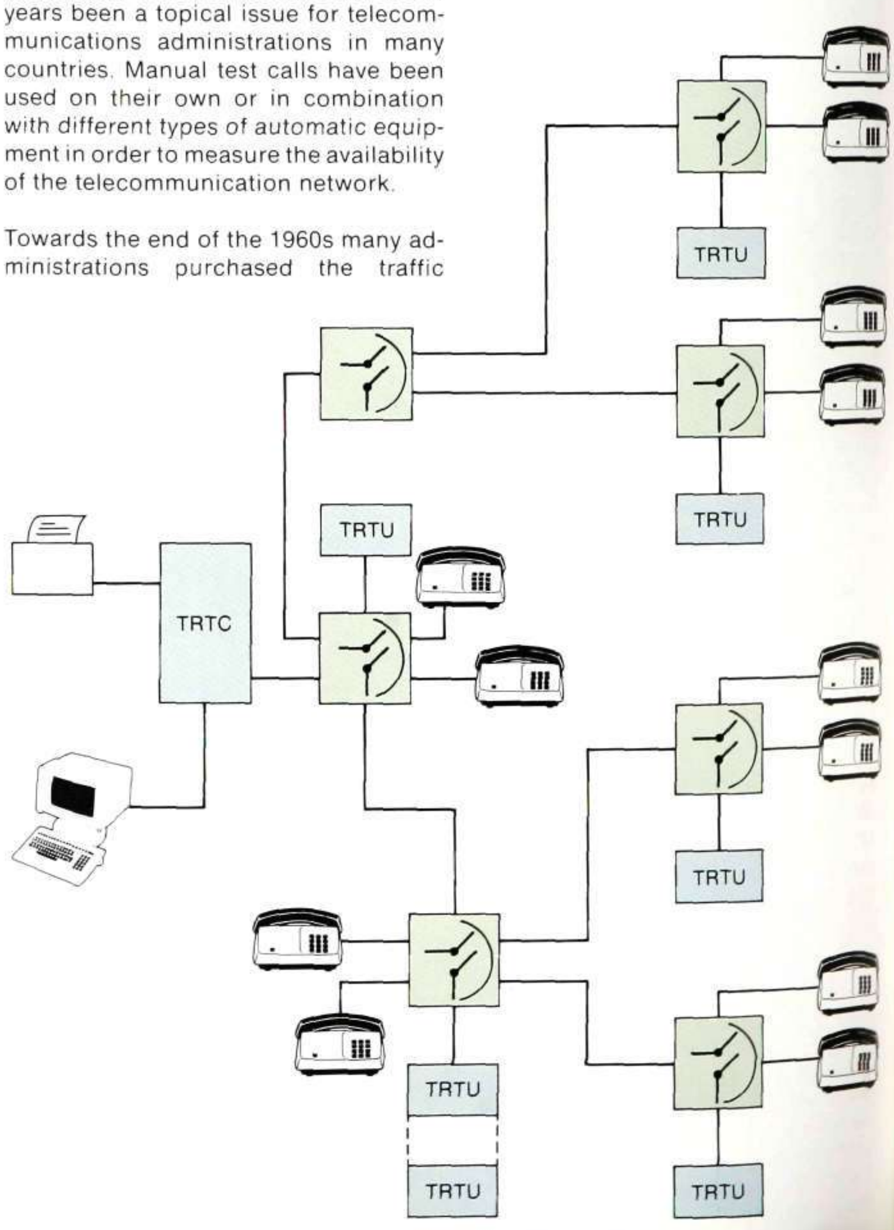


Fig. 1
TRTS consists of two subsystems, TRTC and TRTU. TRTC generates test packets that are transferred to TRTU for subsequent execution

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Calculation of confidence intervals

A series of M independent test calls is made. The sought probability of a call being lost in the network is p. The number of lost test calls is assumed to be a stochastic variable with a binomial distribution. In accordance with the central limit theorem the normal distribution

$N(Mp, Mp(1-p))$

with the mean value Mp and the standard deviation $Mp\sqrt{(1-p)}$ is reached when M is increased. This is taken as a practical approximation when $Mp > 10$.

$\hat{p} = x/M$

is then introduced. It is an extrapolated point estimator for p which coincides with the maximum likelihood estimator. Using the approximation above we obtain the following $(1-h)$ 100% confidence interval for p:

$$\hat{p} \pm z_{h/2}\sqrt{\frac{\hat{p}(1-\hat{p})}{M}} = \hat{p} \pm d$$

where z has the standardized normal distribution N(0.1).

The normalized one-sided interval is given by

$$r = \frac{d}{\hat{p}} = z_{h/2}\sqrt{\frac{1-\hat{p}}{\hat{p}M}}$$

Telecom during the development stage of the traffic route test system TRTS. This collaboration has actually been going on since Ericsson delivered TRT m60 and m70. During this period EB Telecom has kept abreast of developments and has been in regular contact with several telecommunications administrations. This has resulted in the new TRTS equipment, which has already been supplied to two customers.

Theoretical background

Two of the main objectives in the use of TRTS are to optimize

- the operation and maintenance of the existing network
- the planning of further extensions to the network.

The size and complexity of a national telecommunication network mean that a large amount of measurement data is required to obtain a picture of the current state of the network. All such data can be collected by means of TRTS. A description of how to calculate the amount of data and the test time required is given below.

A more detailed description of the basic theory and the application of TRTS was given at the *5th Nordic Teletraffic Seminar.*[2]

Statistical confidence

The basic principle of the traffic route tester is that a small, controlled volume of test traffic is generated and used to observe the quality of the network. The volume must be so large that the resulting data concerning lost calls and faults reflect the quality of the network with a sufficiently high degree of confidence. On the other hand the test traffic must not be so large that it interferes with the normal traffic handling in the network. The fact panel "Calculating confidence intervals" describes how the number of test calls required to obtain a given statistical confidence coefficient is calculated. In fig. 2 the results for some typical confidence intervals are shown graphically. After 500–1000 calls it is normally quite clear which parts of the network require a separate follow-up by the operation and maintenance staff. Fig. 2 also shows another interesting fact. For a given confidence limit the required number of test calls decreases rapidly with an increase in the probability of lost calls or faults. For example, the number of calls can be reduced from approximately 1200 to around 370 when the probability of lost calls is increased from 5% to 15%. This means that areas in the network that are likely to cause trouble can be pinpointed very quickly with the aid of TRTS.

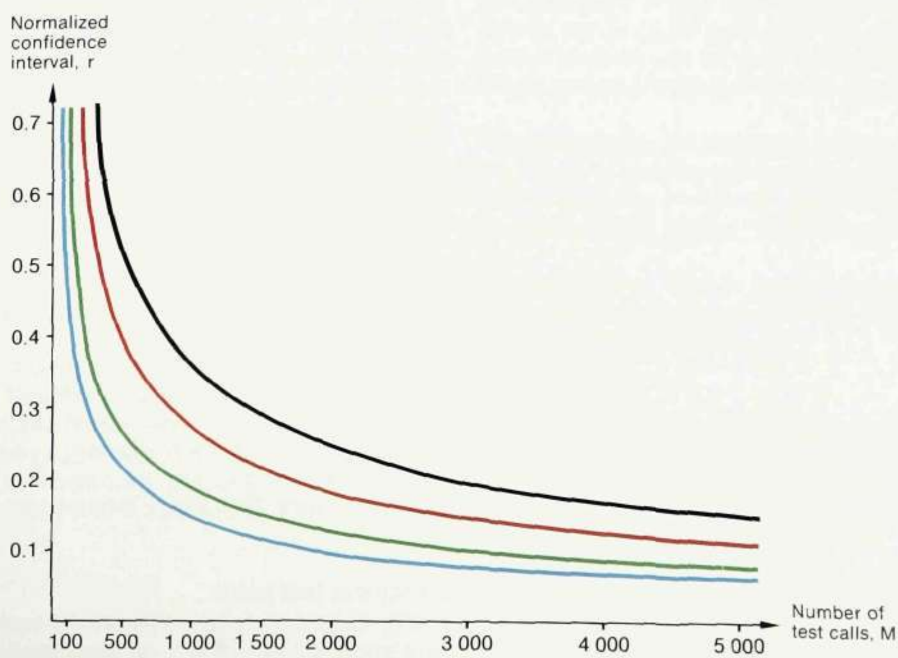


Fig. 2
Confidence intervals for some values of the estimated loss probability, $\hat{p}$
Confidence level: $1-h = 95\%$

- $\hat{p} = 3\%$
- $\hat{p} = 5\%$
- $\hat{p} = 10\%$
- $\hat{p} = 15\%$

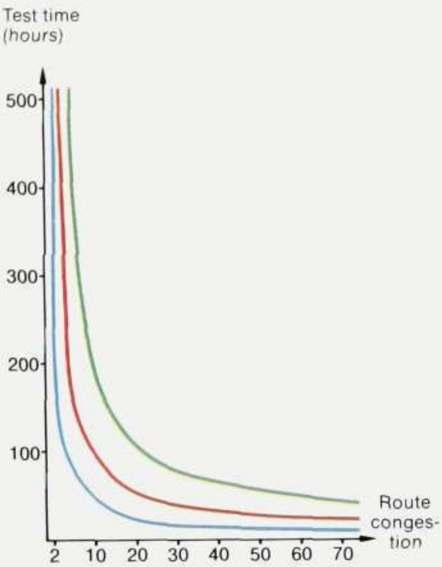


Fig. 3
Test time for a given statistical confidence value

— $L_{tot} = 1.1\,\%$
— $L_{tot} = 1.2\,\%$
— $L_{tot} = 1.5\,\%$

Dimensioned congestion:	$L = 1\,\%$
Estimated total loss:	$\bar{p} = 4\,\%$
Confidence level:	$(1\text{-}h)\ 100\,\% = 90\,\%$ $r = 0.25$

Test time

The test traffic can be spread out in time and adapted to the available capacity in the network. The dimensioning of the test traffic for the busy hour is a very important factor. Assume that the normal subscriber traffic experiences a congestion probability of L. The test traffic will mean a small increment, ΔL. The total congestion will then be

$$L_{tot} = L + \Delta L$$

Assume that the traffic, including the randomized test traffic, follows a Poisson distribution. The necessary test time will then be in accordance with fig. 3. In this case route congestion is used as a dimensioning criterion. With the given values a route of 30 lines will be sufficiently tested in 40 hours if an increase in congestion from 1.0% to 1.2% is permitted during this period.

In practice the lines in a route will be shared by many traffic directions between individual pairs of test units (TRTU). This means that only a small part of the available capacity in the route is available for testing each individual traffic direction. This potential problem is avoided by grouping the test units. TRTS can then carry out separate test sequences between geographical groupings having only one or a few lines between them. Such tests are then run at different levels in the network in the way described below.

It should be emphasized that the calculations above only apply to the busy hour. TRTS will of course be used outside the busy hour, both day and night, particularly in order to obtain different types of fault statistics. The test traffic can then be increased considerably, and statistical confidence is reached very quickly.

System concept

The basic system concept, which led to the development of a system comprising a central unit, TRTC, and a test unit (TRTU) installed in each exchange, is described below. See also fig. 1 and the fact panel with TRTC and TRTU functions.

Generated test traffic

Traffic theory studies resulted in the decision to base the system on generated test traffic, since this method offers good control of when and where in the network calls are generated, as well as of the number of such calls. The method also makes it possible to guarantee that terminating equipment is free, so that any congestion detected in the network cannot have been caused by the terminating equipment being engaged.

Moreover it is possible to measure various characteristics of the test call, such as: waiting time for dial tone, waiting time for ringing tone, attenuation or noise, and also to measure intervals between metering pulses if relevant. In addition, various types of faults that can occur during the setting up of a call can be detected and reported.

Independent system

The system is designed for use in any telephone network regardless of the types of exchanges and technology used in them. The system can therefore be used in existing as well as future networks, and will not become obsolete because of the technological development. The system uses an ordinary telephone subscriber interface and is therefore very easy to install.

Operation

An operator terminal connected to a central unit (TRTC) is provided for the operation of the system.

The basic aim of the system operation is that it should be as simple and self-explanatory as possible.

All communication between the operator and the system is based on menus displaying a specified set of functions from which the operator makes a choice. The first choice is a main function; then sub-functions are successively chosen until the procedure ends in the display of a form in which the operator has to enter data.

In the daily operation of the system the operator will only use a few menus (2–3), which the system then uses to generate detailed test call patterns in accordance with set rules. This is an improvement on previous system generations, in which the operator to a large extent had to feed in detailed control information.

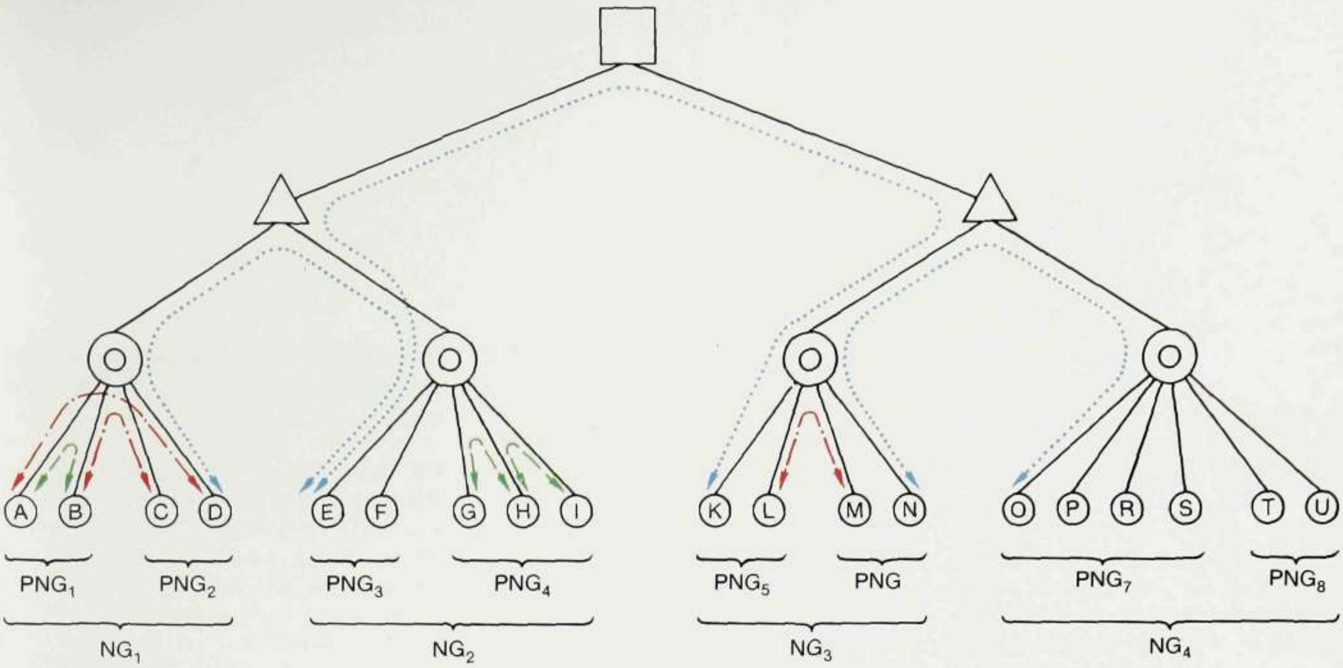


Fig. 4
***The exchanges in a telecomunication area (Ta)* are divided into a number of network groups (NG) and sub-network groups (PNG)**

Area centre

District centre

Group centre

Terminal exchange

Test traffic at the test number group level (PNG): e.g. between G, H & I in PNG_4, i.e. test traffic only between exchanges in the same PNG

Test traffic at network group level (NG): e.g. between A & D, B & C, L & M, i.e. test traffic only *between terminal exchanges belonging to different PNGs in the same NG*

Test traffic at the area level (Ta): e.g. between E & K, N & O, D & E, i.e. test traffic between terminal exchanges belonging to different NGs

Functions in TRTC

- To generate test call specifications for the TRTUs concerned
- To transmit the test call specifications to TRTU
- To collect the test results from TRTU when the test sequence has been completed
- To process the collected data so as to give the required result formats
- To present the results.

Functions in TRTU

- To receive test call specifications from TRTC
- To generate test calls in accordance with the received specifications
- To record all specified data during the course of test calls, i.e. only for calls during which faults are observed or for all test calls
- To return the collected data to TRTC for further processing.

Generation of test calls
The system generates autonomously a specification for all test calls that are to be carried out, on the basis of received information concerning the desired type of test and time of day, and also stored data regarding the network. The network data is fed into the system when it is first started up, and then only when required in connection with changes in the network.

The network is divided into groups at three levels (PNG, NG, Tlo), fig. 4, in order to ensure that the distribution of test calls throughout the network is similar to the real traffic. This method reduces the number of test traffic routes in the area where the system is installed to an amount that is reasonable and manageable, and which also reflects the real traffic requirements and the physical structure of the network. Test calls are carried out simultaneously in all groups at one level at a time, i.e. within an NG for example.

When generating the specification of the test calls the system always checks that the called equipment is available. This is possible because of the time slot principle used, whereby each individual test call is allocated a specific time slot, which is synchronized in all calling equipment (TRTU).

Test reports
The basic philosophy for the presentation of results is that the system should provide as few and as informative printouts as possible in view of the large amount of data it handles.

Functions

Due to the comprehensive tasks to be performed and the requirements for continuous system operation, independent of operator interventions, the system is equipped with primary functions and auxiliary functions (see the section "Practical use").

The primary functions comprise test functions, result reporting and test traffic handling, whereas such functions as defining the telephone network, compiling test schedules and requesting reports are auxiliary functions that are necessary in order to get the primary system functions performed in a rational way. The test functions are described briefly below.

Grade-of-service test
This is the main function of the system. This test is used to check the availability of the network during the busy hour. It is run simultaneously in all groups at the same network level. The level to be tested is specified in the test schedule. The results are accumulated over the test period (one month) for each group and can be presented in the form of a matrix.

The basic data for calculating the number of test calls to be generated is specified in advance in matrices that follow the division into groups in the network.

The results of these measurements show congestion caused by technical factors and the traffic load.

Network test
This test is used outside the busy hour and is basically like the grade-of-service test, apart from the fact that the volume of test traffic is scaled up since the network can take more test traffic.

When this test is used during appropriate parts of the 24 hours of the day it will primarily show the technical quality of the network, since the large traffic volumes that cause congestion will seldom occur. Consequently the influence of traffic on the results will also be small.

Extended transmission test
This function is designed to carry out level and noise measurements when the test calls have been connected up. The level is measured at three predetermined frequencies in both directions of transmission. The function is a further development of the level measurement or test tones, which is carried out for each test call.

The result printout gives a summary of lines which deviate from the given normal requirements, with the possibility of detailed presentation of the calls that deviate from the standard.

Charging test
This function can check whether the call metering equipment in the exchanges operates correctly as regards tariff and time charged.

Special sets of data that define groups of TRTU (exchanges) with the same rate, and the different rates that apply within and between the groups, are fed into the system as the basic information for the generation of test calls that check the charging.

The results show any cases of overcharging and undercharging, and a detailed presentation of test calls with errors shows measured and nominal values for the interval between metering pulses (in those cases where random metering is used for charging).

Individual call
This function enables an individual call to be set up between two TRTUs, using a TRTC. The function can be used when taking a TRTU into service, and when faults detected in other tests have to be followed up. The system operator can then follow the setting up of the call via the headset, and the data from the setting up will be shown on the operator's terminal.

Special follow-up
This function permits the definition of special groups and is intended for those cases where it might be desirable to deviate from the groups that have been defined in the telephone network specification.

Fig. 5
TRTC uses BYB cabinets holding five magazines each. The I/O equipment consists of a visual display terminal and a printer

Hardware

Fig. 5 shows a central unit with an operator's work station and fig. 6 a TRTU. The construction practice used is Ericsson's BYB.

