ERICSSON REVIEW

MD 110–A DIGITAL SPC PABX
DIGITAL SIGNAL PROCESSING IN SYSTEM MD 110
NORDIC PUBLIC DATA NETWORK WITH AXB 30
CONTROL AND SUPERVISION SYSTEM FOR RAILWAY TRAFFIC
ERICSSON SUNWIND
AXE 10 IN THE STOCKHOLM TELECOMMUNICATION AREA

1 1982

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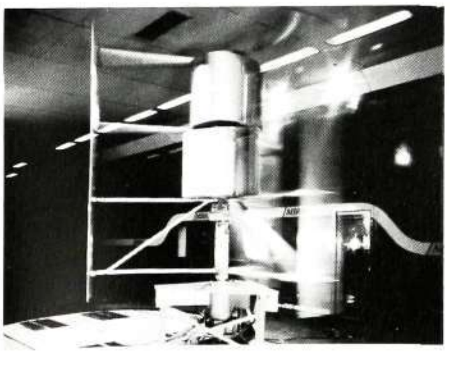

COVER
The wind generator in the ERICSSON SUNWIND power supply system undergoing testing in a wind tunnel

MD 110—a Digital SPC PABX

Rolf Mörlinger

MD 110 is a digital PABX for 100–10000 extensions designed for use in analog and digital environments. The PABX is characterized by modularity and flexibility, permitting geographical dispersion of its units. This has been achieved by using standard 32 channel PCM techniques and distributed stored program control. MD 110 provides the facilities associated with a large, modern PABX and it also permits integrated voice and data communications. It is being developed by ELLEMTEL for Ericsson and the Swedish Telecommunications Administration.

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A distinctive feature of the digital SPC PABX MD 110 is that only two superior types of unit are included, namely line interface modules and group switches.

The Line Interface Module, LIM, is a microprocessor-controlled unit that can be equipped with any combination of line circuits, trunk circuits and other telephony devices. The LIM possesses an internal digital switch and can function as an autonomous PABX or as an integrated part of a larger system. The capacity of a LIM in normal traffic conditions is 200 extensions approximately. Larger PABXs are achieved by interconnecting LIMs via 32 channel PCM links for traffic and control purposes. Up to three LIMs can be interconnected directly, whereas larger systems need the group switch.

The Group Switch, GS, is a modularly expandable digital switch, whose task is to transmit PCM-voice, data and control signals between LIMs. GS itself possesses no control equipment; it is controlled fully by the connected LIMs.

It is a simple matter to adapt MD 110 to a company's geographical structure as the exchange LIMs can be placed together or assigned to nearby or remotely situated company units. 2 Mbit/s line systems are used when the distance between LIM and GS exceeds 500 metres. A physically dispersed but functionally coordinated MD 110 is an attractive alternative compared to a traditional network composed of a number of separate exchanges.

MD 110 offers extensions and operators numerous facilities. The extensions can be furnished with conventional telephones, for rotary dial or DTMF keypad, or with digital system telephones. Digital system telephones and also digital operator consoles are connected to the exchange via normal two-wire lines. The majority of extension facilities are accessible from rotary dial telephones but a DTMF telephone or digital system telephone is required if all the system's facilities are to be utilized.

Fig. 1
The operators keyboard and visual display unit are separate units in MD 110

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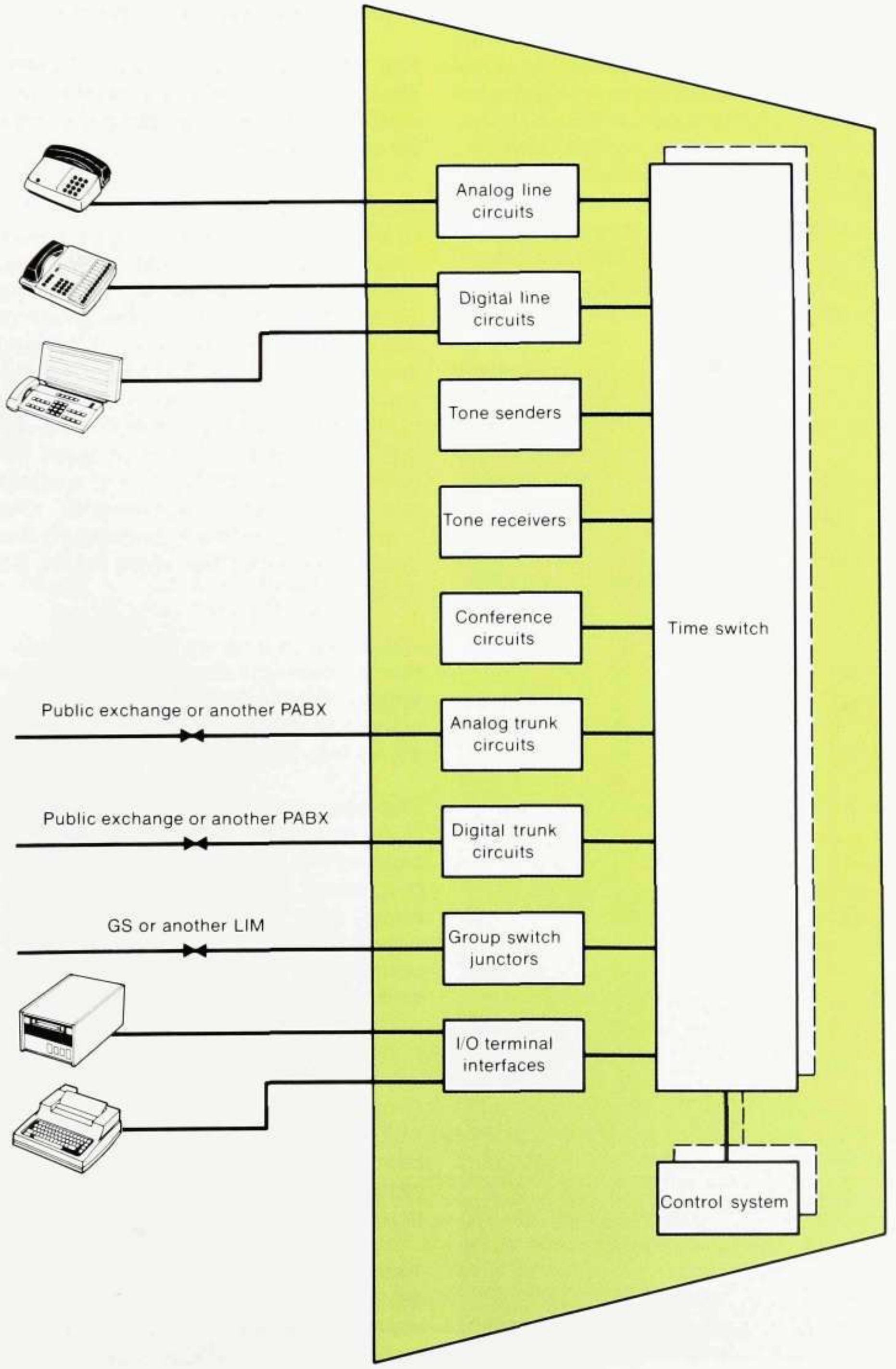
The digital system telephone:
- simplifies use of the system's facilities
- replaces separate systems e.g. intercom systems and call diversion/interception systems
- serves as connection point for data terminals.