TRTC
The basic hardware units consist of a computer module with secondary store and other peripheral equipment (PPO) and a telephone unit (TU).

The computer module, PPO, is built up using Ericsson's general computer system APN 167. It comprises a CPU, primary store and connected peripheral units, such as floppy and hard disc units and a number of asynchronous terminal or modem lines. It also includes a connection to the telephone unit. The system has a 16-bit data bus and a total address capacity of 16 Mbytes.

The telephony unit, TU, comprises eight separate units, consisting of modem and terminating equipment for a normal telephone line.

Technical Data

TRTC	
Processor	MC 68000 (APN 167)
Memory	1.5 Mbyte RAM Hard/floppy disc
Serial communication lines	11
Number of simultaneous connections to TRTU	8
TRTU	
Processor	MC 6800 (APN 165)
RAM	9 kbytes
EPROM	36 kbytes
Number of test lines	40 max.
I/0 equipment	
VDU, for example	Tandberg TDV 2230
Printer, for example	Facit 4542
Power consumption, 48 V d.c.	
TRTU	75 W
TRTC	500 W
Dimensions	
TRTC: Two five-magazine cabinets	1860×600×300 mm
TRTU: One single-magazine cabinet	470×600×300 mm

TRTU
TRTU is built up around a processor (APN 165) developed by Ericsson, and a number of I/O circuits. See "Technical Data".

Software

TRTC
The basic principles for the software and program interaction are the same as in system AXE 10, which means well defined program modules with interfaces in the form of program signals. Experience has shown that this has many advantages, such as more convenient handling, better facilities for function changes, greater reliability because of higher program quality and greater utilization of the programs.

With the exception of a few central parts of the operating system all programs are written in the high-level language ERIPASCAL. This language is based on PASCAL with a few additions for real-time use. The module and signal concepts are important additions.

TRTU
The software is built up around a real-time monitor core, which comprises exchanges of messages between processes, handling of interrupts and queues of messages and processes.

Practical use

Definition of a test area
TRTS has a defined, fixed position in the telephone network, and the test area, for example a telecommunication area, is allocated a central unit, TRTC. The area is then divided into a number of network groups (NG) and sub-network groups (PNG), and all terminal exchanges are allocated to a group according to geographical position, traffic requirements and exchange size, fig. 4.

Input of network data
The structure of the test area determines the data to be input for the central unit and network.

The central unit data includes numbering scheme, area code, test tone level and time of communication between TRTC and TRTU. The network data includes lists of the various groups, device distribution, last choice routes and the maximum amount of test traffic.

Specification of traffic periods
Traffic periods are customer-defined data based on real traffic volumes. They are specified as busy hour, high-traffic and low-traffic periods.

Traffic-limiting parameters
Traffic-limiting parameters are specific factors that limit the test traffic during a traffic period in order to prevent uncontrolled traffic loads.

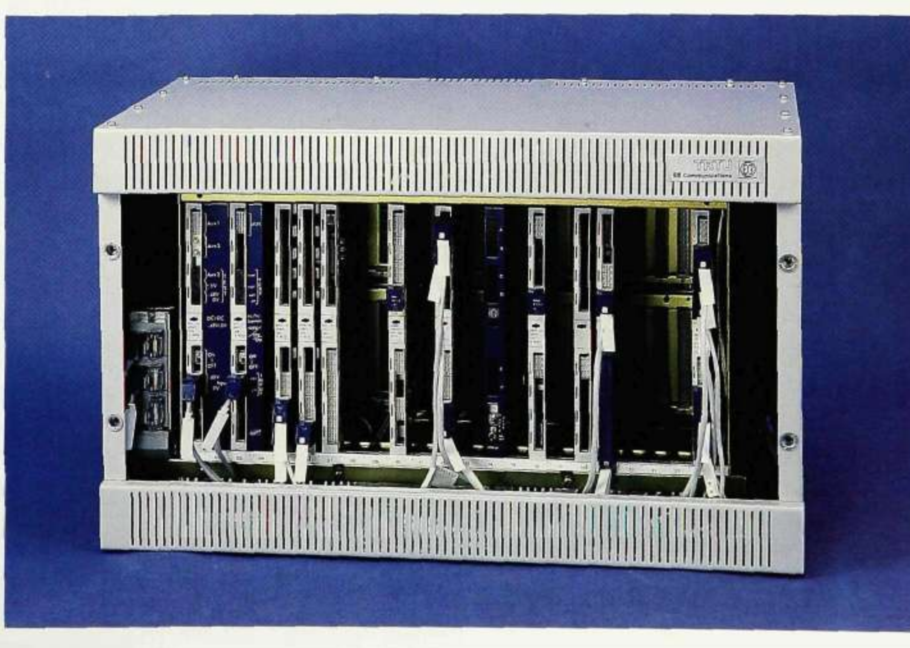


Fig. 6
TRTU is normally installed in a single-magazine cabinet, which can easily be mounted on a wall

Statistical norms

The measured results are compared with set statistical norms, in the form of limiting values, categorizing the results as faults, reduced quality of service, lost calls or satisfactory quality of service.

Choice of test type

TRTS gives the user great freedom of choice of test types, from the most restrictive lost traffic test, which is tied to the busy hour with preset limit values, to unscheduled individual calls and special follow-up.

Test schedule

This display form is the central control and information screen, fig. 7. The test schedule shows the test history for up to four weeks, with a summary of test runs and whether the results have been processed, accumulated or presented. The test schedule controls all tests to be made for the next four weeks. The test sequences are initiated by the operator entering start and stop times for the desired test functions.

Handling of test traffic

TRTC carries out test traffic as specified in the test schedule. The unit scans the schedule and finds the test sequence that is to be executed. Depending on the test function a test call pattern is generated with the aid of an algorithm designed for this function. The call pattern is then sent out to the TRTUs that are included in the test sequence. The TRTUs carry out the specified calls, and at the end of the sequence TRTC calls all TRTUs in order to collect the results. The collected data are then processed.

Presentation of results

Reports

The system provides different types of reports depending on user and purpose.

Grade-of-service printouts in matrix form, fig. 8, for presentation of the service test results as well as summeries of the overall grade of service, are provided. Fault statistics, based on all test calls performed, can be presented with different degrees of detail and in various combinations. The charging test and extended transmission test have their own result printouts.

Printouts are either programmed or obtained on command. The programmed printouts are produced at the end of an accumulation period (month). Moreover, all printouts can be ordered by the operator for immediate presentation, either on the VDU or on a printer.

Sequence summary

The sequence summary is a set of results that give a summary of the state of the network at the moment.

Detailed results

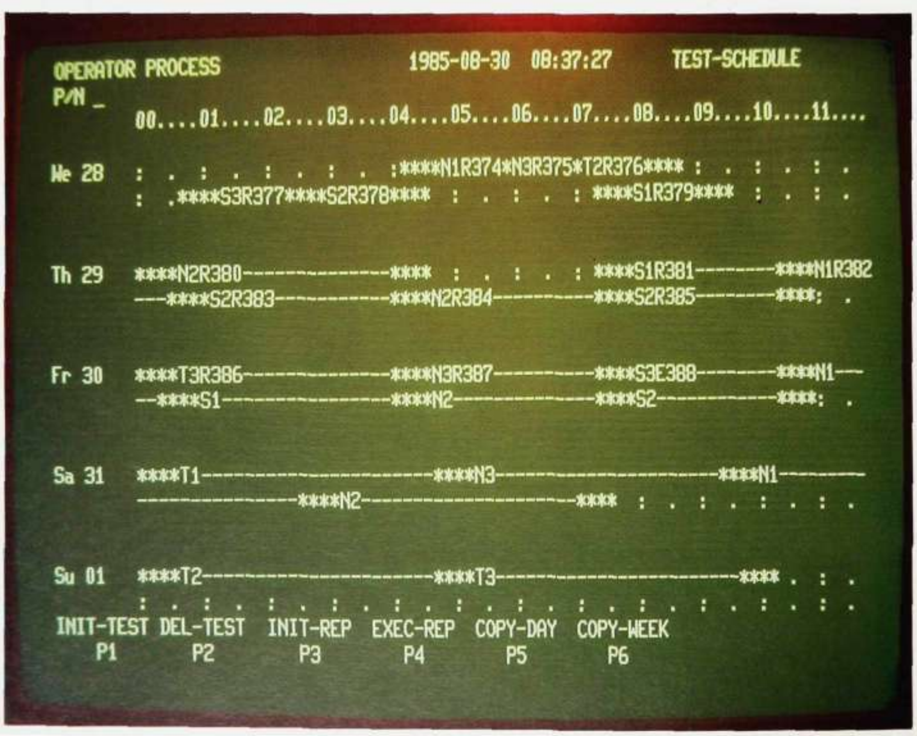
Detailed results can be ordered for each measurement sequence and provide the necessary operating information so that fault clearing work can be administered efficiently.

Service observations

Service observations give the telecommunications administrations a survey of the quality of the subscriber services they offer, with respect to:
- waiting time
- speech quality
- grade of service.

On the basis of this information the administrations can plan corrective actions and, if necessary, reconsider work and investment priorities.

Fig. 7
The figure illustrates how the 24 hours of the day can be used for different types of tests during a week



Fault statistics

The fault statistics should provide information about the technical quality of the network. The number of fault registrations and trends can indicate whether any special measures are required and where.

Types, distribution and frequency of printouts

The result printouts must have a receiver address in order to be meaningful. An organization and procedure for the distribution of result printouts must therefore be available. For example, in Norway a national operating centre has been set up which is also responsible for TRTS. Each telecommunication area also contains operational centres where operation and maintenance staff are stationed. Central traffic planning staff and administrative staff are also included in the result distribution lists. The different result reports are issued at different intervals. Detailed fault printouts can be output several times a day, whereas the grade-of-service statistics are issued once a month. The latter printout includes the results of the previous year and the current year. The results for the current year include accumulated totals as well as details for each month.

Fig. 8

A printout from a grade-of-service test with a summary of service matrices having four different selection parameters. The four quadrants show:

1 Lost calls as a percentage of the total number of calls when it exceeds the limit value (set to 3% here)
2 All lost calls as a percentage of all calls
3 As in the first item, but the number of lost calls is also given
4 All lost calls, both the actual number and as a percentage of the total number of calls

From NG	To network group										Total
	GK	GNR	DOK	FAG	LM	FBU	OT	LOM	DM	VSR	
GK				5.2		2.7	3.3	4.2	0.0	3.3	162 3.1
GNR					5.7	3.3	3.3	8.5	4.2	0.0	95 4.2
DOK				5.2		0.0	2.2	4.4	5.0	2.5	148 4.1
FAG	6.3				6.6	8.5	0.0	0.0	6.6	0.0	119 3.4
LM			3.3			0.0	3.1	0.0	0.7	7.5	175 1.2
FBU				13 5.3		0 0.0	8 5.0	9 1.1	9 2.2	13 7.6	92 3.3
OT	8 5.0	8 5.0			25 8.0	8 5.0	0 0.0	44 4.5	20 5.0	23 4.3	152 1.3
LOM						7 8.5	55 0.0	0 0.0	7 4.2	5 0.0	104 3.8
DM	6 3.3				21 4.7	10 0.0	15 3.3	6 6.6	0 0.0	19 0.0	98 3.1
VSR			8 5.0	9 5.5		9 2.2	28 4.2	6 0.0	22 4.5	0 0.0	142 1.4
Total	149 2.0	74 2.7	123 1.6	165 1.8	150 1.3	75 5.3	187 3.2	106 3.7	117 3.4	141 3.5	1287 2.7

Interpretation of results

As has already been mentioned it is essential that the result printouts are distributed to the correct addresses for the test results to initiate the appropriate actions. For example, detailed fault printouts are sent to the operation and maintenance departments, whereas service observations and fault statistics are distributed to administrators and planners.

In other words, the choice of result printouts and their interpretation give rise to different types of reactions over varying periods of time. Technical faults can result in immediate clearing action or a combination of several activities. This is where the sequence summary becomes very useful. In an operation and maintenance philosophy that is based on controlled corrective maintenance the work can be planned and the tasks of the operating staff be given different priorities and concentrated to points with the largest number of faults, highest error rates or least efficient traffic handling. In the cases where the sequence summary shows faults it may be appropriate to get detailed fault printouts, possibly also carry out new tests, before a repairman is sent out. Such situations are typical in the day-to-day operation.

Grade-of-service and fault statistics give another angle of the situation. The interpretation of such results has a more long-term aspect and is more oriented towards the general traffic handling ability of the network.

Savings when introducing TRTS

The introduction of TRTS can lead to savings in a number of areas for the administration:

- It will primarily result in service improvements in the form of greater availability in the network. This in its turn will mean higher revenue for the administration and shorter waiting times for the subscribers.
- It should also be possible to reduce the preventive maintenance and routine tests.
- A reduction of manual traffic testing in the form of test calls.

Service improvements
The results from TRTS mean that more technical faults can be repaired more quickly than with manual test methods.

Reduction of preventive maintenance
The following factors have to be taken into account when calculating the savings made in preventive maintenance:
- The number of tests made on each device is increased considerably when using TRTS instead of routine tests.
- The tests are carried out more frequently and at set times.
- The whole connection path is tested because instead of testing one device at a time the various devices are connected together in a number of different combinations and then tested.

The cost of introducing TRTS

The cost of introducing TRTS can be divided into investment costs and annual costs. The former include:
- direct costs for the purchase and installation of equipment
- indirect costs, such as the cost of test numbers and the extra load on the network caused by TRTS.

The annual costs include:
- depreciation, interest, maintenance, premises etc. for the equipment required
- personnel costs for the personnel required to operate TRTS.

The annual profit to be gained by using TRTS is made clear by comparing the total savings and annual costs resulting from the introduction of the system.

For a typical TRTS installation with a depreciation time of ten years the payback time is approximately two years and the internal rate of return approx. 60% when tax is not considered.

Conclusion

Good availability and high operational quality in the telecommunication network are two important requirements for modern networks, particularly in view of the increasing provision of new services (data, telefax etc.). In such multi-purpose networks there will also be a mix of technologies (analog, digital), which makes the network even more complex. The network itself is often forgotten in the debate on future telecommunications. The network has to be there, but ideally it should not be noticed. All attention is focussed on the sea of equipment that is now available at each end of a communication line. But when people can no longer reach the number they want, or if they have to wait for a long time, the consequences can be enormous in a modern society where good telecommunications are taken for granted. The need for modern equipment for supervising the quality of the network is therefore urgent.

Modern telecommunication networks contain a large amount of equipment for automatic routing of the traffic over alternative paths in order to make the optimum use of the network. This can have the disadvantage that disturbances in one place can easily spread to other parts of the network and make the problem worse. With TRTS, and a rapid and efficient analysis of the results, these propagative effects can very quickly and easily be detected and limited.

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Transmultiplexer for the 24-Channel Hierarchy

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The gradual digitalization of the telecommunications networks means that FDM and TDM systems will be used side by side for a long time. The transmultiplexer provides the most efficient conversion between the two multiplex structures. Ericsson has developed a third generation of transmultiplexers for the 24-channel hierarchy which meets the latest CCITT recommendations. The new equipment has a wide range of functions while its volume has been reduced to one third of the previous generation.
The author describes system applications and the properties and structure of the new generation of transmultiplexers.

UDC 621.395.4
multiplexing equipment
transmultiplexing
frequency divison multiplexing
time division multiplexing

Digital switching equipment has been shown to have considerable technical and economic advantages and is now being introduced on a large scale in the telecommunication networks.

In the long-distance network analog transmission (FDM) has hitherto been superior as regards capacity and economy. New amplitude-modulated radio systems that permit single sideband (SSB) transmission have been introduced. This has meant a doubling of the capacity compared with the traditional frequency-modulated systems.

However, digital transmission is now being introduced into the long-distance network on a large scale, among other things because optical fibre has proved to be the most economical transmission medium.

The above-mentioned factors mean that there is a need for interconnection of analog and digital links of the network. This interconnection can be carried out by means of transmultiplexers with the task of converting an FDM signal to a TDM signal and vice versa. The transmultiplexer is efficient; the telephone signals are connected through without any loss of capacity. The well defined interfaces mean that the equipment can be utilized in a flexible manner in different applications.

System applications

The transmultiplexers have several applications, which can be divided into two main categories:
- termination of the analog long-distance network to digital exchanges
- interconnection of the analog and the digital network.

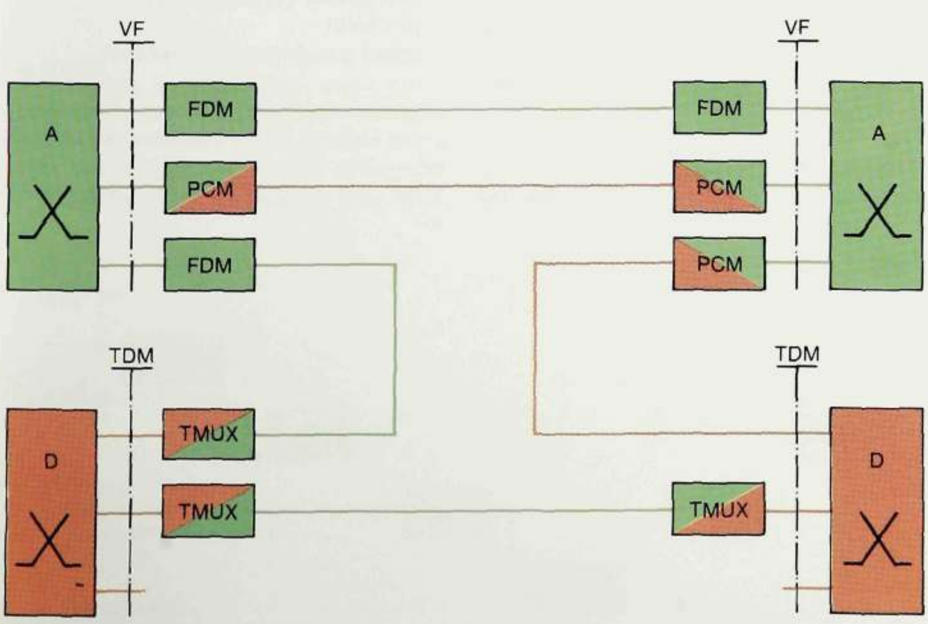
The first category has been predominant up to now because digital switching technology offers such great advantages that it has already been introduced on a large scale.

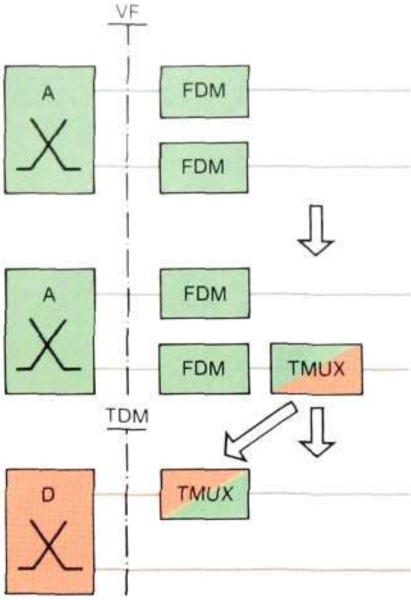
The second application is expected to become more common as digital transmission is introduced in the long-distance network.

Fig. 1 shows the different cases that can occur in traffic between analog and digital exchanges. The interface towards the analog exchange can be designed in many different ways. However, they all have one disadvantage in common: for each channel there are two or more wires that transmit speech and signalling information. Consequently much work would be required in order to introduce modifications or new installations at this interface.

The interface towards the digital exchange, on the other hand, consists of a simple, standardized 1544 kbit/s bit stream which contains 24 channels with signalling information included.

Fig. 1
Mixed analog and digital network
VF Speech and signalling interface
Analog line system (FDM)
Digital line system (TDM)





a

b

c

Fig. 2
Possible strategy for conversion to a digital network
- a Analog connection to an analog exchange
- b Analog and digital connection to an analog exchange
- c Analog and digital connection to a digital exchange

Exchange connection
New exchange systems, switching digital transmission paths, are becoming increasingly common in the networks. The transmultiplexer with its standardized interfaces provides rapid, simple and space-saving connection of the analog long-distance network to the digital interface of the exchange.

There are good reasons for considering the transmultiplexer alternative also when connecting digital routes to an analog exchange that will eventually be replaced by a digital one. The more immediate alternative of connection via a PCM-multiplexer does not only mean that the equipment will become redundant when the exchange is converted to digital operation, but will also require extensive installation work at the channel-associated speech and signalling interface.

The transmultiplexer application shown in fig. 2 ensures easy installation at a high network level, without the channel-associated interfaces being affected. Furthermore, the latter will have to be dismantled when the digital exchange is installed.

When the exchange is converted to digital operation the transmultiplexer is used to terminate FDM routes.

Interconnection
The analog and digital multiplex equipment both have a hierarchic structure, so that interconnection within each type can easily be arranged. Efficient networks can thus be built up at a reasonable cost.

The transmultiplexer offers a means of simple interconnection between the two hierarchies, because of the standardized interfaces. Moreover, as long as the traffic consists of telephone signals the interconnection does not entail any loss of capacity.

Fig. 3 shows the interconnection of some channels from an optical fibre line system to a radio relay link.

System characteristics

CCITT has standardized transmultiplexers for the 30-channel and 24-channel hierarchies. The recommendations comprise both interfaces and performance data. Ericsson's transmultiplexers are in accordance with these recommendations, and they are thus both safe and efficient. The application, characteristics and construction of mainly ZAJ-60 have been described in a previous article[1]. The present article deals with the third generation of Ericsson's transmultiplexers for the 24-channel hierarchy, ZAJ 120-3. This system converts five 24-channel 1544 kbit/s PCM bit streams to two standardized supergroups in the frequency band 312–552 kHz and vice versa.

The transmultiplexer operates in the conventional way, i.e. analog/digital conversion takes place at speech frequency. The conventional method has many valuable advantages:
- high realiability
- low power consumption
- flexibility
- small group delay increment.

In order to meet the requirements for small volume the speech and signalling interfaces have been optimized. The TDM and frequency generation parts

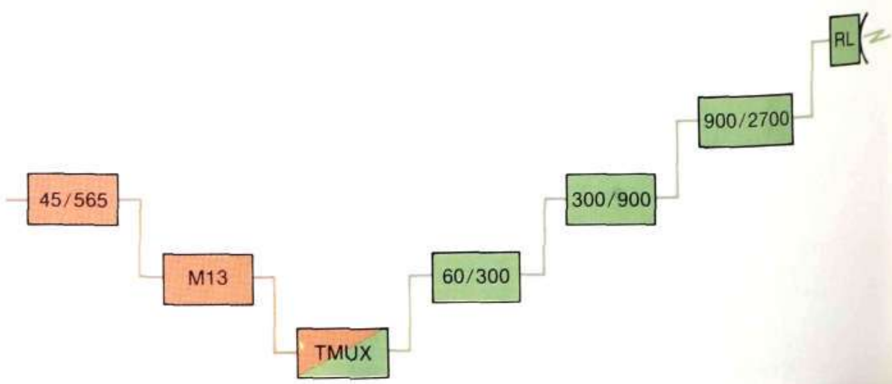


Fig. 3
Interconnection of a digital fibre system and an analog radio relay link

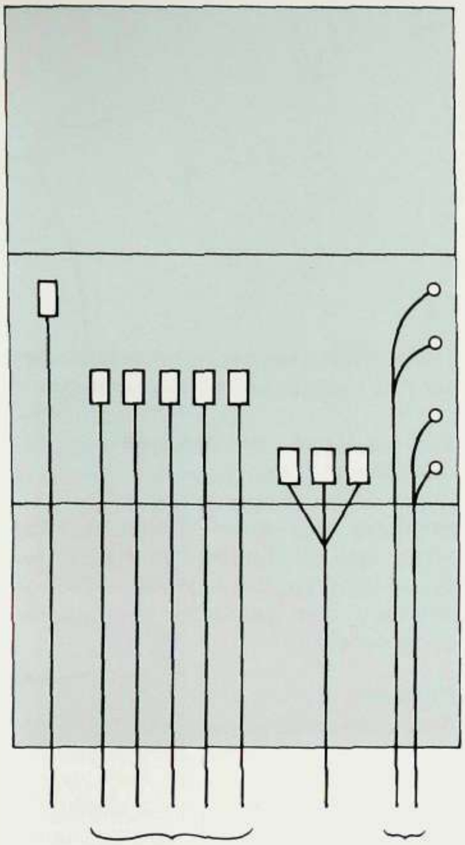


Fig. 4
External connections to ZAJ 120-3
1 Battery voltage −20 to −72 V
2 Supergroup inputs and outputs
3 Five 1544 kbit/s inputs and outputs
4 Alarm interface

have been designed exclusively for this purpose.

Installation
The transmultiplexer is designed as a mechanical and functional unit. This means that the extensive installation and wiring work required previously, when the system consisted of several magazines, has now been eliminated.

Installation of the equipment requires only the connection of a few cables to easily accessible connectors at the front of the equipment. The great advantage of this connection method is that all external cabling can be done before the equipment is installed. Furthermore, the bays can be placed back to back, giving a considerable reduction of the amount of floor space required. Fig. 4 shows that only the power supply, inputs and outputs and any alarm supervision have to be connected.

Adjustment to correct levels at the TDM and FDM sides, to compensate for different lengths of station cabling, is carried out by means of plug-in U-links.

Synchronization
The synchronization principles used in earlier transmultiplexers have been retained in ZAJ 120-3. This means, for example, that the risk of slip has been eliminated since the five TDM bit streams are synchronized independently of each other.

However, the function has been improved and augmented, compared with previous transmultiplexers. The whole equipment can now be synchronized from the incoming TDM bit streams. If the frequency accuracy of the bit stream is good, there is not even any need for an external basic frequency for the FDM part. The accuracy is usually satisfactory if the bit stream is obtained from a digital exchange. The equipment is provided with automatic changeover to a second bit stream in order to ensure continued operation in the case of loss of the first stream. If the accuracy of the bit streams is unsatisfactory, it is still possible to have the FDM pilot and carrier generation controlled by an external basic frequency or frequency comparison pilot.

Figs. 5 and 6 show how the equipment can be connected for different general synchronization alternatives.

Loop connection of the recovered timing from the incoming to the outgoing TDM bit stream can be arranged individually for each of the five streams.

Signalling
The frame for the 1544 kbit/s stream consists of 193 bits, divided into 24 eight-bit time slots and one synchronization bit. Two signalling channels per speech channel are obtained by "stealing" the eighth bit in the PCM word and using it for signalling in the sixth and twelfth frame of the multiframe. Low-level channel-associated out-band signalling, 3825 Hz, is used on the FDM side. This choice of low level (−20 dBmO) means that both discontinuous and semi-continuous signalling schemes can be used without restriction.

Fig. 5, right
Synchronization of ZAJ 120-3 when working towards a synchronous network (digital exchange)

– – – Incoming timing

—— TDM signals

—— FDM signals

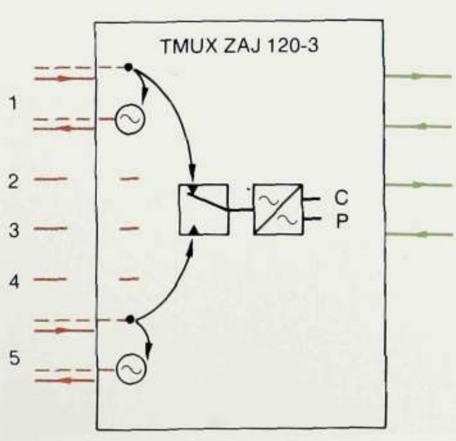
C Carriers

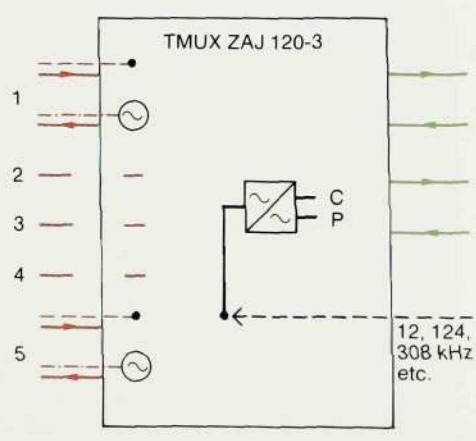
P Pilots

) Synchronization path

Fig. 6, far right
Synchronization of ZAJ 120-3 when working towards an asynchronous network

– · – Outgoing timing





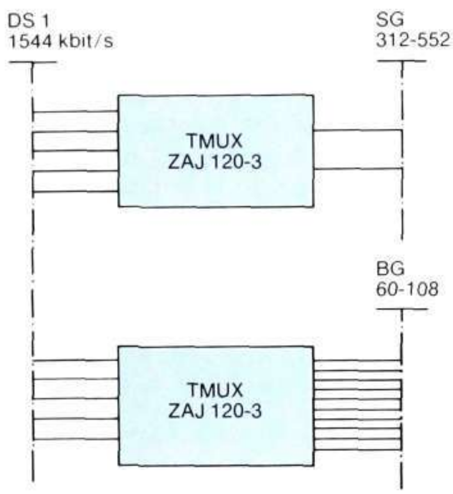


Fig. 7
Connection of groups and supergroups

BG	Standardized basic group interface 60–108 kHz
SG	Standardized basic supergroup interface 312–552 kHz
DS1	Standardized 1544 kbit/s interface

The transmultiplexer translates between the channel-associated signalling channel in the TDM and FDM signal respectively.

The equipment is fully transparent to different signalling schemes as long as only one signalling channel (A-bit) is used per speech channel. It can also be adapted, independently for each of the two transmission directions, to either of the two possible signalling states in which "tone" corresponds to "1" or "0".

The transmultiplexer can also be adapted for the conversion of a two-bit (A- and B-bit) signalling diagram to out-band signalling and vice versa. This signalling diagram, FX (Foreign Exchange), requires extensive recoding of the signalling information. The recoding is carried out with the aid of instructions stored in a PROM.

Transmultiplexers with FX conversion can easily be switched between different directions of traffic, as well as for the fully transparent E&M conversion.

The equipment is also fully transparent to channel-associated in-band signalling and can be used for links with common-channel interoffice signalling, CCIS.

Reliability
Loss of traffic in the long-distance network is expensive. High reliability is therefore a prerequisite for Ericsson's transmultiplexers. It has been achieved through very careful choice of components and by adhering to strict design rules. The number of components has been kept low and extensive tests are made. Low power dissipation gives a low operating temperature; another factor that contributes to high reliability.

The structure of Ericsson's transmultiplexer is also such that few parts are common to several channels. Consequently, the probability that a fault will affect several channels has been reduced considerably. A careful choice of structure contributes to high operational reliability.

Flexibility
Particularly in large networks the supergroup is very suitable as the smallest extension unit. Hence the transmultiplexer can be equipped with a supergroup interface on the FDM side. This means that separate group/supergroup equipment and distribution racks are not needed, with consequent savings in both cost and space.

In some parts of the network it is desirable to be able to branch the traffic into smaller routes. In such cases the transmultiplexer can easily be equipped with a group interface, fig. 7.

Operation and maintenance

Ericsson's transmultiplexers are designed so that the equipment is extremely stable throughout its life. Preventive maintenance is therefore not necessary.

The equipment monitors a large number of system parameters. If a fault occurs an alarm is initiated, together with consequent actions that are dependent on the type of fault, fig. 8. For example, certain faults result in signal blocking and, after a certain time, in release. The alarm information is processed in a collective alarm unit which also contains an alarm interface. Local and remote alarms are indicated by a red and yellow LED respectively on the alarm unit. A digit display shows the faulty TDM bit stream(s).

Loop connection of individual TDM streams can be initiated by means of a push-button. At the same time a green LED is lit and information is transmitted to the far end. Alarm interfaces are available for bay and system alarm outputs.

A push-button gives alarm cut-off (ACO).

Fig. 8
Alarm indications and consequent actions. Certain actions are optional and delay can be arranged in som cases

AIS	Alarm indication signal

Frame alignment error	→	– **LED indication** – **Bay and system alarm** – **Alarm information to the far end** – **Channel blocking** – **Signalling channel to idle state** – **Group pilot cut-off**
Failure of the incoming group pilot	→	– **LED indication** – **Bay and system alarm** – **AIS to the far end** – **Signalling channel to idle state**

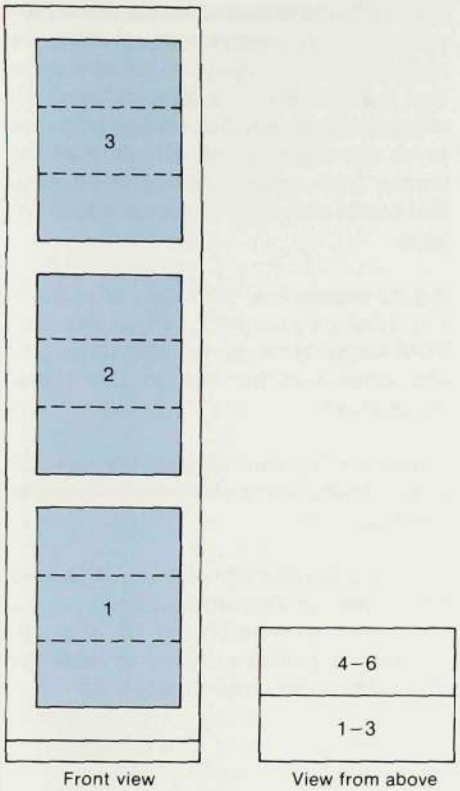


Fig. 9
Six ZAJ 120-3 transmultiplexers, for a total of 720 channels, mounted in two transmission bays placed back to back

If desired, the LEDs can be fed from a separate alarm voltage supply in order to ensure indication during power failures. For the same reason the two relays used for this purpose are always in the operate state during normal operation. At the outputs a choice can be made between the make or break function.

The system and primary alarm interfaces can be connected to Ericsson's transmission maintenance system ZAN 101[2] or ZAN 201[3].

In a complex network the incoming FDM signals can have residual level errors. Such errors can easily be compensated by means of manual level regulation.

Regulation can be carried out jointly for a whole group and individually for each channel. The pilot frequencies of incoming groups are supervised by a pilot receiver. Test points are located at relevant points in the system.

Fault-clearing maintenance consists of changing faulty printed board assemblies, and a suitably dimensioned stock of spare parts is recommended. A stock of only twelve different units makes it possible to replace any unit in the equipment, including the alarm unit, pilot receiver and d.c./d.c. converters.

System construction

Construction practice BYB[4] is now used for all new Ericsson transmission systems. One of the features of this construction practice is that the whole system is mounted in a pre-wired magazine, which also contains the necessary d.c./d.c. converters. The magazine is then installed in, for example, a bay. A new transmission bay has been developed which provides full flexibility and easy installation. The flexibility is ensured by consistent use of the "bookshelf" principle, with magazines as the books. The magazines have front connection and are placed on shelves, on which the magazines are mounted and which also carry the cables. The whole front of the bay is then covered with front plates in different sizes depending on the magazine sizes.

An alarm and power distribution panel is mounted at the top of the bay. Primary power to the various equipments is distributed in parallel, via eight automatic circuit breakers mounted in the panel, to a power distribution bus in the right-hand bay upright.

The panel also includes an alarm unit which collects alarm information from the different equipments and indicates bay alarms.

Fig. 10
A transmultiplexer plant in the US

Fig. 11
ZAJ 120-3, transmultiplexer for 120 channels

Two transmission bays, 8'2" (2489 mm) in height, can be mounted back to back and together hold six ZAJ 120-3 transmultiplexers, fig. 9. Fig. 10 shows a transmultiplexer plant in the US.

ZAJ 120-3

The transmultiplexer for 120 channels, ZAJ 120-3, fig. 11, converts five 1544 kbit/s PCM bit streams to two standardized basic supergroups in the frequency band 312–522 kHz and vice versa. Alternatively the bit streams can be converted into ten standardized basic groups in the frequency band 60–108 kHz.

The mechanical construction of the equipment is in accordance with Ericsson's construction practice BYB. The whole of the equipment is mounted in a pre-wired triple magazine having a width of 12 building modules (488 mm) and a height of 18 building modules (732 mm). The magazine is an independent unit which also contains the necessary d.c./d.c. converters. It can be mounted in Ericsson's transmission bay, a BYB row or a BYB cabinet. All external cabling is accessible from the front. An adapter kit, which also includes front plates, allows the equipment to be mounted in standardized 23" bays.

Fig. 12 shows the functional structure. The TDM part consists of five identical PCM subsystems, which are developed and optimized for use in the transmultiplexer.

Custom integrated circuits are used in order to reduce the volume of the equipment.

The units in the FDM part are similar to those used in Ericsson's FDM systems but have been optimized for this application and also adapted to meet the alarm handling requirements.

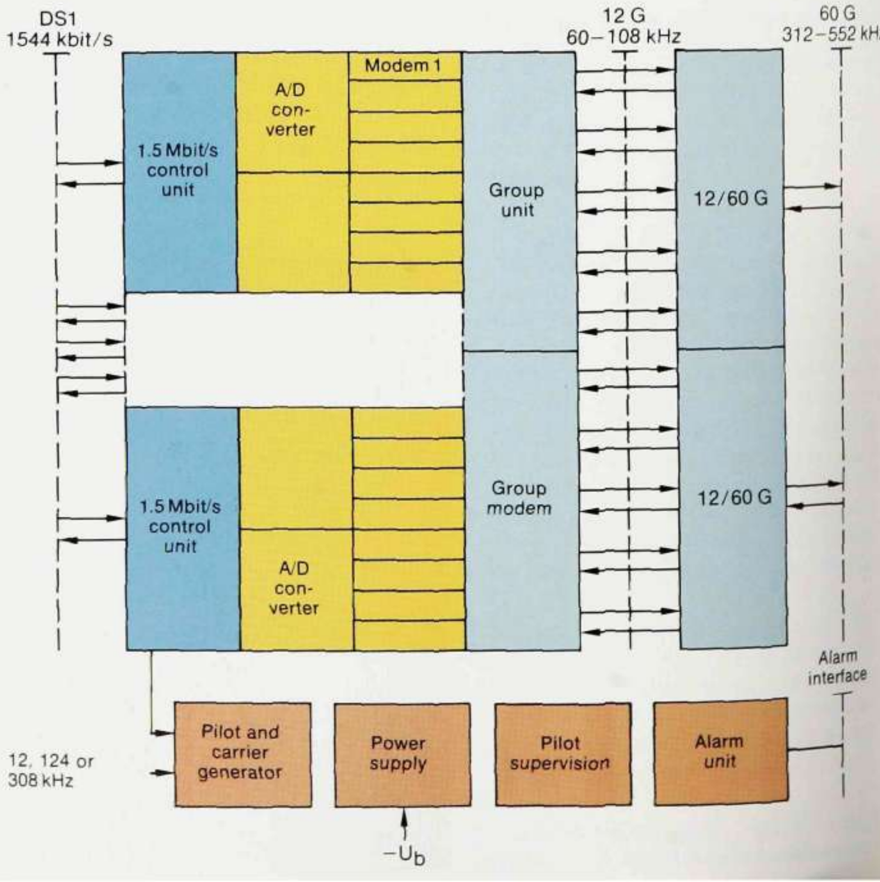


Fig. 12
Block diagram of transmultiplexer ZAJ 120-3

The necessary carrier, signalling and pilot frequencies are all generated internally with the aid of phase-locked oscillators. Three newly developed units handle these functions. One of these units can be omitted if only group interfaces are used on the FDM side.

All frequencies, including the group and supergroup pilots, are derived from a single control frequency. This frequency can be obtained from an external basic frequency or frequency comparison pilot, or from the incoming PCM bit stream if it has sufficiently high frequency accuracy.