Fig. 2
Block diagram for line interface module, LIM



The function and software modularity that is characteristic for Ericsson's AXE 10 system is to be found in MD 110 also. The programming language, PLEX, and the support system, APS, that is used in the design and production of software, are also common for MD 110 and AXE 10.

Like earlier SPC PABXs from Ericsson, ASB 100 and ASB 900, MD 110 is built up in cabinet packaging structure BYB 201. The system makes considerable use of microprocessors and LSI circuits. All integrated circuits have ceramic incapsulation in order to obtain the best possible operating characteristics.

Development of MD 110 progresses in stages. The first version of the system contains the majority of the voice communication facilities described in this article. Continued development means inter alia that the system will be equipped with all functions named here for voice communications and with functions for data communications. Special system variants, e.g. a transit exchange, are also included in the development work.

Line Interface Module, LIM

The LIM is the unit in MD 110 to which extensions, operators and trunk lines are connected, fig. 2. One LIM has capacity for 200 extensions approximately and can function as an autonomous exchange or as an integrated part in a larger system. The basic equipment, that can be duplicated, is formed by the control system and time switch. In addition, the LIM can be equipped with an arbitrary mixture of analog and digital line circuits and trunk circuits as well as devices for tone sending, tone reception, multiparty conference, group switch connection and I/O terminal connection.

A complete LIM is housed in one cabinet with the dimensions 1800x600x300 mm, fig. 3. The printed board assemblies (boards) are placed in two magazines, each with three shelves. The wiring between the boards in a magazine is achieved largely via the latter's back plane. Intermagazine wiring and wiring to the MDF always utilizes plug-ended

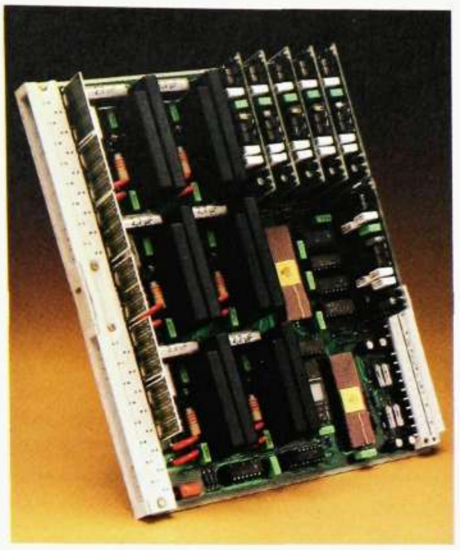
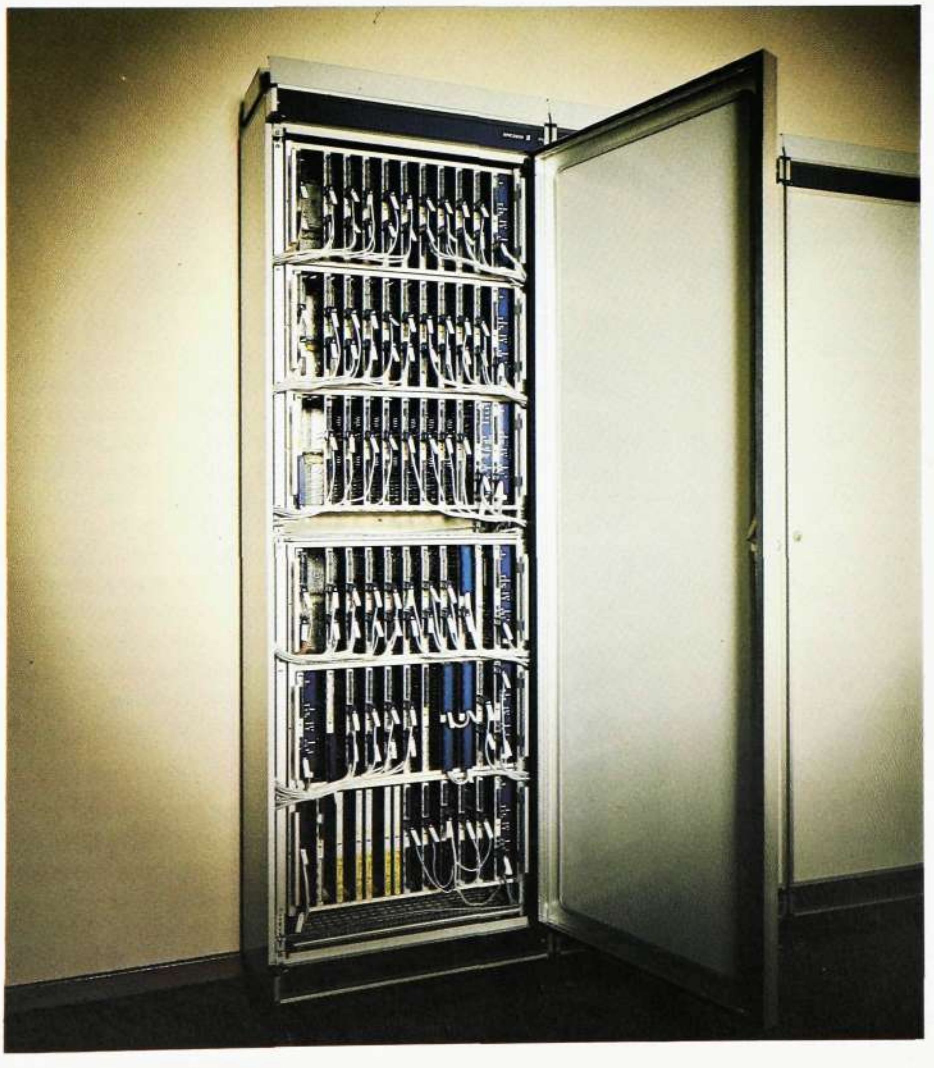


Fig. 4
Printed board assembly with six analog line circuits

Fig. 3
A LIM is housed in a cabinet with the dimensions 1800×600×300 mm



cables that are connected to the fronts of the relevant boards. In some cases front connection is also used to interconnect boards in the same magazine.