The conversion between the out-band signals in the FDM part and the signalling bits in the TDM part takes place in the same units that carry out the speech channel conversion.

The transmultiplexer is powered by three d.c./d.c. converters which are fed from a battery voltage of between −20 and −72 V.

The common alarm unit collects and processes all alarms. It also contains the necessary alarm interfaces.

Conclusion

The efficient transmission transfer function and the well defined interfaces of the transmultiplexer makes it a simple and flexible equipment. Ericsson's modern transmultiplexers in the BYB construction practice are characterized by great flexibility, small volume, high reliability and good performance.

Technical data for ZAJ 120-3

Capacity	120 channels
Digital interface	
Code	Bipolar, μ-law
Nominal bit rate	1 544 kbit/s
Tolerances	± 50 ppm
Number of bits per frame	193
Number of frames per multiframe	12
Built-in equalization for up to	655 feet
Analog interface	
Two basic supergroups	312–552 kHz
Nominal output level	−35, −36 dBr
Nominal input level	−23, −30 dBr
Impedance	75 ohms
Ten basic groups	60–108 kHz
Nominal output level	−26, −36, −37 dBr
Nominal input level	−8, −23, −30 dBr
Impedance	75 or 150 ohms
Power consumption from battery	110 W
Magazine dimensions height×width×depth	732×488×220 mm

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565 Mbit/s Optical Fibre Line System

Björn Johansen and Bo Stjernlöf

Ericsson has developed a 565 Mbit/s optical fibre line system to meet the demand for high capacity systems, especially in North America. The system which has a capacity of 8064 telephone channels operates over single mode fibre at a wavelength of 1300 nm. Repeater spans of 30–40 km can be bridged corresponding to a system attenuation of typically 28 dB. Systems of this capacity have proved economically competitive compared with other transmission media. Their main application will be in the long distance and trunk networks.
The authors describe the system and its design, construction and performance.

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telecommunication
transmission lines
optical cables

Optical fibre line systems operating at a wavelength of 1300 nm and at bit rates higher than 140 Mbit/s demonstrate considerable economical advantages compared with systems using other transmission media, such as radio links, coaxial cables or satellites. This result is achieved thanks to wide repeater spacing (30–40 km) and by installing the fibre cable alongside railroads and highways at low installation costs.

The demand for systems of very high capacity arose initially in the US. To meet this demand Ericsson therefore decided to develop ZAM 565-1, a 565 Mbit/s system for 8064 channels per fibre pair. The system is a longwave single mode system for a wavelength of 1300 nm. The optical light source is a laser diode and the detector is a PIN-FET module.

The system tolerates a dispersion of 0.1 ns/nm and bridges an attenuation of typically 28 dB at BER 10^{-11}. These data in combination with the low attenuation of the single mode fibre (< 0.5 dB/km) and the long delivery lengths of the cable permit repeater spacings of 30–40 km.

The fundamental system requirements are

- system attenuation ≥ typically 28 dB
- permitted dispersion 100 ps/nm
- high jitter tolerance
- low output jitter
- high reliability
- low power consumption.

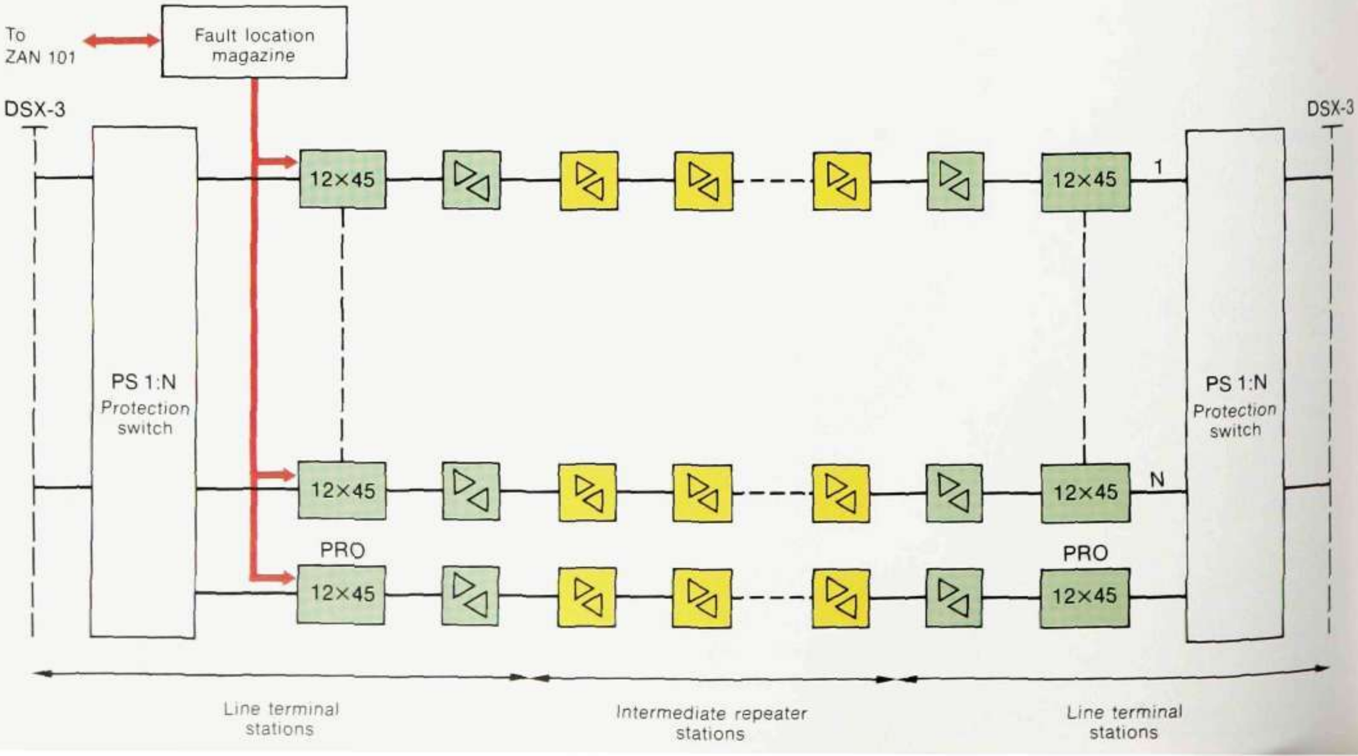
These requirements have been met thanks to the use of up-to-date technology, such as high-speed gate arrays in the digital sections of the multiplexer and advanced hybrid circuits in the line repeaters.

Fig. 1
Equipment in the 565 Mbit/s system

Muldex and terminal repeaters

Intermediate repeaters

PRO Standby line



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Technical Data

Electrical interface	
Bit rate	44 736 kHz ± 20 ppm
Code	B3ZS
Impedance	75 ohms res. unbal.
Return loss 0.5–50 MHz	15 dB
Equalization range	0–450 feet of W.E.728A cable
Frame structure	
Frame alignment word	1 1 1 1 1 0 1 0 0 0 0 0
Loss of frame alignment	Detection of 4 consecutive incorrect frame alignment words
Frame synchronization	Detection of 3 consecutive frame alignment words
Transmission quality	
Bit error rate per repeater	$< 10^{-11}$
Input jitter tolerance at DSX3	> 5UI at 2.3 kHz
Output jitter at DSX3 with no input jitter	< 0.3 UI peak to peak (1 UI = 22.35 ns)
Optical interface	
Symbol rate	600 Mbaud
Optical signal code	Scrambled binary plus parity bit insertion 16B1p
Modulation format	NRZ
Wavelength	1.29–1.32 μm
Spectral width (FHWM)	3 nm
Typical system attenuation at BER 10^{-11}	28 dB
Permitted dispersion	0.1 ns/nm
Max. permitted input power	−15 dBm
Transmission medium	
Fibre	Single mode optical fibre
Fibre diameter	125 ± 3 μm
Mode field diameter	10 ± 1 μm
Cut-off wavelength	1.1–1.27 μm
Primary power source	
Battery	−36, −48 or −60 V
Tolerance	± 20%
Power dissipation	
Muldex	240 W
Terminal repeater	25 W
Intermediate repeater	30 W
Fault location	
Transmission medium	Traffic carrying fibre
Max. number of fault detector units connected to the fault location channel	255
Service channel	Via traffic carrying fibre 2-wire or 4-wire interface

CAD-support has been used extensively throughout the design stages of component, circuit and system realization. Great care has been taken to ensure that the design margins are so large that reliable performance during the entire operating life of the system can be guaranteed.

ZAM 565-1 is provided with fault location, alarm and service channels over the traffic-carrying fibre. The fault location system ZAN 201, which is Ericsson's general fault location system for digital line systems, allows error rate measurements during operation. The fault location system uses a standardized communication channel. This permits centralized fault location of a transmission network that contains different types of digital line systems. These may use different transmission media, such as fibre cables or coaxial cables. ZAN 201 may also be integrated into the transmission maintenance system ZAN 101.

ZAM 565-1 offers the following advantages:
- wide repeater spacing, 30–40 km
- temperature-stabilized laser diode for maximum reliability
- the use of a PIN-FET module gives a reliable receiver design with high sensitivity and high immunity to temperature variations
- scrambled binary line code with parity bit insertion for simple error monitoring and rapid resynchronization
- custom-designed monolithic and hybrid circuits with a high degree of integration for improved performance and high reliability
- compact mechanical design of terminal and intermediate repeaters
- simple connection of the fibre by low-loss optical connectors
- flexible mechanical design using Ericsson's standard construction practice, BYB. Easy installation and operation. All connections for signal paths, alarm outputs, test and measuring points as well as power supply are accessible on the board fronts.
- fault location and service channel via the traffic-carrying fibre enable the use of non-metallic cables and connection to the transmission maintenance system ZAN 101.

System characteristics

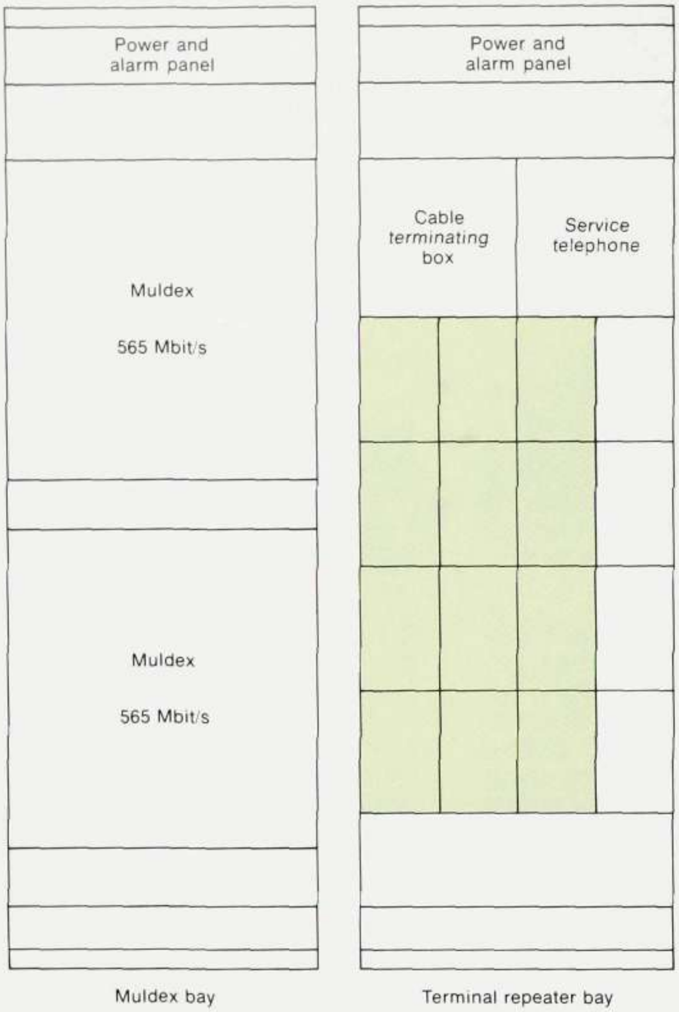
ZAM 565-1 is built with interfaces for the North-American hierarchy. The line equipment together with the optical fibre cable makes up a digital line system, fig. 1, and consists of:
- terminal equipment with an electrical interface DSX3 (45 Mbit/s) and an optical interface 600 Mbaud. In the send direction the multiplexer combines twelve asynchronous DS3 tributaries. The multiplexed signal is scrambled, the parity bit is inserted, after which electro-optical conversion takes place. In the receive direction the procedure is the opposite. The line is supervised and an alarm is issued in case of a fault. The service and fault location channels are connected in via a fault detector unit.
- a two-way repeater which detects, equalizes and regenerates 600 Mbaud optical line signals. Two transceiver boards, one for each direction, perform these functions. Access to the fault location and service channels may be obtained by means of a fault detector board.
- Ericsson's fault location system ZAN 201 may be used for fault location and alarm transfer. For this purpose, a fault location magazine is placed in the supervising terminal, and via a standardized fault location channel the fault location magazine can communicate with the various units and report their status. Access to the service channel is also obtained via the fault detector units.
- To improve the availability of the system, a protection switch, 1:N, may be used in the DSX3 interface. N lines may thereby be protected by automatic switching to a standby line.

Bay

The Ericsson construction practice, BYB, for transmission equipment is described in an earlier article in the Ericsson Review.

The main features are:
- front access to all external cabling
- decentralized power supply
- coordinated alarm handling
- great installation flexibility.

Alarms are indicated by LEDs for each individual system as well as in the bay



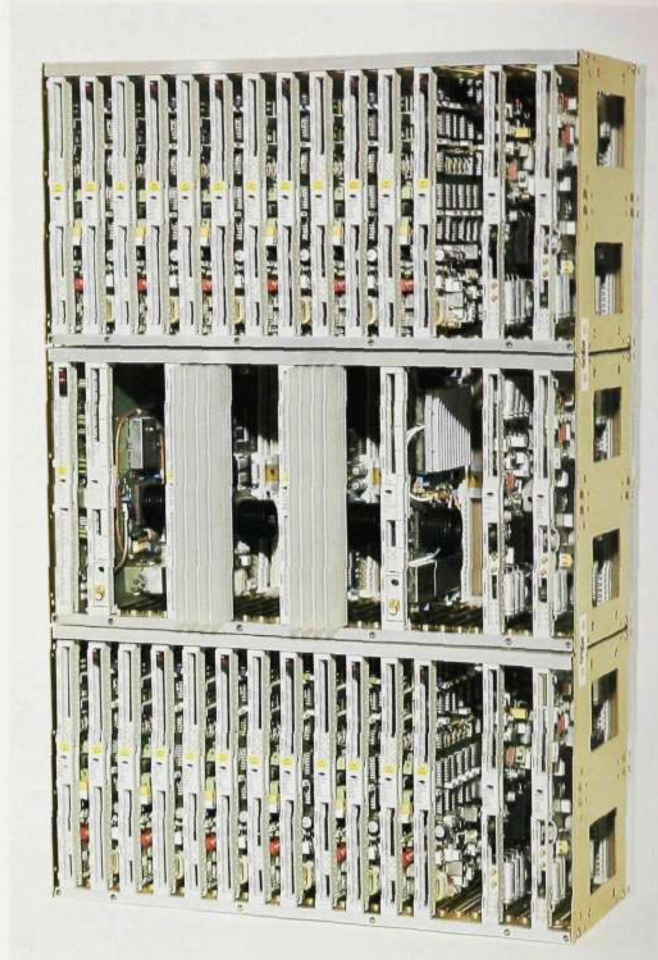


Fig. 2, left
The BYB construction practice permits great flexibility when placing the equipment. Shown here is an arrangement where the Muldex equipments are placed together in Muldex bays while the associated terminal repeaters are placed in separate bays

Terminal repeaters

Fig. 4, right
Muldex for the 565 Mbit/s line system. The interface boards for the DS3 tributaries are placed in the top and bottom parts of the triple magazine while the multiplexing and the high speed signal handling boards are placed in the centre

and in the row. Each individual fault condition is also indicated in the primary alarm interface which may be connected to Ericsson's transmission maintenance system ZAN 101 for centralized alarm collection.

A T-BYB bay may contain two Muldex equipments or, alternatively, twelve terminal repeaters for 565 Mbit/s line systems, fig. 2.

A cable terminating box, a service telephone unit and a power and alarm panel are placed at the top of the terminal repeater bay.

Terminal Equipment

The terminal equipment consists of Muldex (multiplexer and demultiplexer) units and terminal repeaters, fig. 3.

Muldex

The Muldex performs the following functions:

- it adapts the 565 Mbit/s line system to the DSX-3 interface
- it multiplexes twelve asynchronous 45 Mbit/s DS3 tributaries in the send direction
- it demultiplexes the 565 Mbit/s signal into twelve asynchronous 45 Mbit/s bit streams in the receive direction
- it scrambles/descrambles the 565 Mbit/s signal in the send and receive directions respectively
- it inserts and removes parity bits in the send and receive directions respectively
- it adapts the Muldex equipment to the 600 Mbaud electrical interface.

The Muldex is mounted in a triple magazine, fig. 4. The interface boards for the DS3 tributaries are placed in the top and bottom parts. The multiplexing and the high-speed signal handling boards are placed in the centre.

The interface unit receives, detects and decodes the incoming B3ZS coded signal in the DSX-3 interface. The timing of the signal is regenerated and controls the transfer of the signal to the buffer unit.

In the receive direction the incoming binary tributary is recoded to the B3ZS code and adapted to the DSX3 interface.

The buffer unit stores the incoming DS3 signal while monitoring the buffer memory fill. The read-out is timed by the system clock while insertion of the stuffing bits is determined by the degree of memory fill.

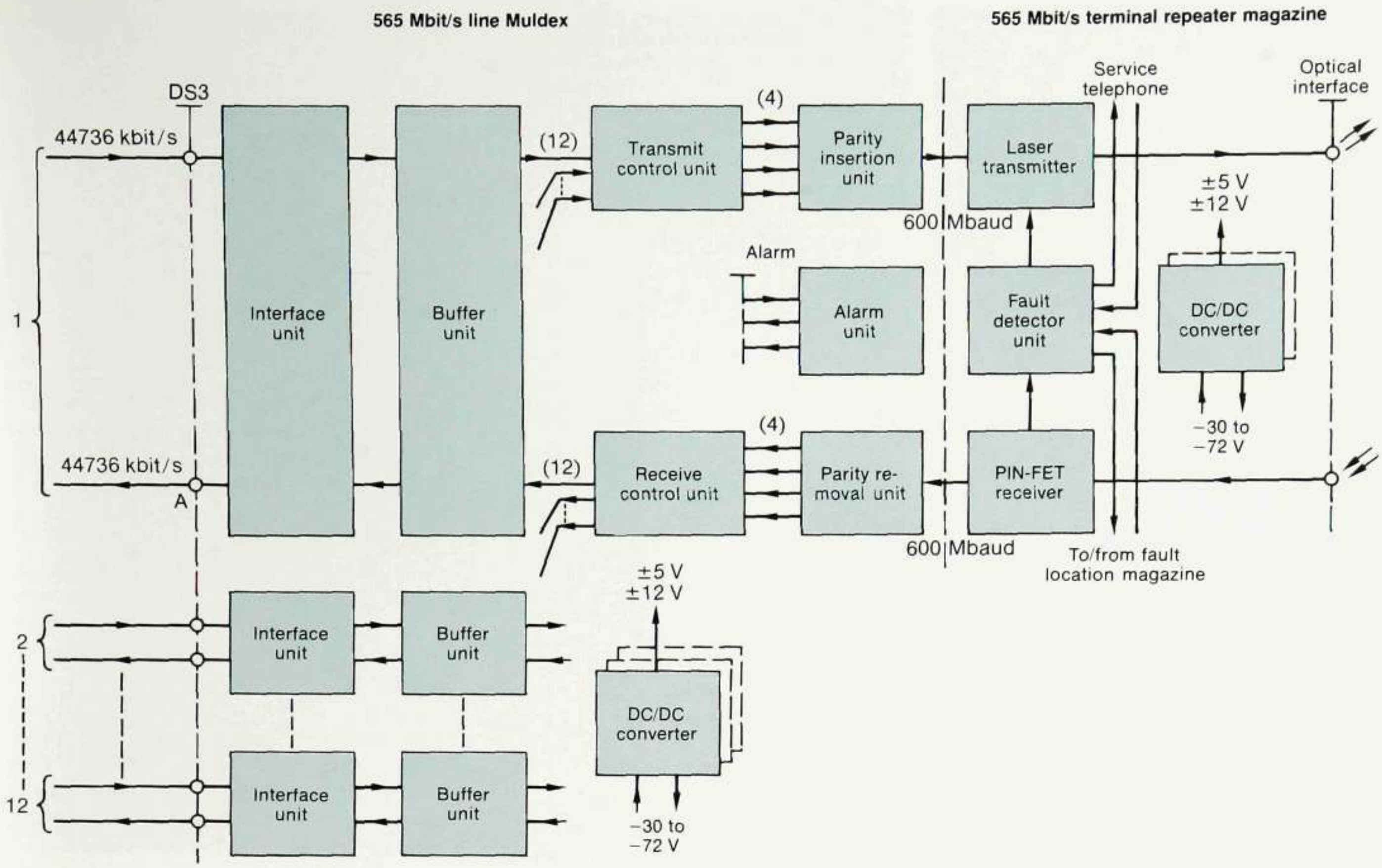


Fig. 3
Block diagram of the 565 Mbit/s line terminal

In the receive direction the incoming tributary is loaded into the buffer memory and the read-out is timed by a crystal oscillator. The frequency of the oscillator is controlled by the degree of memory fill in such a way that the original timing of the tributary is recovered.

The transmit control unit multiplexes the twelve incoming tributaries, including stuffing bits, into four 141 Mbit/s signals. These are processed in a scrambler which is reset at each frame alignment word. Control signals for controlling the buffer unit and the insertion of the frame alignment words are generated in a timing control circuit. In case of loss of the input signal, an AIS (Alarm Indication Signal) or "Blue Signal" may be generated. An interface for access to the "free bits" is provided on the front of the transmit control unit. The four free bits are accessible at a rate of 300 kbit/s each. These may be used to transmit protection switching signals or as an express service channel. This unit also includes the system clock circuit.

In the parity insertion unit the four 141 Mbit/s signals are multiplexed and at the same time a parity bit is inserted after every 16th bit. The timing signal for this 600 Mbaud signal is generated by the system clock. The signal is adapted to a 600 Mbaud electrical interface which enables loop-back during line-up. It also simplifies bit error separation and facilitates the use of protection switching at the 600 Mbaud level.

The parity removal unit receives the 600 Mbaud signal and regenerates timing and data. The parity bit is verified for synchronization and error detection whereupon it is removed and the four parallel 141 Mbit/s bit streams are recreated.

The receive control unit descrambles the four 141 Mbit/s bit streams after frame alignment. Each 141 Mbit/s bit stream is demultiplexed individually into three 45 Mbit/s bit streams. An interface for the four free bits is provided on the front.

The alarm unit indicates the alarm status of the system. An interface for primary, bay and system alarms is provided on the front for connection to a centralized supervisory system.

D.c./d.c. converters are placed in each magazine. The voltages used are ±12V, ±5V, and −4.5V.

Terminal repeater
The function of the terminal repeater is to convert the electrical 600 Mbaud signal into an optical signal in the send direction and vice versa in the receive direction.

The terminal repeater also has an interface for fault location on the line. This interface may be used for connecting a fault location magazine or for through-connection of the fault location channel.

Fig. 5
Two-way intermediate repeater for 565 Mbit/s. Thanks to a high degree of integration it has been possible to build a very compact unit containing two tranceiver units, one fault detector unit and one d.c./d.c. converter. The width is 122 mm, the height 244 mm and the depth 220 mm

Fig. 6
On the 565 Mbit/s transceiver board all high frequency signal processing employs hybrid circuits developed by Ericsson and manufactured by RIFA AB, a member of the Ericsson Group. The picture shows a transceiver unit with the screening of the hybrids removed. The laser unit with internal control of temperature and working point is mounted at the front of the board

Fig. 7
For opto-electric conversion and amplification a hermetically sealed PIN-FET module has been developed in order to achieve high and stable sensitivity within the temperature range. A specially developed photo diode with very low input capacitance, low leakage current and high responsitivity is used. The following amplifier stage consists of a GaAs-FET with high transconductance and a transimpedance resistor made in thin-film technique

An interface is also provided for connecting up the service telephone for communication via the order wire channel.

The terminal repeater is extremely compact and consists of four units housed in a 122 mm wide (size 1) magazine.

The laser transmitter receives the electrical 600 Mbaud signal from the Muldex, controls the gain, recovers the timing information and regenerates the signal. The signal then drives the laser via a driver stage.

The PIN-FET Receiver converts the optical signal to an electrical signal which is gain-controlled and regenerated with recovered timing. The signal is then adapted to the 600 Mbaud interface of the Muldex. The parity bits are monitored for errors.

The fault detector unit is needed for fault location and service channels. Laser alarms and other external alarms are transferred via the fault detector unit to the alarm unit in the Muldex.

D.c./d.c. converters are used for decentralized power supply.

Intermediate repeater

The intermediate repeater is housed in the same compact type of magazine as the terminal repeater (width 122 mm), fig. 5. It consists of two transceiver units in addition to a fault detector unit and a d.c./d.c. converter.

The purpose of the intermediate repeater is to receive optical signals from both directions and, after amplification and regeneration, to retransmit them.

The repeater is designed for local power supply and may be placed in buildings or in manholes.

The transceiver unit, fig. 6, receives the optical signal, and converts it to an electrical signal which is amplified and amplitude-regulated. The timing information is recovered and the signal is regenerated. Via a driver stage the signal then drives the laser which recreates the optical signal.

Technology

Thick-film hybrids and integrated circuits have been used extensively in order to obtain a compact system with high reliability and low power consumption.

Multiplexing and demultiplexing of the tributaries are performed by means of custom-designed integrated circuits in which the highest clock frequency is 600 MHz.

The error rate is monitored in each terminal and intermediate repeater with the aid of a custom-designed integrated circuit working directly on the 600 Mbaud signal.

In the terminal and intermediate repeater the signal processing is mainly analog. All high-frequency signal processing uses hybrid circuits. In addition to soldered component hybrids, a number of chip & wire hybrids are used for the more advanced functions, such as:
- opto-electric conversion and amplification (the PIN-FET module), fig. 7
- timing recovery
- regeneration.

The transistors in these hybrids have a gain bandwidth of 8 GHz.

All hybrids have been developed in co-operation with RIFA AB, Sweden, which also manufactures opto-electronic components. The components used are subjected to exhaustive tests in order to secure high system reliability.

The main transmitter component is an InGaAsP laser built into a special module. Both working temperature and output power are stabilized by special control circuits built on a custom-designed hybrid. The temperature stabilization employs a thermoelectric element which maintains a constant temperature of 20°C in order to ensure maximum reliability.

Great care has been taken to ensure efficient heat dissipation without the need for forced cooling. The laser module with its thermoelectric element and integrated control circuits is therefore built to form a laser unit, which is mounted on a heat sink, fig. 6, with large effective cooling area.

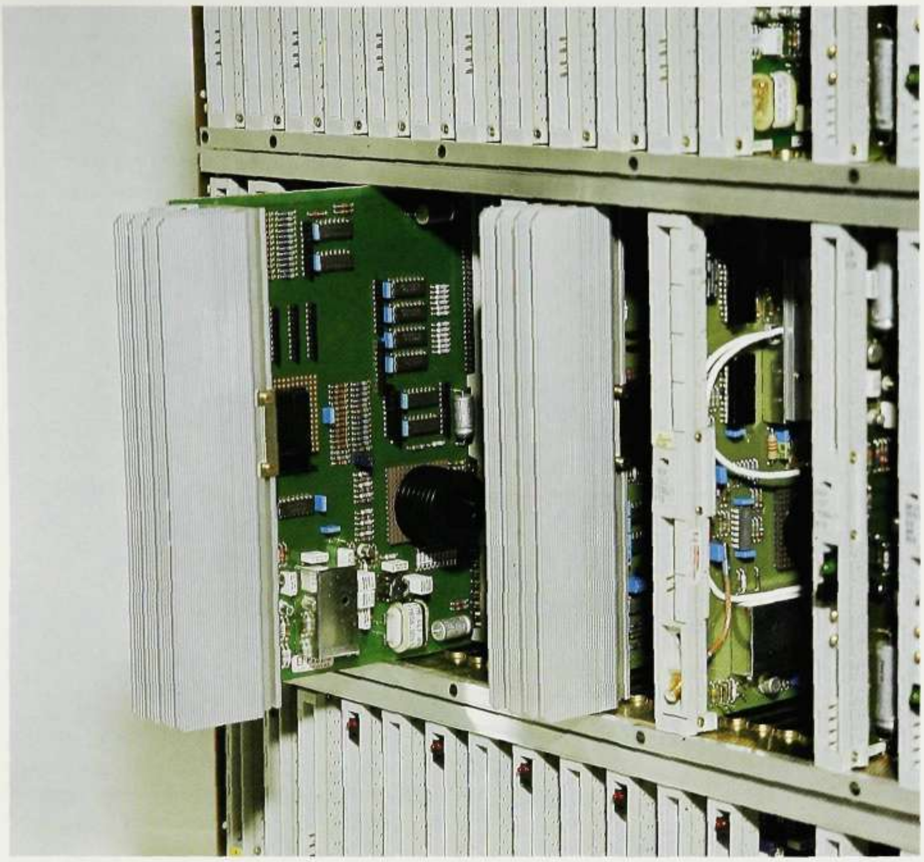


Fig. 8
In the centre part of the Muldex magazine, front cooling is used for certain integrated circuits. In the transmit control unit the connection of an integrated circuit to the frontal heat sink can be seen. For circuits with lower heat dissipation requirements, "tower"-shaped heat sinks are used. These cooling devices are used in order to avoid the need for forced cooling using fans in the bays

Fig. 9
The traffic-carrying fibre is used also for fault location, alarm and service channel by using analog modulation of the digital signal. The fault location system permits bit error measurements during traffic. A central work station can be used for fault location throughout a transmission network consisting of different types of line systems

The opto-electric conversion in the receiver is one of the most critical functions in the system since it is one of the major factors determining the maximum repeater spacing. Special design efforts have therefore been made to attain state of the art performance. The receiver employs a custom-designed PIN-FET module in chip-and-wire technique, and the opto-electric conversion is performed by an InP-type PIN diode, fig. 7. PIN diodes are characterized by low input capacitance, low leakage current and high sensitivity. The input stage is of the transimpedance type using a GaAs transistor. The critically important transimpedance resistor is a specially designed thin-film component.

The FET module must be hermetically sealed in order to ensure a long life. Special attention has therefore been paid to the sealing of the module where the fibre enters it. A gold film is vapour deposited onto the surface of the fibre, which can then be soldered to the capsule. This method ensures a perfect, long-lasting seal.

Mechanical Design

ZAM 565-1 has been developed for high reliability without the need of forced cooling. Special cooling techniques have therefore been developed to dissipate heat effectively, such as front cooling of certain boards. The front cooling is used for certain integrated circuits, fig. 8, and for the laser. All the cooling arrangements fit into the normal BYB construction practice.

Fault location and service channel

For ZAM 565-1 Ericsson has developed a fault location system which is used to pin-point faulty line repeaters, fig. 9. The fault location system permits error rate measurement during traffic. In optical fibre systems it is often desirable to use non-metallic cables. A system has therefore been developed that allows the fault location information to be transmitted

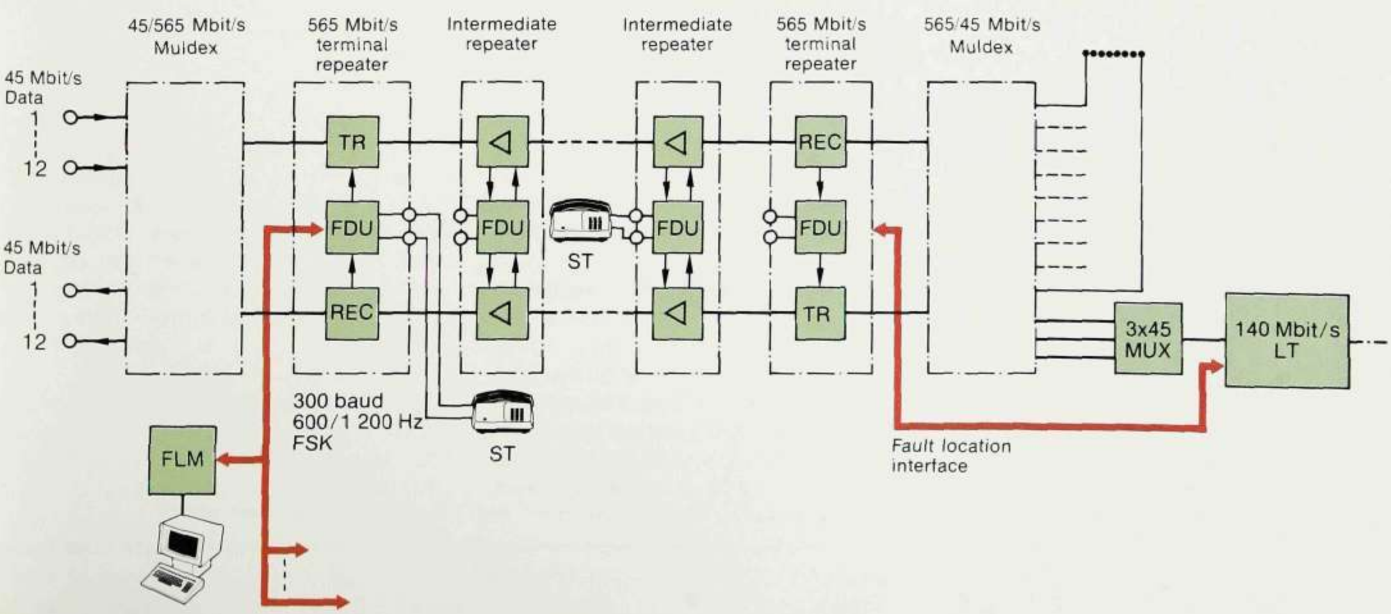
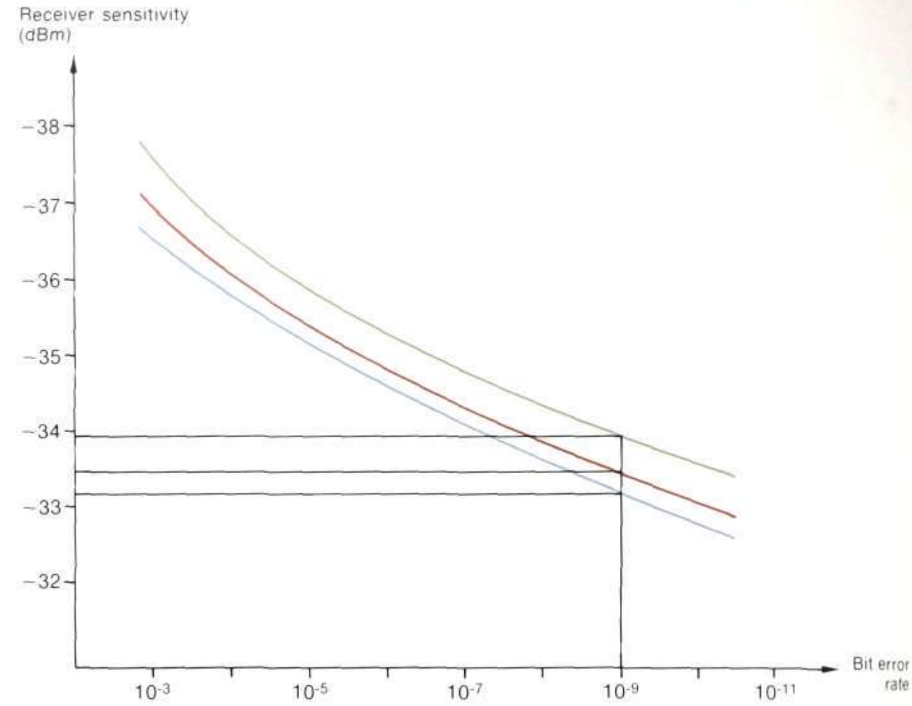


Fig. 10
The receiver sensitivity of the transceiver as a function of the bit error rate is shown in this figure (three different temperatures). The measurement was made after 30 km of fibre and an optical attenuator for regulating the input power

25°C
70°C
80°C



on the traffic-carrying fibre. The fault location and order wire signals are transmitted by means of low-frequency modulation of the mean optical power of the line signal.

For fault locating purposes, line terminals and intermediate repeaters are equipped with a printed board assembly for monitoring bit errors in the transmission. This board, FDU (Fault Detector Unit), is connected to the transmitter and receiver boards in the terminal repeater magazine and to the transceiver units in the intermediate repeater magazine via the rear plane. The board receives and processes bit error pulses and also contains circuits for the transmission of laser alarms and service channel communications via the fibre cable. The service telephone is connected via a connector on the front of this board.

System Measurements

Measurements on the system show very good agreement with the calculated performance. The sensitivity of the optical receiver is very stable over the whole temperature range, fig. 10, and is in full compliance with the specified requirements. A typical eye diagram is shown in fig. 11.

Exhaustive jitter measurements at the DSX3 interface show an extremely low output jitter, fig. 12, and very high input jitter tolerance, fig. 13. The output jitter in the DSX3 interface has been measured during a period of 54 hours in order to obtain the desired resolution.