Control system

The control system consists of one processor board, LPU, and a number of memory boards, MEU. LPU contains a processor system built up around two commercial 8-bit microprocessors. One processor functions as the LIM's main processor and the other works as signal processor whose task is to handle the direct communication with the control circuits of the switch and the telephony devices.

The memory boards, in whose circuits the LIM programs and data are stored, contain RAM type storage components. Each board contains 256 kbytes.

Time switch

The switch is non-blocking, has 512 time slots (multiple positions) and consists of one basic board, BSU, and two supplementary boards, SSU.

BSU contains the voice and control memories for the time switch as well as a microprocessor, that controls the internal functions of the switch and has contact with the signal processor.

The SSU boards each serve 256 time slots and undertake serial/parallel conversion of the PCM signals to and from the device boards.

Telephony devices

Analog line circuits are used for connection of conventional telephones using decadic or DTMF signalling and analog trunk circuits for connection of traditional type exchange lines and lines to other PABXs. The analog/digital conversion takes place on the boards using single channel codecs. The design of the line circuits and trunk circuits is influenced by signal systems and other market requirements. One board, ELU-A, normally contains six line circuits, fig. 4 and one board TLU-A, 2-3 trunk circuits.

Digital line circuits are used for connection of operator consoles and digital system telephones via normal two wire lines. One board, ELU-D, contains eight digital line circuits.

The group switch junctor and the digital trunk circuit are both terminals for 32 channel PCM links and each occupies one board. The group switch junctor board, GJU, is designed to facilitate connection to two parallel group switches. A LIM is normally equipped with two GJU boards.

A digital trunk circuit, TLU-D, corresponds to 30 conventional analog trunk circuits and it is used to connect MD 110 to other digital exchanges. Digital connection of MD 110 to the AXE 10 group switch will be the first application of this technique.

Tone senders and tone receivers are devices for digital generation and reception respectively of tones (dial tone, busy tone, etc.) and also dual tones for

DTMF *signalling*. A *tone sender unit*, TSU, consists of one board and supplies one LIM with all tones. A tone receiver unit, TRU, consists of two boards and contains eight DTMF receivers and four receivers for dial tone. One TRU is normally adequate for one LIM.

A multiparty conference unit, MPU, occupies one board and is used, by digital methods, to achieve a flexible number of conferences with 3-8 participants. One MPU is normally included in each LIM.

An I/O terminal interface unit, IOU, consists of one board and facilitates connection of cartridge tape units for backing up programs and data, and also three I/O terminals for operation and maintenance purposes. One or several IOUs are included in one MD 110.

Flexible device configurations

Each device board possesses a microprocessor, device processor, that controls the detection and manoeuvre functions on the board, and manages communications with the signal processor. The use of device processors has facilitated the introduction of a standardized interface towards the wiring in the back plane of the magazines.

The distribution of the switch time slots to the device boards in a LIM is shown schematically in fig. 5. The majority of a LIM's approximately 50 device board positions can be used for all board types requiring up to eight time slots, i.e. primarily line circuits and trunk circuits. The distribution of the time slots is also arranged so that boards with requirements up to 32 time slots can be placed on every fourth board position provided that the following three re-

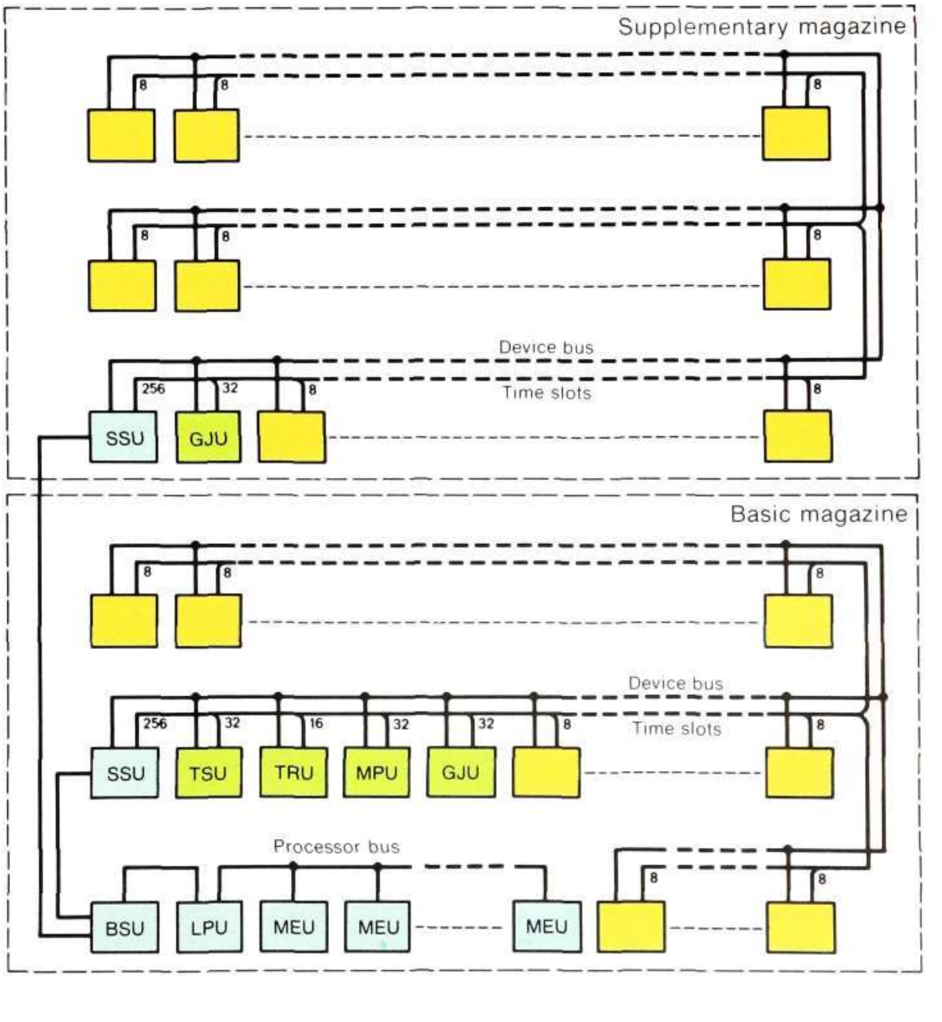
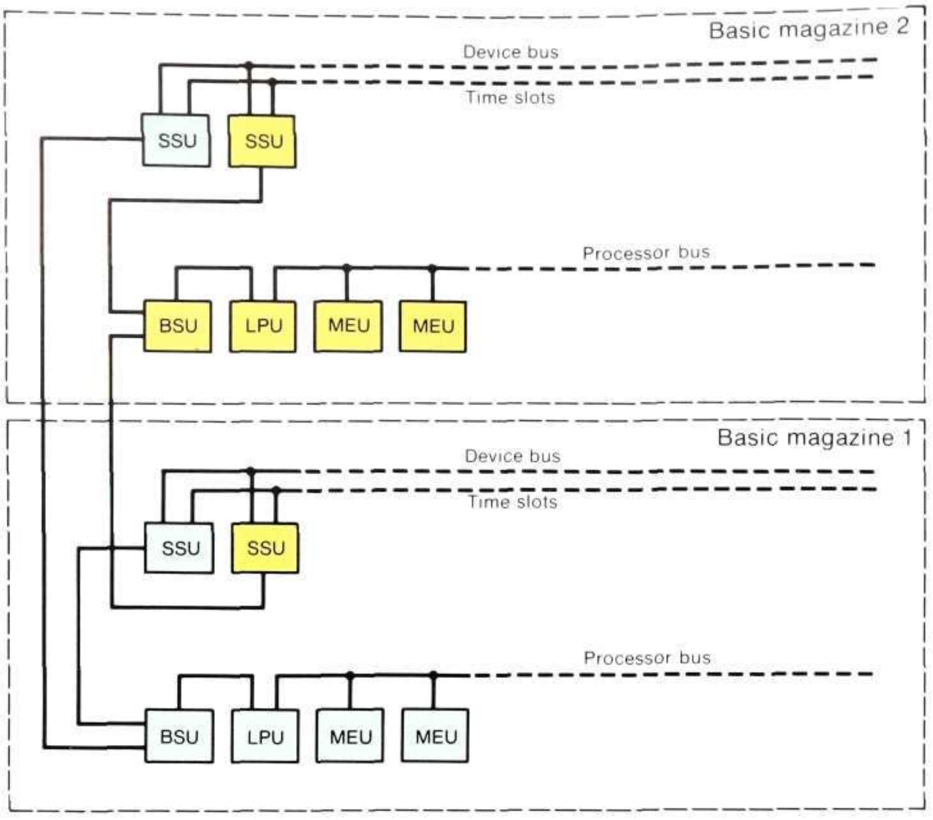