Fig. 11
Eye diagram at the detection point after 30 km of fibre

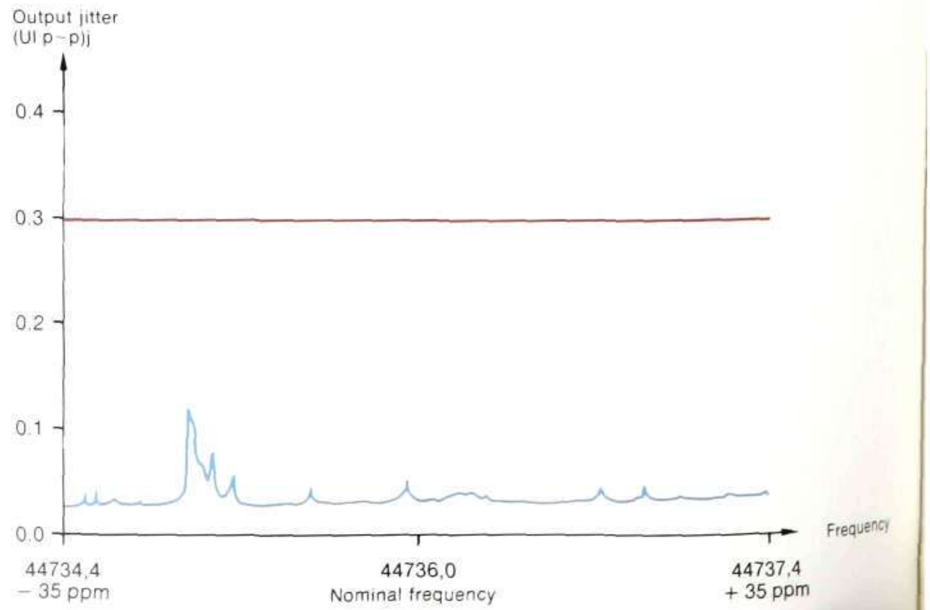
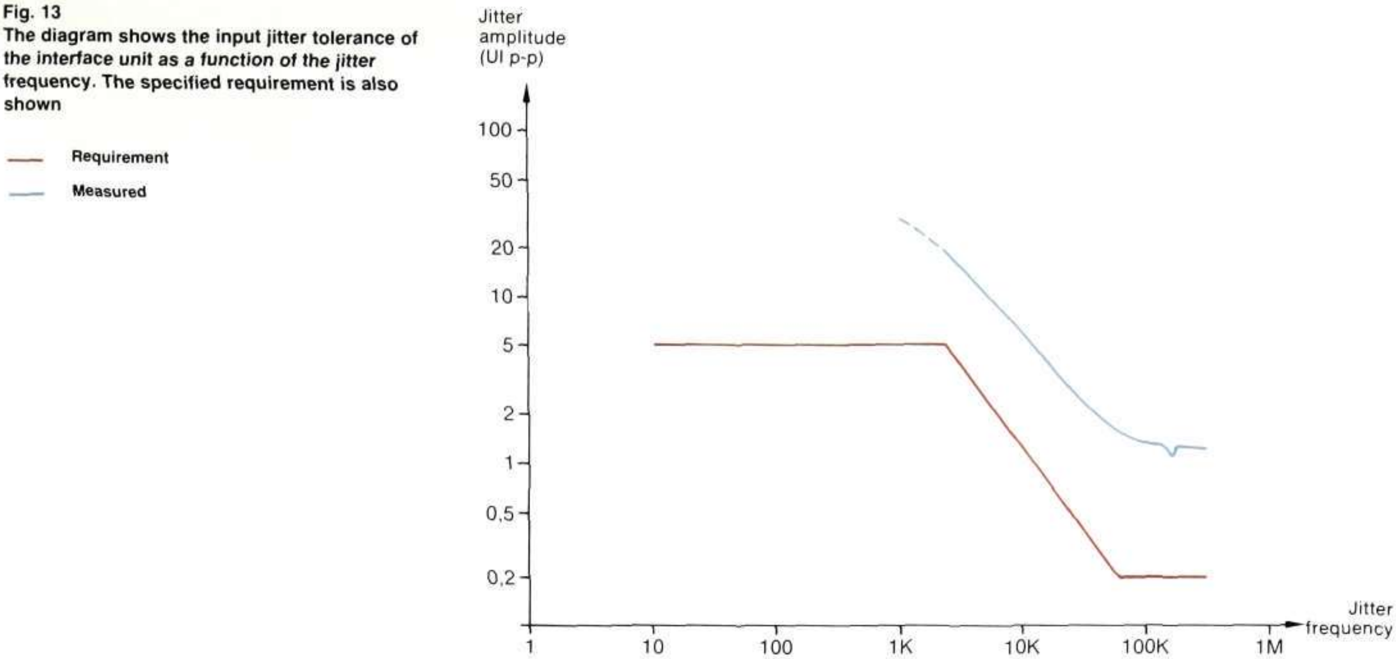


Fig. 12
The diagram shows the total output jitter from an interface unit in the Muldex as a function of the nominal frequency of the signal (45 Mbit/s). The jitter was measured during 54 hours in order to obtain the desired resolution

Requirement
Measured

Fig. 13
The diagram shows the input jitter tolerance of the interface unit as a function of the jitter frequency. The specified requirement is also shown



Conclusions

The optical fibre line system ZAM 565-1 is a product which combines excellent technical performance with high reliability. It is modular, robust and easy to handle and install; characteristics which are important from the user's point of view. This has been achieved by:

- exploiting experience gained from the design and installation of other digital optical line systems, such as ZAM 34-2[4], ZAM 140-1[5], ZAM 140-2[6] and the coaxial line systems ZAY 140-1 and ZAY 140-2.
- providing an electrical 600 Mbaud interface which permits the Muldex and the line equipment to be measured and lined-up separately
- using plug-in, low-loss, non-adjustable optical fibre connectors
- mounting the equipment in T/BYB or 23" bays
- arranging fault location and service channels over the traffic carrying fibre
- using Ericsson's general fault location system ZAN 201 which permits error detection during traffic
- using active continuous temperature stabilization and performance monitoring of the laser
- using PIN-FET receivers for state-of-the art performance and reliability
- using custom-designed monolithic and hybrid circuits to achieve high reliability and performance
- computer-aided simulation and optimization of system performance and circuit design.

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All Telephone Functions in One Chip, PBL 3780

Bengt Berg

RIFA AB has designed and is now marketing an integrated circuit, PBL 3780, in which most of the active functions of a standard telephone set have been incorporated on a single chip. The chip can also be used together with other components in special and system telephones to provide an extended range of functions.

The author describes the requirements and demands met by the chip, the design of the circuit and how it is used together with other components to provide optional functions.

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621.3.049.77
integrated circuits
telephone sets

Previously the outer appearance of the telephone used to be changed often, whereas electrically it remained the same. This is no longer the fact. During the last ten years or so the classical components induction coil, carbon microphone, receiver inset, bell and dial have been replaced by modern components together with advanced electronics, including one or several integrated circuits.

Special and system telephones have also been developed whith many sophisticated functions in addition to those of the standard set, such as repetition of the last number dialled, abbreviated dialling, loudspeaking function, cordless telephone, personal ringing signal, automatic call-back and automatic answering.

Demands made on the electronics in the telephone

Any telephone, standard as well as special and system sets, has three basic functions: transmission of speech, transmission of digits, and ringing. Speech transmission in its turn comprises the three functions sending signals from the microphone to the line, receiving signals from the line for transmission to the receiver and providing the correct level from the microphone to the receiver in the set (sidetone balancing).

In spite of standardization the demands for functions made by different telecommunications administrations vary considerably. The demands are usually stringent and sometimes they border on the theoretically impossible. In addition to this variation in user demands there is the variation in demands made on the active circuit by different telephone manufacturers, who want modifications as regards, for example, amplification, frequency response and current feed to suit just their microphones, receivers etc.

The demand as regards price versus performance varies greatly. For example, in many countries there is an open market with a price squeeze. Designing an active circuit that meets all these demands has proved to be a complicated matter. Integrated circuits for certain functions have long been available, but no circuit where all functions have been incorporated. Some such circuits have recently been developed experimentally in different parts of the world, but they have not met the different market requirements sufficiently well for general acceptance.

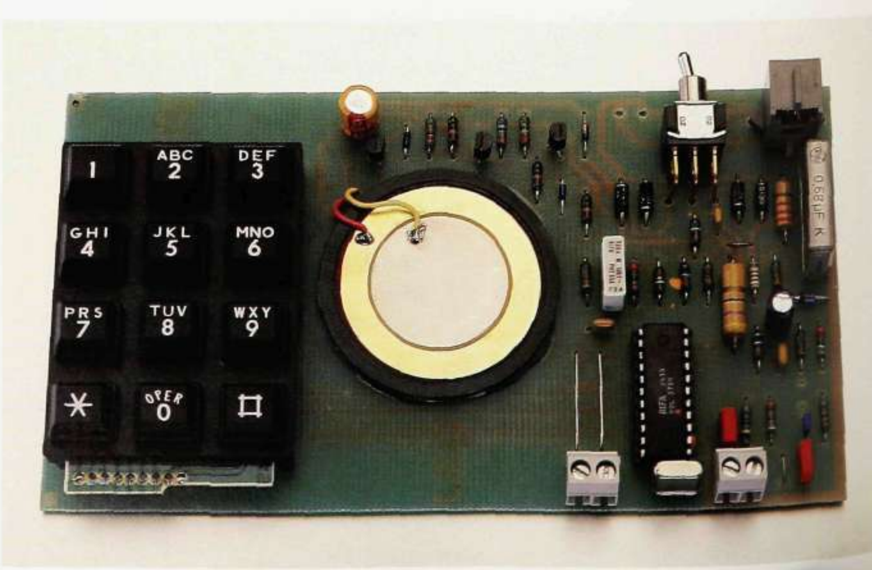


Fig. 1
This printed board assembly, with the integrated circuit PBL 3780, and external components, contains all electronic circuits for a telephone

BENGT BERG
RIFA AB

One chip telephone

The purpose of the one chip telephone is to have one integrated circuit that incorporates all the electrical functions of the standard telephone. The circuit must meet all realistic requirements for the various markets. It must be possible to modify the circuit by means of suitable external components. In addition the circuit must be able to provide the same basic functions in special and system telephones and interwork well with the other components in these sets.

The circuit design must be economically viable, with regard paid to such cost-saving factors as:
- small area for the integrated circuit
- large permissible spread of the process parameters
- suitable division of functions between internal and external, with few and cheap external components
- few connections to the circuit
- design for rapid and simple testing.

PBL 3780

The circuit developed by RIFA for the one chip telephone is designated PBL 3780. It is described here with reference to figs 3 and 4.

Voltage and current requirements

Designing electronic circuits for telephone sets has particular problems because the telephone line transmits speech in both directions and also carries the current feed. The design must therefore provide separation of the parameters for the different functions so that they can be adjusted independently of each other. In addition the amount of voltage and current available is very limited. It is therefore not possible to design the circuit using existing blocks; each part of the circuit must be custom-designed. In addition to feeding the various output stages the current of approximately 1 mA available in the normal speech state must be sufficient for approximately 300 transistor functions down to a voltage across the circuit of only approximately 1.3 V. It should be noted that an a.c. voltage signal is superposed on this voltage, and as a result there is only 1 V available to power the electronic circuits. This requirement is made in order to ensure that parallel connection with mainly older telephones having a carbon microphone and a transformer is possible in cases where the telephone line is long. When the line is short the voltage can be 10-15 V and the current over 100 mA.

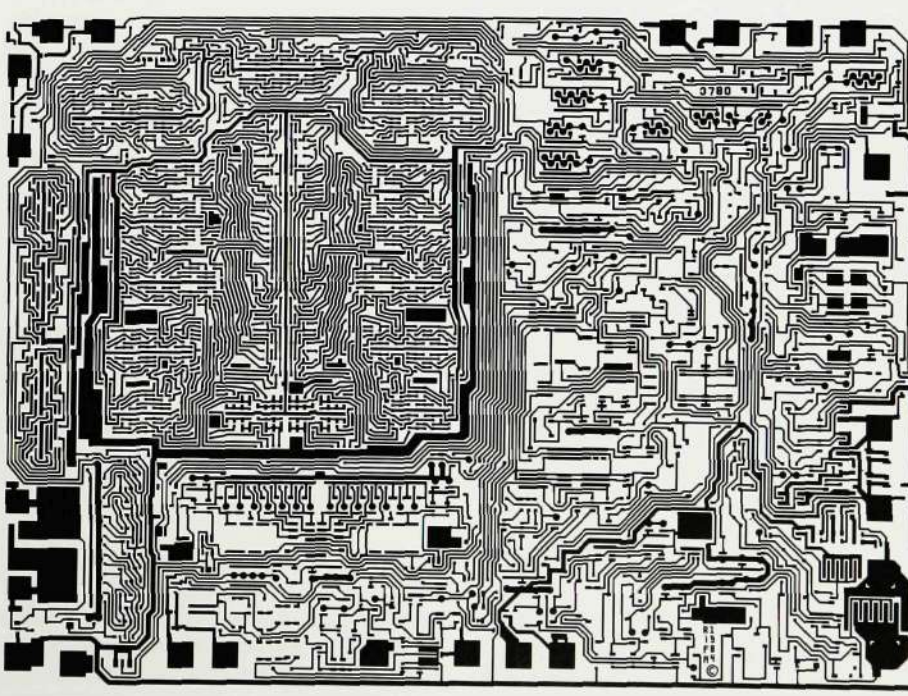


Fig. 2
Connection layout of PBL 3780

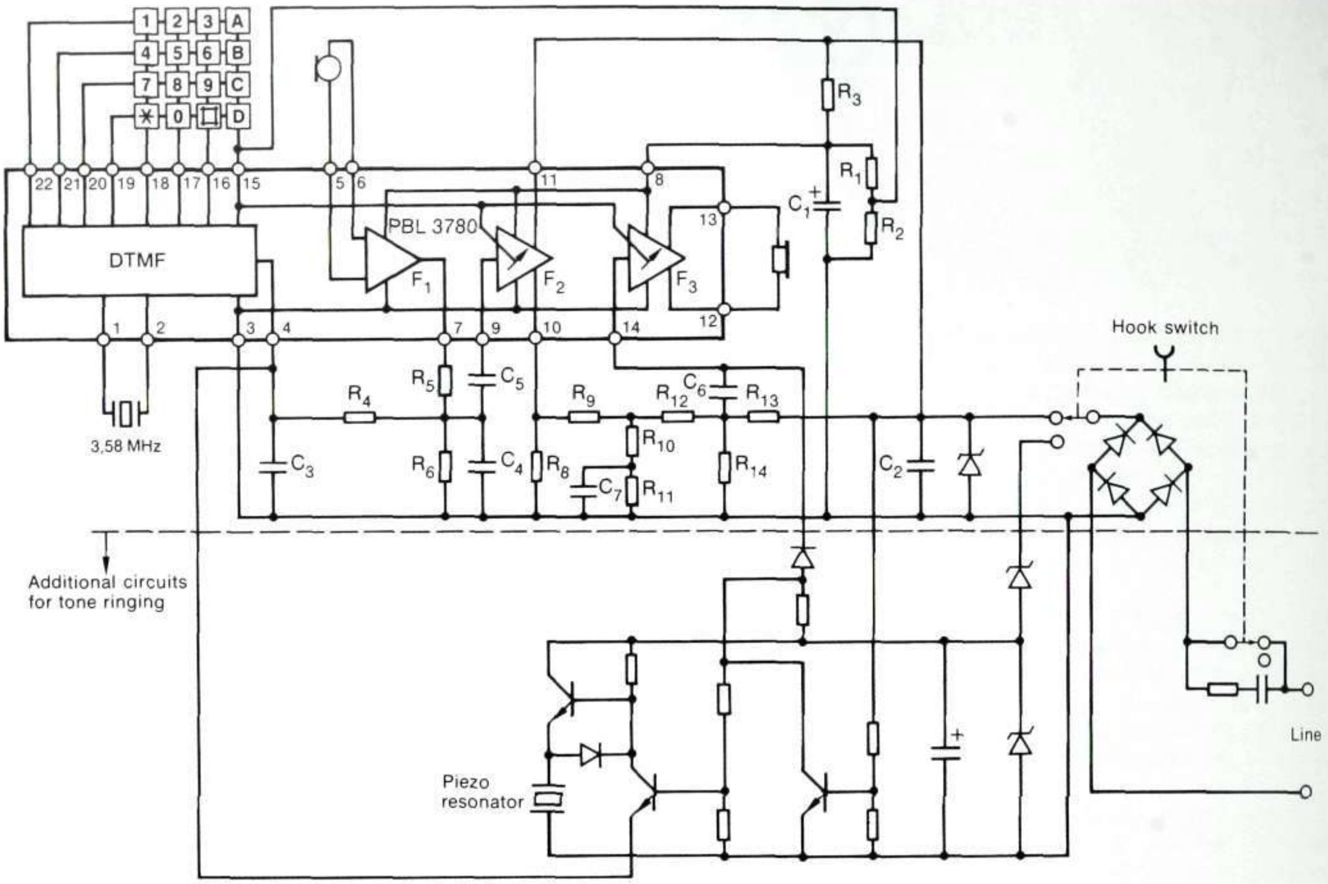


Fig. 3
One application for PBL 3780 is a standard telephone with push-button dialling and additional circuits for tone ringing

PBL 3780

- comprises speech circuit, DTMF generator and tone ringer
- contains approximately 2000 microcomponents
- operates at 1.3 V and low current and hence can also work in parallel with older types of telephones
- has amplification regulation that is dependent on the line length
- permits connection of many different types of microphones and earphones
- contains facilities for controlled digit transmission from a bus
- has built-in current generator for feeding a microprocessor or an abbreviated dialling circuit
- contains circuits that bridge contact bounces in the push-button set
- is equipped with a temperature-stabilized reference voltage for the regulatory functions
- is encapsulated in a 22-pin standard DIL plastic package.

Integrated circuit process and package

RIFA's bipolar integrated circuit process no. 8 was chosen for PBL 3780 in order to meet the above requirements and other demands made on the circuit. This process includes Schottky circuits for fast logic operations, I^2L circuits for compact, low-power logic and compact linear components. The components have good properties, such as low noise; they can take high currents and they can be dimensioned to operate linearly at a collector-emitter voltage of only 0.2 V. However, their dimensions are too small for handling the high voltages that can occur on short lines and with overvoltages. In such cases transistor stacking can be used.

A 22-pin package is used for PBL 3780. This was achieved by giving some connections more than one function.

Impedance towards the line and transmit output stage

PBL 3780 is connected to the telephone line via a polarity inversion bridge and overvoltage protection, the complexity of which is adapted to suit the requirements of different markets.

For the sake of simplicity the a.c. impedance towards the line is adjusted by means of external, passive components. Fig. 3 shows a simple network (R_3 in parallel with C_2).

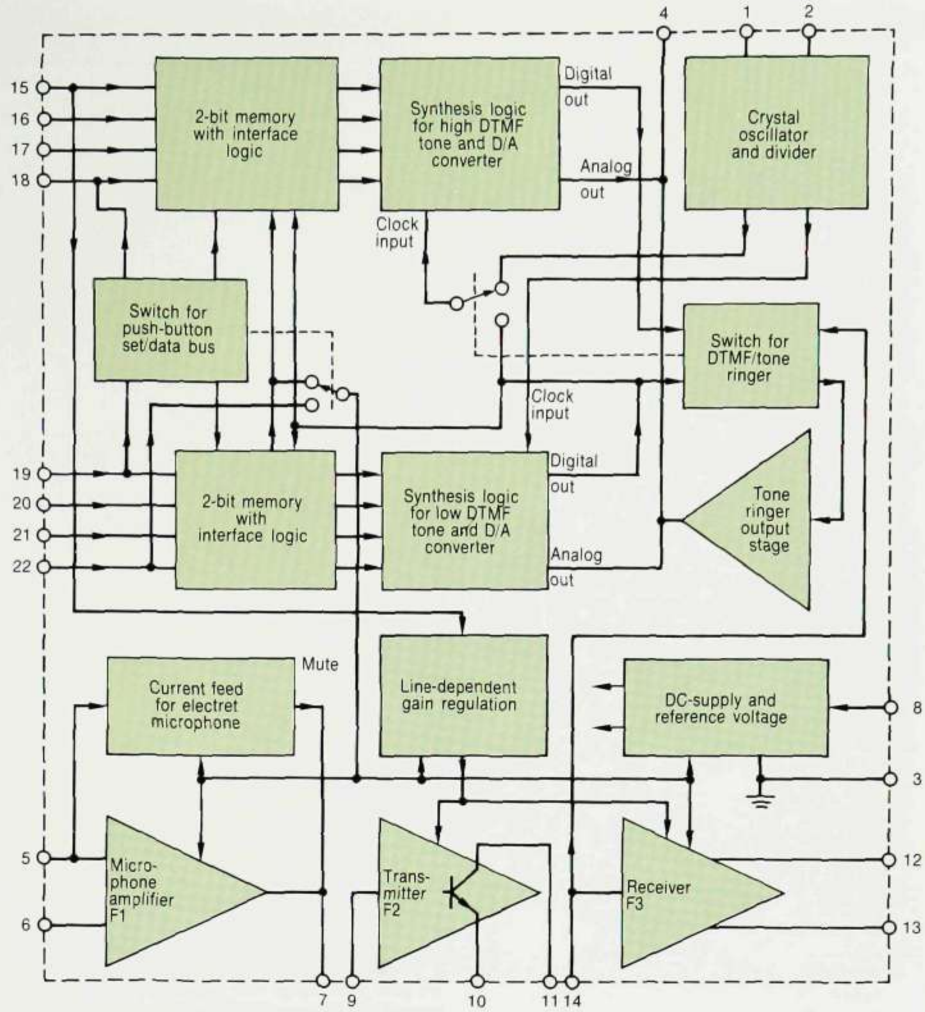
Capacitor C_1 is large and has three functions in addition to its basic function of breaking up the d.c. path in the resistive part of the connection. It also filters the supply voltage to the circuit, controls the d.c. voltage characteristic of the transmit output stage, F2, and works as a filter in the sensing of the line length for the gain regulation. The transmit output stage, F2, functions as a current generator for the signal that is to be sent out on the line and as a shunt regulator in order to give a suitable current-voltage characteristic towards the line. F2 and resistor R_8 constitute a voltage divider. The d.c. characteristic is adjusted with the aid of the resistor, which also takes care of a large part of the power generated in the set in cases where the telephone line is short.