Fig. 5
Schematic structure of LIM showing how the switch time slots are distributed to the device boards

LPU	LIM Processor Unit
MEU	Memory Unit
BSU	Basic Switch Unit
SSU	Supplementary Switch Unit
TSU	Tone Sender Unit
TRU	Tone Receiver Unit
MPU	MultiParty conference Unit
GJU	Group switch Junctor Unit
	Arbitrary device board with maximum 8 individuals (primarily line circuits and trunk circuits)

Fig. 6
LIM with duplicated control system and duplicated switch. The duplication is based on the active/passive principle



main unoccupied. In the corresponding manner it is possible to place boards requiring up to 16 time slots on every second position. In order to avoid empty board positions adjacent to the two GJU boards and to the board TRU, TSU and MPU, that are virtually always to be found in a LIM, these units have permanent board positions, i.e. where the requisite 16 or 32 time slots are available.

Redundancy as required

The probability of a serious fault in a LIM, can be minimized when required, by duplicating the control system, the time switch and other units that normally exist only singly. Duplication is achieved by interconnecting two basic magazines in the manner shown in fig. 6. An active/passive configuration is used with automatic switching in the

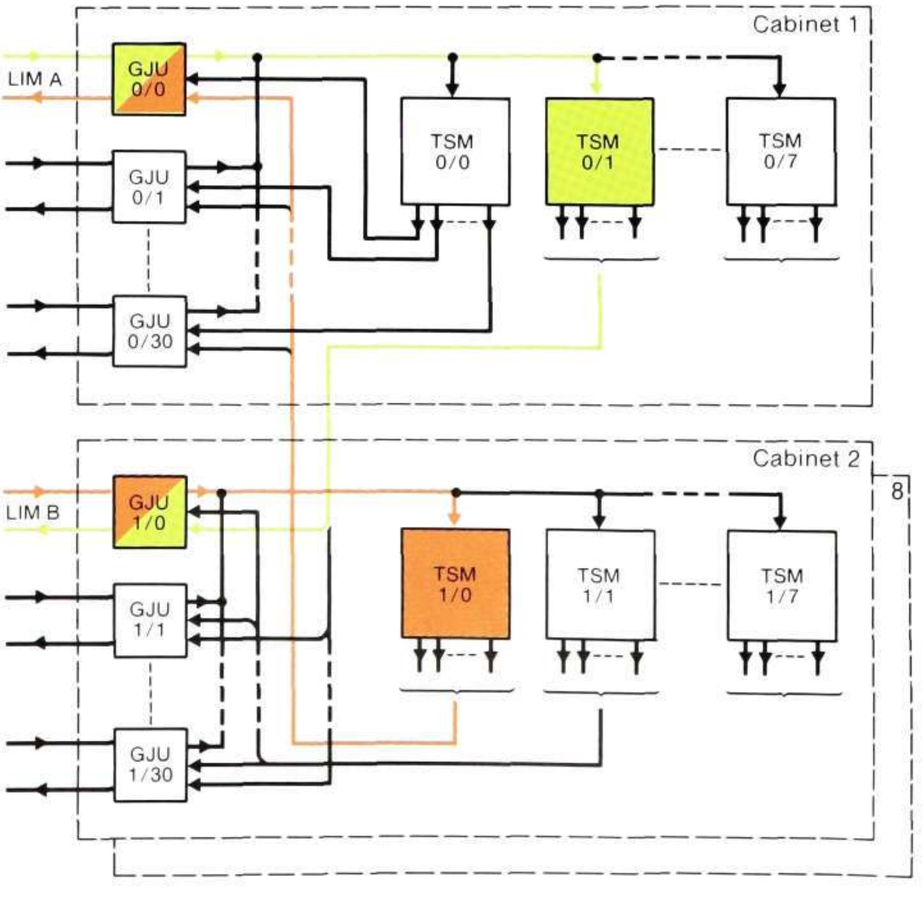


Fig. 7
Block diagram for group switch, GS

GJU Group switch Junctor Unit
TSM Tone Switch Module

event of a fault. The active system controls and connects all devices via its set of switch boards. The quantity of device board positions is reduced to about 40 in a LIM of this character.