Microphone and microphone amplifier

PBL 3780 contains a microphone amplifier, F1, with a balanced input for microphones having a low output level, such as dynamic and magnetic microphones. Electret microphones with a built-in field effect transistor can be connected via a coupling capacitor to either input. In all these cases the signal is fed from F1 to F2 via an external resistor and a capacitor network which is used to set the required transmission gain and frequency response.

As an alternative an electret microphone can be used, together with RIFA's elec-

Fig. 4
Block diagram of PBL 3780



tret microphone follower PBL 3747. The microphone current feed is obtained on the output of F1 if its inputs are connected to the filtered supply voltage. The output signal from the follower is connected via a suitable resistor-capacitor network to the input of the transmit output stage.

Earphone and receiving amplifier

The signal from the line is fed via the sidetone balancing network and an RC network, in order to obtain the desired amplification and frequency response on the receive side, to the input of the receiving amplifier F3. The amplifier in its turn drives the earphone via a low-impedance balanced output.

The output impedance is low and the driving ability high, sufficient for low-impedance (150 ohms) dynamic as well as magnetic receivers, but the latter require a series resistor. When high-sensitivity telephones are required, e.g. for people with impaired hearing, these telephones can instead be equipped with a high-sensitivity earphone of the rocking armature type.

Sidetone balancing

The sidetone balancing determines to what extent one hears one's own voice when speaking over the telephone; it corresponds to the amplification between the microphone input and the receiver output. The level would be too high if the signal was not attenuated. In PBL 3780 this is done as follows. A signal is obtained across resistor R_8 which is in anti-phase to the transmitted signal. The anti-phase signal is fed via a resistor to an RC balancing network (R_{10}, R_{11} and C_7). The signal from this network and the line signal are weighted and combined by resistors R_{12} and R_{13} so as to obtain a suitable input level to the receiving amplifier. These components are external and can therefore be varied to meet different specifications.

Other speech circuit functions

In addition to the above-mentioned functions the speech circuit comprises three important functions, namely:

Gain regulation of the transmitter, F2, and receiver, F3, which is dependent on the line length and which compensates for the line attenuation. The regulation, which can be switched off, senses the voltage from capacitor C_1 mentioned earlier and shares an input with the push-button scanning function.

Mute control from the tone signalling part to the speech part, which switches off the speech transmission during signalling, reduces the receive amplification in order to give suitable monitoring

PBL 3747

- consists of an electret microphone follower
- has low power consumption <300 μA
- has high input impedance 66 Mohms
- has low output impedance 100 ohms
- is monted in a TO92 package

PBL 3781

- consists of the speech circuit from PBL 3780
- is mounted in a 16-pin bat wing plastic package

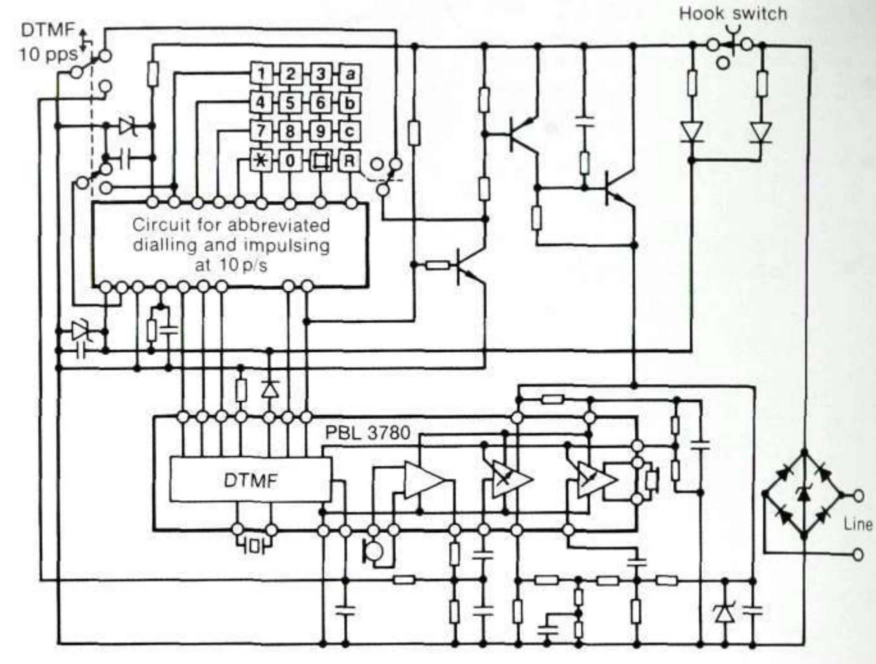


Fig. 5
Application with PBL 3780 and a CMOS circuit for abbreviated dialling and the generation of 10 p/s impulsing. A telephone equipped with this circuit combination provides abbreviated dialling and can be switched between dual tone multi-frequency signalling and conventional impulsing

of the tones and redistributes the current feed in order to optimize the current consumption.

A common, temperature-stabilized reference voltage is used in the regulation of the voltage-current characteristic towards the line, of the line-dependent amplification and of the amplitude of the tones during tone signalling.

The speech circuit is available as a separate circuit in a 16-pin package designated PBL 3781.

Tone generation
PBL 3780 contains a dual tone multi-frequency (DTMF) generator for transmitting the dialled number to the exchange. When one of the push-buttons is depressed two tones are sent, one from each of two four-tone groups, determined by the position of the push-button in the rows and columns of the set.

The tones are synthesized by an oscillator having a basic frequency of 3.58 MHz provided by an external crystal or ceramic resonator. The frequency is divided down to the required values, first by Schottky and then by I^2L logic. The digital signal then controls two four-bit digital/analog converters, each of which generates a stepped sinewave signal. A period has sixteen levels and thirty steps. The step length has been optimized in order to suppress harmonics and is therefore different for different frequencies. When the tones have been mixed, a simple RC network transfers the signal to the transmit output stage. The transmitted signal meets the requirements as regards levels and harmonics that apply to the different markets.

Connection of the push-button set
The DTMF generator can be used with a matrix push-button set having single closures. Each button closes a row to a column. The rows control one frequency and the columns the other. After the sensing the two inputs are treated in the same way. A memory controls the divider logic that generates the different frequencies. The purpose of the memory, which retains information for a couple of milliseconds, is to suppress contact bounce in the push-buttons so that it does not interfere with the operation. The previously mentioned mute signal to the speech circuit is generated simultaneously with the tone generation.

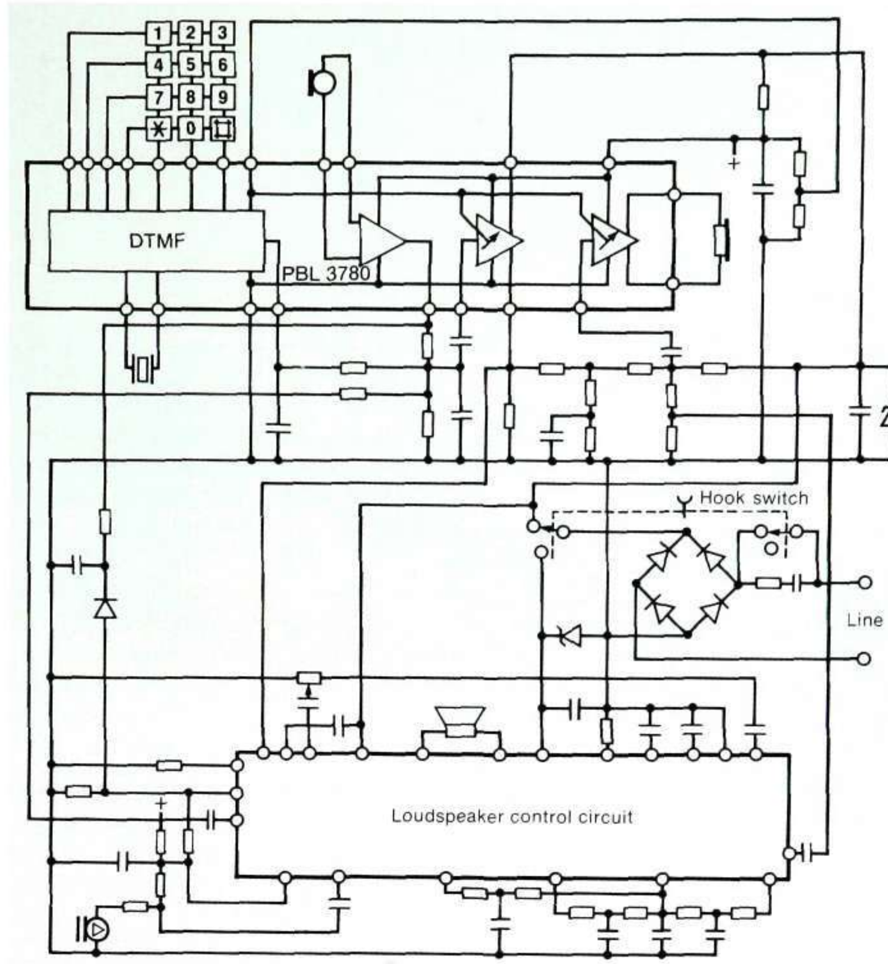
Data bus input
PBL 3780 can also be controlled by a microprocessor or special abbreviated dialling circuits via a data bus input. The push-button set connections are then connected through to a four-bit bus input, an address input and a current output for feeding the external circuits. The memory on the inputs of the circuit is then used as a latch, which means that the four-bit bus can be common to PBL 3780 and a memory, for example.

The tone signalling part, together with a linear output stage that can work directly towards the telephone line, is also available as a separate circiut, designated PBD 3551.

Tone ringing
PBL 3780 includes a tone ringer. The tone signalling logic can be conncted to generate a tone ringer signal to an external output stage, which drives the sound source. The output stage can be adapted to different sound sources, such as a loudspeaker or piezo-ceramic resona-

PBD 3551
- comprises DTMF generator and the output stage of PBL 3780
- is mounted in a 16-pin standard DIL plastic package

Fig. 6
A combination of PBL 3780 and a circuit for loudspeaker control gives all the circuits required for a loudspeaking telephone with tone ringer



tor. The part of the circuit that controls the start and stop of the tone ringing also prevents false ringing signals. There are many external facilities for modifying this function. The signal consists of a low-frequency switching between two higher tones. The tone ringing function uses a very small part of the circuit area and shares pins with other functions. The marginal cost is therefore low and does not affect the economy of the circuit even if special requirements, for example a demand for three tones, makes it necessary to use external ringing circuits.

PBL 3780 in special and system telephones

Special and system telephones with an extended range of functions and services require more integrated circuits than standard sets. Some common additional functions in such telephones are repetition of the last number dialled and abbreviated dialling, either by means of DTMF or impulsing (10 p/s). Fig. 5 shows how PBL 3780 can be used together with a CMOS circuit in order to obtain such a function. The CMOS circuit provides the logic for direct control of PBL 3780. The high woltage transistors that generate the impulsing towards the telephone line are almost the only external addition necessary.

Fig. 6 shows the circuit for a loudspeaking telephone using PBL 3780 together with a circuit for loudspeaker control.

PBL 3780 can also be used in larger systems. External control of the three linear signal paths for sending, receiving and sidetone is easily arranged since all three are accessible at suitable points and related to the same earth potential. The tone signal transmission can be controlled via the data bus. PBL 3780 can thus be used in digital applications, for example in telephones with digit indicators and microprocessors.

Ericsson BCS 10

Mikael J. F. Janson

Ericsson Information Systems AB have developed and are now marketing a small business communications system, Ericsson BCS 10, for up to 15 extensions and six trunk lines. The system can easily be adapted to suit different requirements; it has a compact central unit and is suitable for mass production and sales direct to the customer. Compared with traditional PBXs and systems requiring special telephones, BCS 10 includes more features and functions and permits the connection of more types of telephone sets.
The author describes the properties that make the system easy to market, and also the design and operation of the system, together with the different functions and services that can be obtained with different types of telephone sets.

UDC 621.395.2
private telephone exchanges
features*

Ericsson BCS 10 is a microprocessor-controlled business communications system particularly suitable for small companies or large enterprises and organizations that are divided into a number of smaller units. The system is also intended for domestic use in cases where several telephones and main lines are required. It can be equipped with up to 15 extensions and six trunk lines. The elegant, slimline central unit is designed for wall mounting. Different types of telephone sets with varying possibilities of functions and services can be connected in as desired.

Complete communications system

Both standard telephone sets and advanced sets with more features can be connected to Ericsson BCS 10. For example, a customer can initially use existing telephones and will be able to add new sets of different types at any time.

Incoming traffic can be distributed automatically to different extensions, to a group of extensions or to a common answering position. Ericsson BCS 10 also includes functions that make it an efficient executive-secretary system, multiline telephone system, common call distribution system or internal direct communications system.

The system meets the special requirements of a wide range of applications, for example in banks, travel agencies, mail order firms, garages, security centres, offices and homes.

Rational distribution

The system is designed as a standard product without variants in order to ensure long production series and thereby

Fig. 1
Ericsson BCS 10 – four types of telephones and the central unit

MIKAEL J. F. JANSON
Ericsson Information Systems AB

Fig. 2
The central unit fits in anywhere. It works silently, has an attractive appearance and small dimensions, 660 x 410 x 80 mm

good operational quality and a competitive price. The basic design meets the requirements of many markets. A certain amount of market adaptation is carried out during installation by activating different functional variants.

The central unit, including the power unit, weighs only 8.7 kg in the basic version and is easy to transport and install. The system is built up from a relatively few stock items during installation, and it is in operation as soon as power is connected.

Telephone sets

The telephone sets, which belong to the same family as regards function and appearance, can be system phones (BCS 10-F1, -F2), feature phones (Executive 605) and standard phones (DIAVOX TR-100). The system and feature phones provide more features than the standard telephone. The system phones have been developed for the system, whereas the feature phone can be connected to different systems.

Ericsson BCS 10-F1

Fig. 3 shows a system telephone (F1) which is suitable for executives, secretaries, staff at a common answering position and the system operator. In addition to the normal telephone function it provides two additional main functions. One is the possibility of monitoring the state of all trunk lines by means of a liquid crystal display (LCD). For example, if the third line is parked the figure 3 will be flashing slowly. See also the panel "Indication of trunk line state in F1 and F2".

The second function means that F1 is the data terminal and control unit of the system, from which it is programmed and system data read and modified. For example, if a person in a company changes room and also receives authority to make national calls, the old telephone number is reprogrammed to the new telephone and the possibility of making national calls from that telephone is written in. Programming is simple and can be carried out during traffic. The data communication between the microprocessors in F1 and the central unit is independent of F1 speech communication in progress.

Other features include multi-line access, parking, traffic indication, alarm indication and switching between day and night service.

Ericsson BCS 10-F2

Fig. 4 shows a system telephone (F2) which is suitable for executives, secretaries and staff at a common answering position. This set also provides traffic information, indicating the state of the trunk lines in the way described in "Indication of trunk line state in F1 and F2".

Indication of trunk line state in F1 and F2

extinguished	line free
steady light	line engaged
rapid flashing	incoming, unanswered call
slow flashing	line parked for common access

Fig. 3
The Ericsson BCS 10-F1 is a system telephone with liquid crystal display for system information, traffic indication and call information. It is also a data terminal, multi-line access telephone etc.

Fig. 4
The functions offered by the system telephone Ericsson BCS 10-F2 include multi-line access for all extensions, trunk line indication, traffic indication, indication of switching between day and night service etc.

With this set the system's features for distributing incoming traffic can be exploited to the full. Incoming lines can be allocated to offices or persons – line 2 to a business service department, for example – but calls on these lines can be answered also by other telephones. It is thus possible to provide assistance during periods of temporary undermanning, give priority to the manager's calls and park and resume calls. In particular F2 can be used in a subsystem as a common answering position for temporarily unattended telephones. Other features include parking and multi-line access, traffic indication, alarm indication and indication of switching between day and night service. With the BCS 10 system and F2 telephones large traffic volumes can be handled quickly and easily, a property that is extremely valuable in many businesses where the telephone plays an important role in the striving for profitability.

Ericsson Executive 605

Fig. 5 shows a feature phone (Executive 605), suitable for executives, secretaries and other people who use the telephone frequently. The more important functions include two-way loudspeaking, 36 single-button commands with labelling, nine abbreviated numbers, repetition of the last number keyed, clock, stopwatch and reminder signal.[1]

The loudspeaking function is of high quality and makes it possible to move around the room or, for example, work at a personal computer during the call. It also means that handsfree answering can be used. By this is meant that internal and extended calls are introduced by a short signal, after which the speech connection is established immediately.

Another time-saving feature is the possibility of obtaining 36 individually programmed telephone numbers or system features, such as diversion of calls to a secretary, and do-not-disturb. These functions are ordered by means of 12 function keys, which are depressed once, twice or three times in order to obtain a function.

Ericsson DIAVOX TR-100

Fig. 6 shows standard telephones (DIAVOX TR-100) which are suitable in cases where the information and features offered by the phone described above are not required. DIAVOX TR-100 contains an electronic tone ringer. The volume of the tone ringer is adjustable and the user can choose any one of three signal characters (frequencies). This is useful in applications with more than one telephone in the room.

All telephone sets, including DIAVOX TR-100, can use the common features of

Fig. 5
Ericsson Executive 605 has 36 programmable single-button commands with labelling for abbreviated numbers and system features, high quality two-way loudspeaking function, last number redial, number display, etc.

the system, such as abbreviated dialling, automatic call-back, multi-party conference, group hunting, diversion and storing of last number keyed.

Other telephone sets
It is possible to connect other standard telephones to the central unit. If only sets with rotary dials are used, the printed board assembly containing the key tone receiver, the KRC board, is not required. See also the panel "Features of Ericsson BCS 10".

Traffic control

Incoming traffic can be distributed in many different ways. It is important that the customer can have the distribution best suited to his requirements. Some variants are described below, and they can also be combined in different ways.

Multi-line system
All incoming calls are indicated acoustically in the receiving telephone and by rapid flashing in system telephones F1 and F2, fig. 7a. In many cases the use of these sets does not only save an operator but also gives a more efficient telephone system.

Calls are answered by keying the number of the trunk line. Any free trunk line can be used for outgoing calls.

Operator-assisted system
All incoming calls are routed to a common answering position, which distributes the traffic, fig. 7b. If the called extension is engaged the new call is camped-on and the extension receives a call waiting indication. If the waiting call is not answered within a minute it is connected up to the operator again. Any extension can be appointed the common answering position, but the latter should be equipped with a system telephone F1 or F2.

Automatic distribution system
Each trunk line can be associated with two (first and second) answering positions, fig. 7c. The answering position is set up by programming and can consist of one extension, a group of extensions or extensions that are called by a common signalling unit. Incoming calls are routed to a second answering position if the first extension is busy, with calls waiting, and does not become free within 16 seconds.

Group distribution system
A group with up to five extensions can be formed among all extensions, fig. 7d. Hunting within the group can be sequential (the first free extension in a predetermined order) or evenly distributed (the first free extension after the last selected one).