Group Switch, GS

The task of the group switch is to connect PCM-voice, data and control signals between the LIMs in an MD 110 exchange. The LIMs are connected to GS via 32 channel PCM links. The time slots T1-T15 and T17-T31, are used for speech and data, T16 for control signals and T0 for synchronization signals.

GS is a non-blocking switch consisting of one or a number of time switch modules with 1024 ports. The modules are arranged in a matrix, fig. 7. The PCM links from the LIMs are connected via terminal boards, GJU. One time switch module has capacity for 31 PCM links, 2×2 modules for 62 PCM links and so on. GS can be expanded to maximum 8×8 modules and can then manage 248 PCM links.

GS is controlled from the connected LIMs. GJU contains a microprocessor, that manages the communications with the processor of the connected LIM via time slot T16. The processors on the GJU boards extend control information internally within the group switch.

The establishment of a connection between two LIMs commences with the LIM processors informing one another about which time slots have been selected on the relevant PCM link. The GJU processors are then ordered to establish the connections via the involved time switch modules. A connection through GS is to its nature oneway and it is consequently necessary, as shown in fig. 7, to establish two paths through the switch in order to obtain a bothway connection. The path from LIM A to LIM B proceeds via switch module TSM 0/1 whereas the path from LIM B to LIM A proceeds via switch module TSM 1/0.

The GS equipment is placed in the same type of cabinet as the LIMs. One cabinet houses the eight switch modules that form a row in the switch matrix, and the corresponding 31 GJU boards. A fully built up GS comprises eight cabinets.

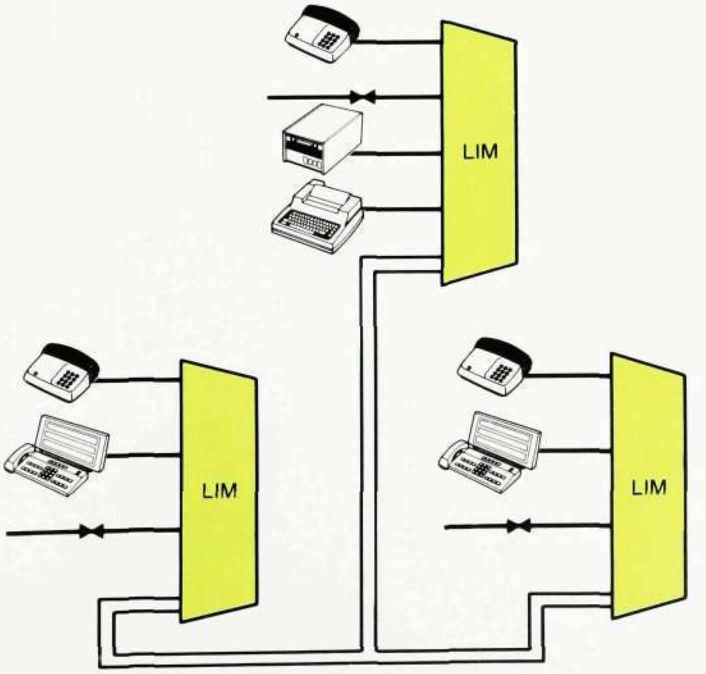
System structure and system function

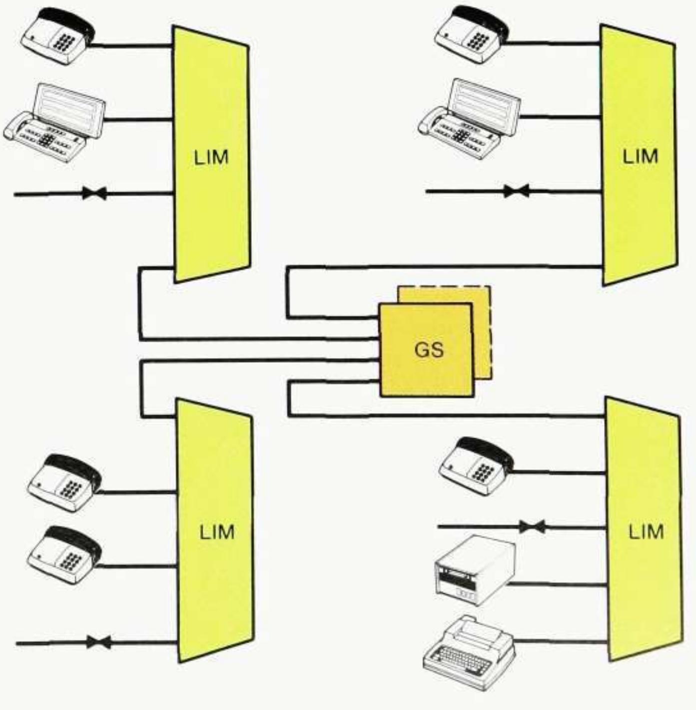
System with and without GS

At its smallest, MD 110 consists of one LIM, that has capacity for about 200 extensions. Larger systems are achieved by interconnecting several LIMs via 32 channel PCM links. Up to three LIMs can be directly interconnected, fig. 8, but thereafter GS is used. Normally, each LIM is connected with

Fig. 8, left
Up to three LIMs can be interconnected directly

Fig. 9, right
Four or more LIMs are interconnected via group switch, GS





two PCM links to GS, the latter duplicated for reliability reasons, fig. 9.

A PCM link normally consists of two coaxial cables only between the terminal boards, GJU, in LIM and GS. Levels, synchronization and coding are in accordance with CCITT recommendations for 32 channel PCM equipment. The signalling on time slot T16 takes place in accordance with CCITT's signalling system No. 7, levels 1 and 2.

Functional system structure
In addition to the physical structure already described, system MD 110 also possesses a functional structure to which the system documentation is associated. The system is divided into an Audio Communication System, ACS, and a Service System, SES. In their turn, these are divided into a number of subsystems that contain characteristic main functions, fig. 10.

Each subsystem contains a number of function blocks. The function blocks constitute natural design objects. They consist of program units or printed board assemblies or both. Analog and digital extension lines, operator lines and various types of trunk lines are thus each represented by their own function block, that comprises the adaptation boards corresponding to the line type as well as a number of program units. Number analysis and abbreviated dialling are examples of function blocks that only contain program units.

Software structure
The program units are independent products with strictly defined interfaces. Each program unit consists of

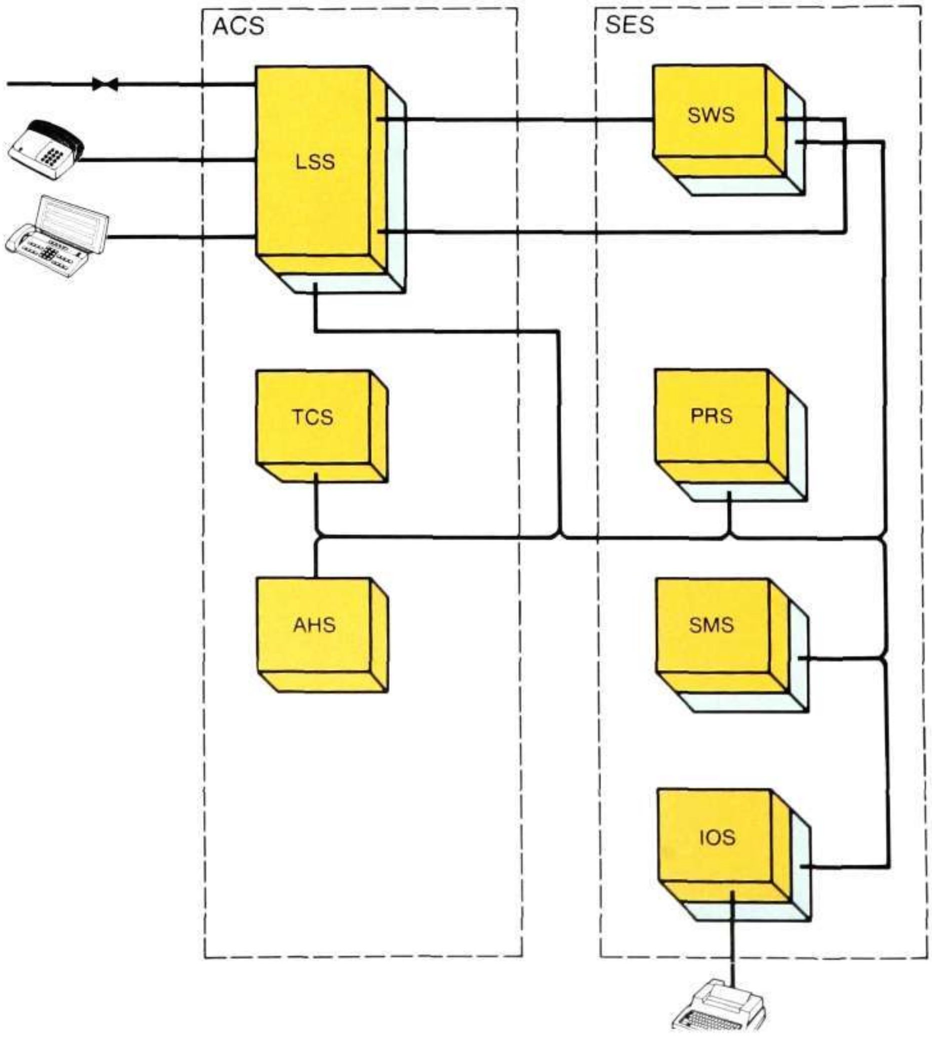


Fig. 10
Functional system structure

ACS	Audio Communication System
LSS	Line Signalling Subsystem
TCS	Traffic Control Subsystem
AHS	Audio Handling Subsystem
SES	Service System
SWS	Switching Subsystem
PRS	Processor Subsystem
SMS	Service Maintenance Subsystem
IOS	Input Output Subsystem
	Hardware
	Software

programs and the corresponding data. The latter can only be reached by the program unit's own programs. All interworking between program units, within and between LIMs, is achieved with formal messages, so called "program signals". The signal channels of the PCM links are used to send program signals between program units in different LIMs.

The majority of the program units are written in the high level language PLEX, originally developed for Ericsson's AXE 10 system. Program units with operative functions are written in assembler language.

MD 110 and AXE 10 also have the support system APS 210 in common. It is used in conjunction with the design and production of software.

Control principles

The main principle for the control functions in MD 110 is that the LIMs shall be as autonomous as possible, while, concurrently, they shall interwork in a manner whereby externally the system functions as one fully co-ordinated exchange. The latter means that the LIM must not constitute any limitation for extension numbering, facilities, operation and maintenance functions, accessibility of trunks and other telephony devices, etc. The desired control characteristics have been achieved by means of a well conceived distribution of the software functions in the distributed control system, that the LIM processors form together. The possibility of rapid, safe signal transmission provided by the signal channels of the PCM links is a basic prerequisite for the interworking between the LIM processors.

In effect, MD 110 software has been designed using the following principles:

1. It shall be possible to use the same software in systems with one LIM and in multi-LIM systems.
2. To as great an extent as possible the processor load in a LIM shall be independent of the total number of LIMs in the system.
3. The inter-LIM signalling requirement shall be as small as possible.
4. Any LIM that is isolated from the remainder of the system shall be capable of functioning as a separate exchange.
5. The quantity of duplicated programs and data shall be minimized, but in a manner that does not disregard the above stated principles.

To exemplify how the general principles have been applied, descriptions follow of how some familiar functions are executed in the system.

For obvious reasons each LIM is equipped with its own set of the program units that manage signalling functions, connection-supervisory functions, etc. for the connected telephony devices.