Fig. 6
Ericsson DIAVOX TR-100 is an electronic standard telephone with 2-wire connection which offers distinctive ringing and many of the system features

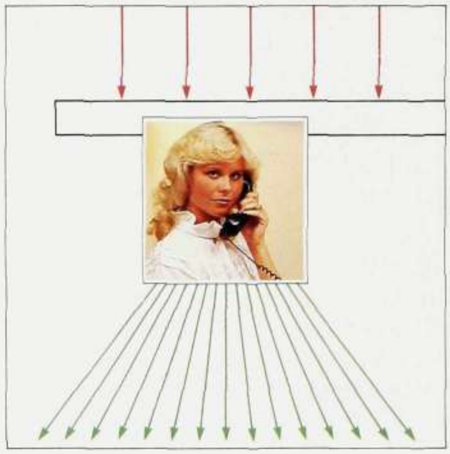
Fig. 7a
Multi-line system with access to all trunk lines

Incoming traffic

The central unit in the system

Automatically or manually distributed traffic to extensions

Executive/secretary traffic



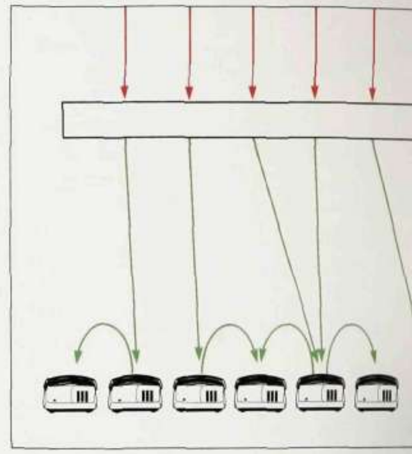
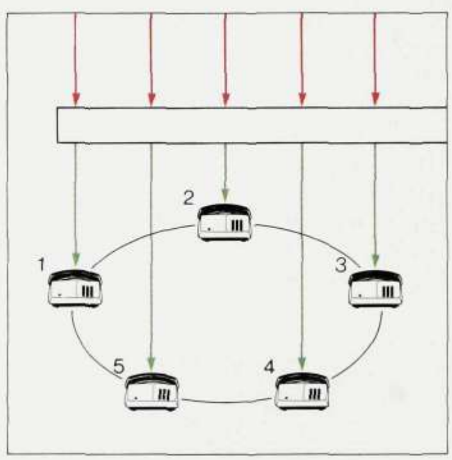


Fig. 7b
Operator-assisted system

Fig. 7c
Automatic distribution system



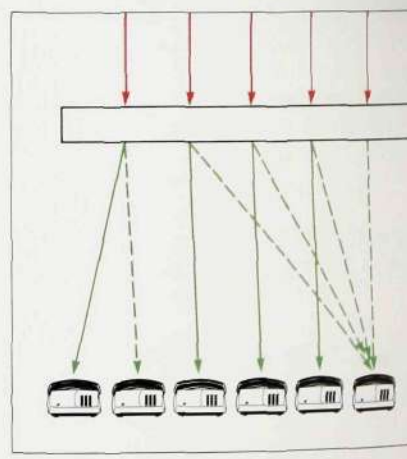


Fig. 7d
Group distribution system

Fig. 7e
Executive/secretary system

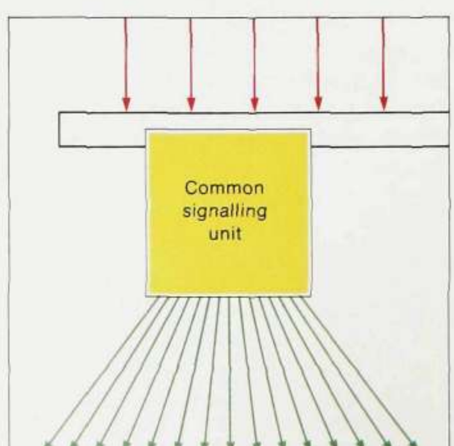


Fig. 7f
Common call distribution system

Fig. 7g
Direct communications system

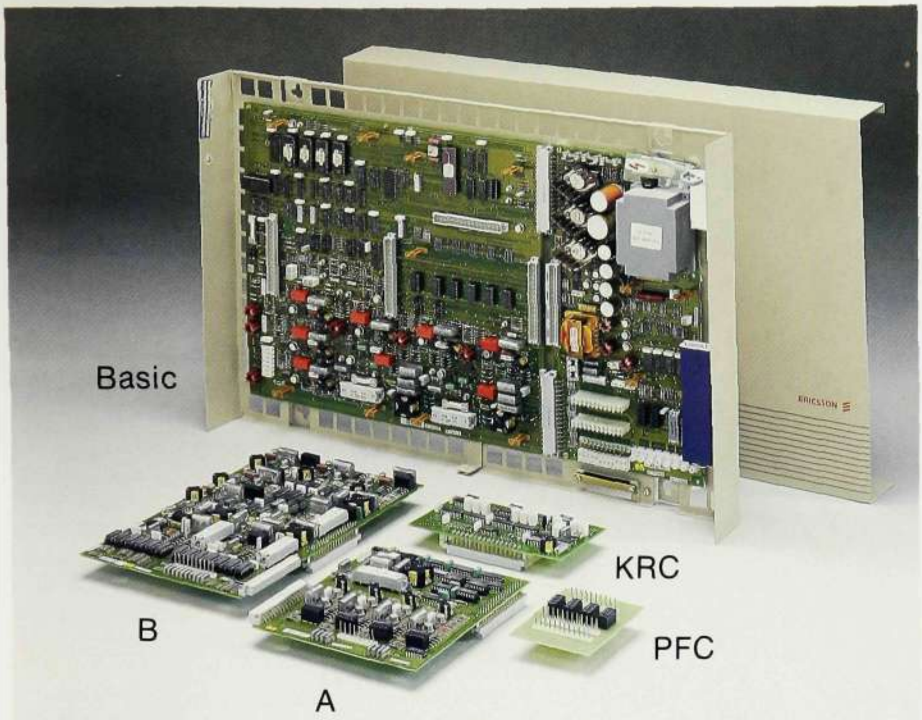


Fig. 9
The basic unit and expansion boards

Size	Number of extensions	Number of trunk lines	Basic unit and expansion boards
1	6	2	Basic
2	10	3	Basic + A
3	11	5	Basic + B
4	14	4	Basic + A + A
5	15	6	Basic + A + B

Fig. 8
Five hardware sizes

The group can be called both internally and externally. The extensions can also be called individually.

When a call comes in to the group and all extensions are busy, repeated camp-on connection attempts are made within the group. In such cases external calls are given priority over internal calls.

Executive-secretary system
Extensions can be given individual trunk lines regardless of whether the system is connected to a PABX or a public exchange. The line can be diverted to a secretary, who should be provided with a system telephone F1 or F2, fig. 7e.

The system includes many functions, such as call transfer, diversion, do not disturb, executive intrusion, recall, parking, automatic call-back and call pick-up, as well as an executive telephone (Executive 605) that can be used for executive-secretary functions even if it is not allocated special lines.

Common call distribution system
Calls that are presented on a common signalling unit can be answered from any extension regardless of type. The first person to key 8 receives the call, fig. 7f.

Direct communications system
Ericsson BCS 10 with loudspeaking telephones constitutes a direct communications system. It is then possible to receive an internal or extended call without lifting the handset or pressing any button, fig. 7g. This facility can be ordered and cancelled by the user.

System with alternative traffic characteristics
Four different traffic distributions can be preprogrammed, for example for daytime, lunch, evening and night. The desired traffic distribution is connected in by means of a simple command.

Hardware

The basic equipment for the central unit permits six extensions and two trunk lines. Five different sizes can be obtained with the aid of two extension boards (A and B), figs. 8–9. It takes only a few minutes to build out the system. A printed board assembly with a key tone receiver, the KRC board, is required if push-button (DTMF) telephones are to be used in addition to system telephone F1. In the basic version two trunk lines can be connected through to predetermined extensions in the case of a power failure. An extra printed board assembly, the PFC board, extends this facility to all trunk lines. The wall cabinet takes all equipment required for a fully equipped system.

Software

Ericsson BCS 10 is equipped with the full range of software right from the start. This means that exclusive functions with added commercial value, which could be offered as options, are provided without extra cost. This simplifies distribution and gives a complete package which increases the competitiveness of the system.

Innumerable possibilities
It has already been described how the system can be adapted to different traffic distributions with features for switching between different preprogrammed distributions, for example at different times of the day. Ericsson BCS 10 can simultaneously function as a multi-line system, operator-assisted or automatically distributed system, executive-secretary system, group distribution system, common call distribution system and direct communications system. The system is adapted to suit different needs by means of different types of telephones and by the way incoming and outgoing traffic for trunk lines and extensions is defined. For example, exten-

Fig. 10
The high service level of the system means rapid adaptation to new requirements through simple programming

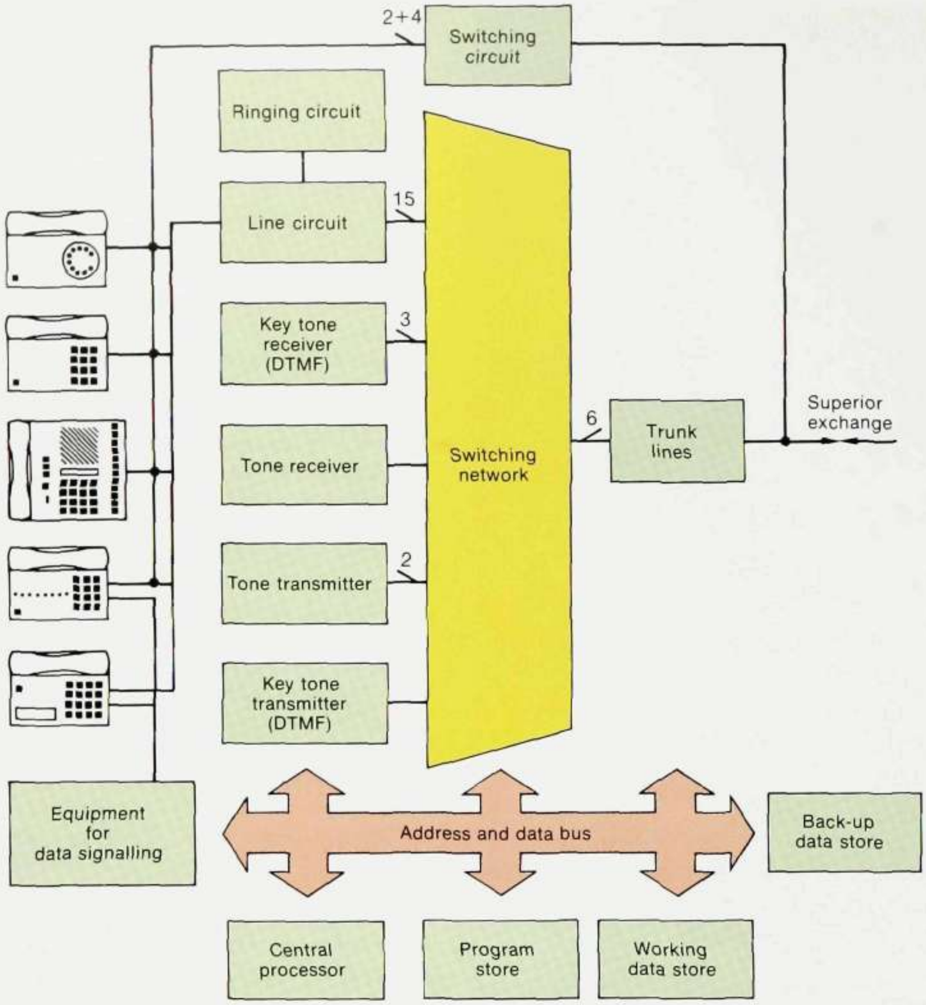


Fig. 13
Block diagram of Ericsson BCS 10

Fig. 11
Ericsson BCS 10 also enables each user to control his/her telephone traffic in a simple and efficient manner, by means of various services, such as direct diversion to optional extension; "do not disturb", parking, answering from optional extension, storing of the last number keyed and automatic call-back

Fig. 12
Today the most efficient and profitable meetings in many companies are telephone conferences. Loudspeaking telephones and features for multi-party conferences ensure the efficiency of such conferences

sions 1 and 2 may be allowed to make external calls, but not 3 and 4. In the evening all telephones except the manager's may be blocked for outgoing calls. Temporary changes can be arranged from the extension concerned, for example diversion to another telephone or an off-duty indication.

Adaptation to market requirements, for example as regards tone signals, and to the superior exchange, e.g. DTMF or decadic signalling, is made by activating different function variants in the software.

Some of the features offered by the system are also shown in the table "Features in Ericsson BCS 10".

Programming authorization
Authorization to modify and program data is arranged with the aid of two code words, one for the service staff and one for the system owner. There are different levels of authorization for the owner.

Supervision
Together with the hardware the program system supervises
- *different voltage levels*
- the operation of the processor
- the program handling
- the operation of the data store
- the contents of the data store
- the contents of the program store
- the links
- the trunk lines.

Many functions are performed automatically, such as the blocking of a faulty trunk line. The blocking is removed automatically when the fault has been cleared.

If a major fault is detected, the system initiates an alarm to system telephones F1 and F2. In F1 the cause of the alarm can be shown in response to a command. Service staff can easily locate faults with the aid of the software subsystem for operation and maintenance.

System structure

Ericsson BCS 10 consists of a switching system and a control system, which exchange information over a bus system, fig. 13.

The designers of Ericsson BCS 10 have combined the latest semiconductor technology with simple circuit designs throughout, from electronic components to software and mechanical construction, and thereby created a competitive business communications system while maintaining the traditional high component quality and operational reliability.

Switching system
The switching system comprises a switching network and line circuits for extensions, trunk line equipment and various devices. The switch matrices contain MOS transistor crosspoints. The switching network is used to connect up calls and also to connect in tone transmitters, dialling tone receivers and key tone transmitters and receivers, fig. 13.

The number of internal links is dimensioned to give a high traffic handling capacity. A fully equipped system has eight internal links and six trunk lines.

A call from an extension to the system is detected in the line circuit. The control system registers the call, selects a free internal link and connects in a key tone receiver and a dialling tone transmitter.

Features in Ericsson BCS 10	Standard push-button	Standard rotary dial	F1	F2	Executive 605
Abbreviated number					
central based (60)					
*common	o	o	o	o	o
*individual	o	o	o	o	o
*complementary	o	o	o	o	o
telephone based (9)					o
Alarm Indication			o	o	
Alternation on inquiry	o	o	o	o	o
Automatic call-back					
extension	o	o	o	o	o
trunk lines	o	o	o	o	o
Call extending	o	o	o	o	o
Call information					
incoming unanswered calls			o	o	
line busy			o	o	
line parked for common access			o	o	
line free			o	o	
Call pick-up	o	o	o	o	o
Call waiting indication	o	o	o	o	o
Camp-on	o	o	o	o	o
Conference	o	o	o	o	o
Date and time					o
Day/night service switching[1]					
four alternatives	o		o	o	o
Day/night switch indication			o	o	
Direct-in lines	o	o	o	o	o
Distinctive ringing	o	o	o	o	o
Diversion					
common	o	o	o	o	o
individual	o		o	o	o
on no reply	o		o	o	o
direct ("follow me")	o		o	o	o
Do not disturb	o		o	o	o
Executive intrusion	o	o	o	o	o
Extension classes of service	o	o	o	o	o
External call					
automatic or manual	o	o	o	o	o
Group hunting	o	o	o	o	o
Hands-free answering					o
Internal automatic calls	o	o	o	o	o
Inquiry call to:					
another extension	o	o	o	o	o
public network subscriber	o	o	o	o	o
extension in superior PABX	o		o	o	o
Last number redial					o
Last number stored	o	o	o	o	o
Loudspeaking function					o
Multi-line access			o	o	
Number displayed					o
Off-duty indication	o		o	o	o
Parking					
short term, with recall	o	o	o	o	o
long term			o	o	
Power failure circuits (PFC)					
for all external lines					
standard (2)	o	o		o	o
optional (4)	o	o		o	o
Programming of system data			o		
Protection against intrusion	o	o	o	o	o
Recall	o	o	o	o	o
Reminder signal					o
Single-button access with labelling (36)					o
System data indication			o		
Traffic indication			o	o	
Transfer	o	o	o	o	o
Trunk call discrimination (TCD)	o	o	o	o	o
Two-wire-connection	o	o			o
Data line security	general system features				
Flexible numbering	general system features				
Line lock out	general system features				
Music on hold	general system features				

1) The first extension position and F1

The extension starts the dialling. The digits are analyzed by the control system. If the number is an internal one the key tone receiver is released, ringing tone is connected to the B-extension and ringing control tone to the A-extension. When B answers, both the ringing and ringing control tones are disconnected and the speech connection is completed.

If the control system detects that the extension has dialled the digit for an external line a free trunk line is selected. The superior exchange is called and the dialling tone receiver scans the line in order to detect the dialling tone. When this tone is received the external number can be dialled.

The system permits trunk call discrimination. The first digits in the dialled number are checked against a programmed digit table, prepared in accordance with the requirements of the customer.

Control system
The control system includes an 8-bit microprocessor, a program store, a working data store and a back-up data store, fig. 13. All programs are stored in static RAMs. The data stores are PROMs. The back-up store is protected and preserves the data in the case of a power failure (EEPROMs). The hardware generates all signals required by the processor when starting up initially and after a power failure.

Software
The software is divided into a switching subsystem and an operation and maintenance subsystem. Fig. 14 shows the software structure in more detail.

The program block for man-machine communication is of particular interest. It has been designed in such a way that the user can easily make the system work in accordance with his own requirements. This is done by means of commands via the system telephone F1. The central unit and F1 each contain an 8-bit microprocessor.

The two system telephones, F1 and F2, receive information about the state of the trunk lines from special data signalling equipment, fig. 13.

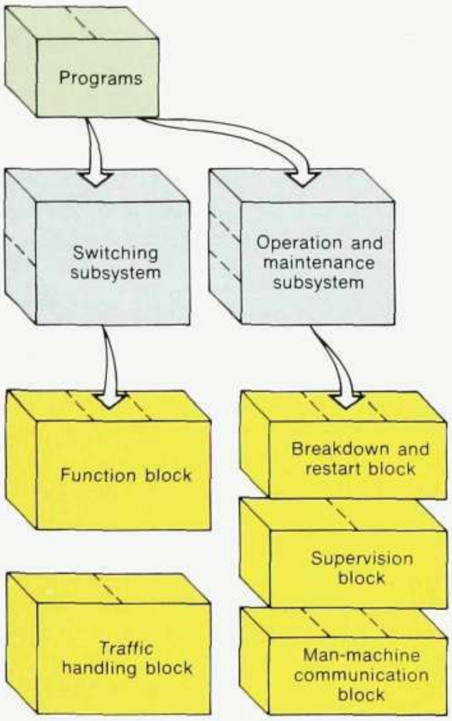


Fig. 14
Software structure

System security
The risk of loss of system data is exceedingly small since the data is stored in a permanently protected back-up store (EEPROM). Automatic trunk line switching for all trunks to predetermined extensions can be provided, and also a comprehensive battery back-up system. The system is equipped with overvoltage protection for trunk lines and critical (example: outdoors) extensions.

Mechanical construction
The cabinet is made up of two units, a wall plate with folded edges and a cover. The basic printed board assembly and the power unit board are mounted on the wall plate. Additional printed board assemblies are mounted on these boards. The boards are connected to fixed connectors and fastened with spacers, fig. 9.

Installation
The wall plate is screwed to the wall. A space is formed between the plate and the wall to ensure good heat dissipation. Extensions and trunk lines are connected direct to the printed board assemblies by means of MOLEX connectors, which cut through the insulation so that the wires do not have to be stripped or fastened by screws. The power lead is plugged in, and the Ericsson BCS 10 system is in opertion.

Summary

Ericsson BCS 10 is a complete business communications system for up to 15 extensions, with a wall-mounted central unit and a number of different types of telephone sets. The software is powerful and provides the user with sophisticated features. The system can easily be adapted to different requirements as regards telephone sets, management of the telephone traffic and control of call costs, and the operation is simple. The system is very suitable for mass production and rational distribution.

Referenser

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