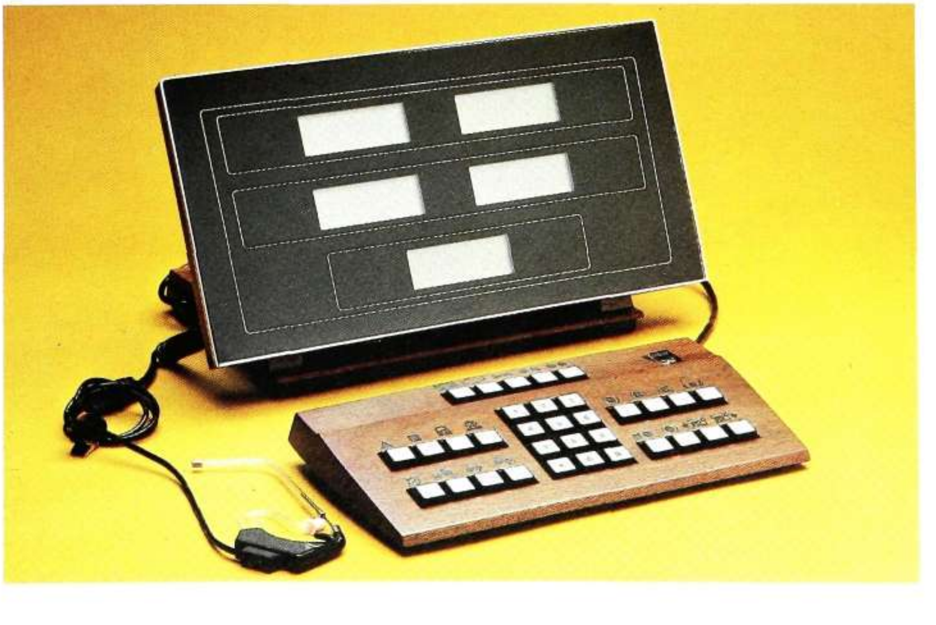


Fig. 11
Operator console for MD 110

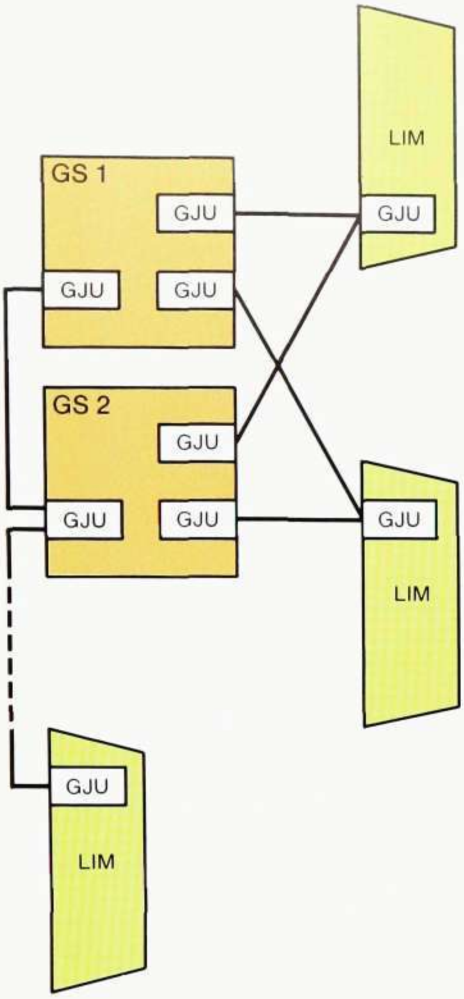
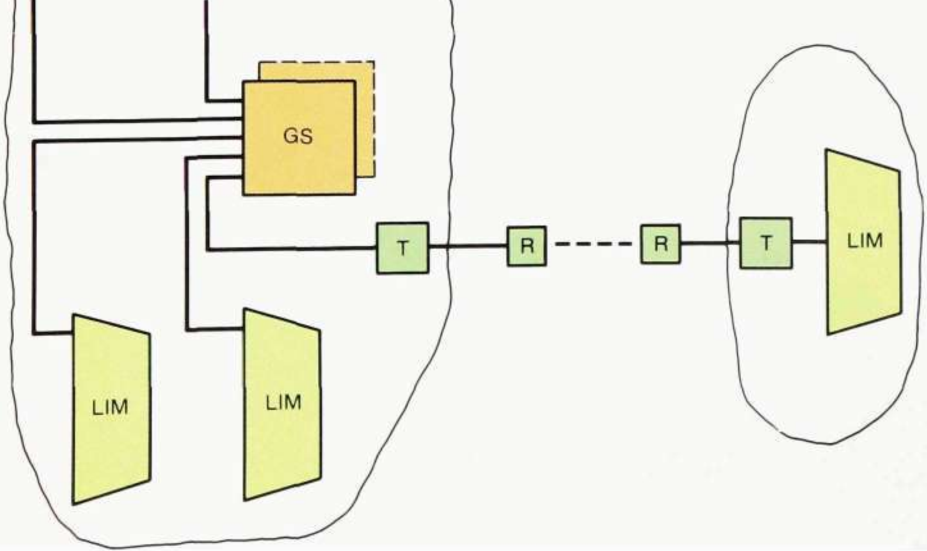


Fig. 13
Connection of local and remote LIMs to duplicated GS

Fig. 12
Remote LIMs are connected to GS by 2 Mbit/s line systems

T Terminal equipment
R Repeater



Number analysis is a function to which all LIMs require frequent access. The data volume is moderate and independent of the system size. This function is placed in each LIM in accordance with the principles 2–4 and without serious disregard to principle 5.

Translation of directory numbers into multiple numbers and class of service codes is a function whose data increase in step with the amount of extensions. A complete function in each LIM would result in very great data volumes in large exchanges. The function has therefore been divided into a central part and a regional part. The regional part exists in each LIM and contains complete data for extensions connected to the LIM. For reliability reasons the central part is contained in two predetermined LIMs. The central part contains only those data that are needed, by directory number, to identify the LIM in which complete data can be obtained. The data volume of the central part now grows at a reasonable rate with the exchange while, concurrently, the regional parts provide the LIMs with the desired autonomy. The central part of the described function is one example of a so called "common function". Other functions that to a greater or lesser degree have been realized as common functions are selection of outgoing lines, queue to operators, loading of storage units and man-machine communication.

Remote LIMs
One of the most distinctive characteristics possessed by MD 110 is the possibility to connect geographically remote LIMs to GS. LIM and GS are interconnected solely with coaxial cables as long as the distance is less than 500 metres. LIMs can be placed at any distance from GS with the aid of 2 Mbit/s line systems, in which are included line terminals and repeaters, fig. 12.

Short range, distributed (remote) LIMs can be used to spread an extension network throughout a number of buildings in a rational manner. Long-range, distributed LIMs can be used in private networks as an attractive alternative to independent satellite exchanges. In such a network the extensions can retain their directory numbers no matter where they move. All facilities function throughout the network and the operators can be situated centrally or dispersed.

A remote LIM can be equipped with line circuits only and then function as a concentrator with autonomous internal traffic. If a higher degree of autonomy is required it is also possible to connect public trunk lines and operators. The LIM will then function as an autonomous exchange if connection with the remainder of the system is cut off for any reason.

Reliability
The modular structure of MD 110 gives the system good reliability characteristics. Operators and lines in incoming and outgoing routes can be distributed among several LIMs.

The functions that are not decentralized, i.e. the common functions and the group switch, GS, can be duplicated.

A duplicated GS consists of two identical units, that work independently of one another, fig. 13. Parallel paths are established via the two GS units and identical information is sent via these paths. The PCM links are normally split on the GJU boards in the LIMs but, as seen in fig. 13, this can also be done on the GS-side. The latter method, that is slightly inferior from the reliability aspect, can be used to decrease the connection costs for remote LIMs.

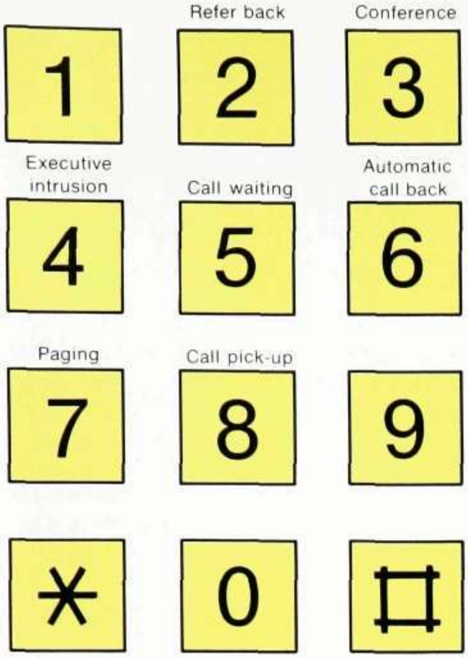
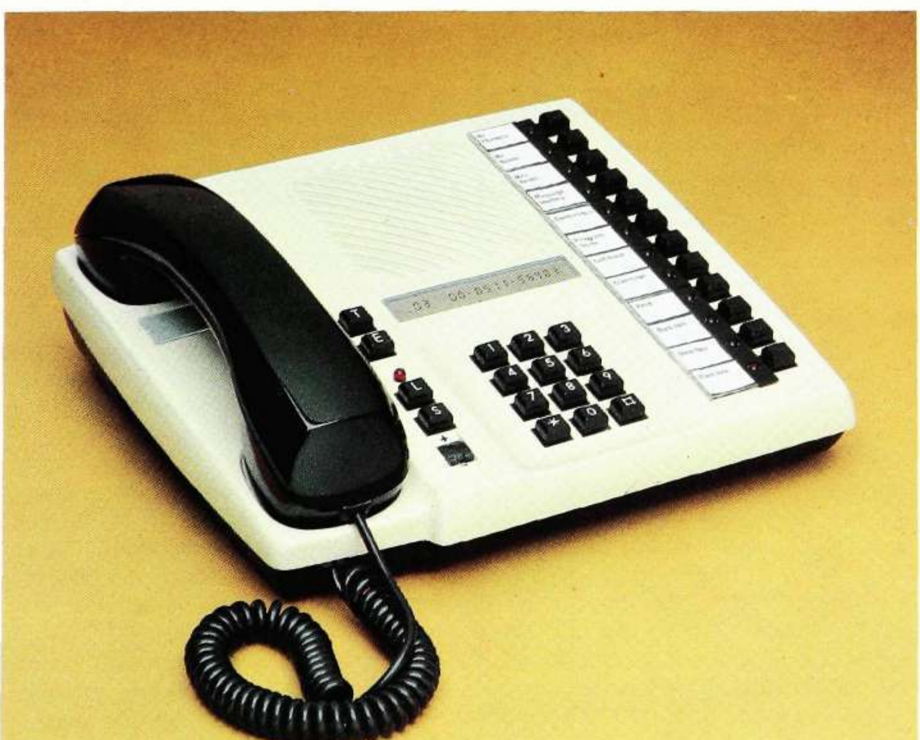


Fig. 14
Single digit codes are used to activate facilities in the inquiry, busy and ring states

Fig. 15
Digital system telephone. A larger version with 36 programmable function buttons is also available



All or certain selected LIMs can be provided with increased reliability by duplication of the control system and time switch.

Each LIM is normally equipped with two group switch junctors but with only one set of tone senders, tone receivers and conference devices. The latter can also be duplicated, but as an alternative devices in other LIMs may be used.

Capacity

MD 110 has a capacity for 10 000 extensions. This quantity is determined by limitations that have been introduced into the software for practical reasons. The hardware structure permits a much greater quantity.

The modular structure and flexible equipment configurations of the system permit extensive variations in the traffic capacity. The LIM's two internal switches and the group switch are nonblocking. Consequently, the traffic capacity is determined only by the number of trunk lines and the number of PCM links between LIM and GS. Up to 90 lines can be connected to each LIM and the LIMs can be connected to GS with up to four 32 channel PCM links.

Facilities

Extensions and operators have access to numerous facilities that make telephoning more effective and simpler, both for internal and external calls. Certain advanced extension facilities are associated with the digital system telephones, but most of the facilities are accessible from conventional telephones.

System-oriented facilities

In addition to the purely extension and operator oriented facilities there exist numerous system-oriented facilities:
- automatic and operator expedited internal, outgoing and incoming traffic
- flexible numbering
- extension classes of service
- trunk call discrimination
- extension group hunting
- group call pick-up
- night service (route/line-associated)
  - universal (call on common signal device)
  - general (call to common answer position)
- interworking with other private exchanges via private trunk lines (tie lines)
- alternative routing on outgoing calls
- call metering
  - counting of meter pulses
  - call information logging
- direct connection to public trunk lines on operational breakdown.

Facilities for extensions equipped with conventional telephones

The facilities listed below are all available from DTMF telephones. The great majority of the facilities are also accessible from rotary dial telephones:
- inquiry
- refer back during inquiry
- call transfer
- multiparty conference
- automatic call back
  - busy extension
  - on no reply
- call waiting indication
- executive intrusion
- paging
- call diversion
  - direct
  - on busy
  - on no reply
  - follow me
  - bypass

- abbreviated dialling
  - common
  - individual
- last number redial
- non-dialled connection to predetermined answer position
- call pick-up
- *night service, individual.*

MD 110 offers simple handling of those facilities that are activated in the inquiry, busy or ring states. The register function is always accessible on these occasions and single digit codes can be used then, without any consideration to how the normal number series has been chosen. Fig. 14 shows how a DTMF telephone can be equipped with auxiliary designations for the facilities in question. The single digit codes can also be used from rotary dial telephones.

Facilities available to extensions equipped with the digital system telephone

The digital system telephone, that forms part of the MD 110 system, allows simpler utilization of the facilities listed earlier. It can also be used for a number of special purposes.

The digital system telephone exists in two sizes and contains pushbutton unit, loudspeaker, handset, liquid crystal display and 12 or 36 programmable function buttons with associated light emitting diodes, fig. 15. The telephone is connected to the PABX by a normal two wire line via which it is also fed current.

Three of the function buttons have permanent functions, that simplify handling of the extension user's own line in conjunction with parking, refer back, inquiry etc. The functions of the remaining buttons are programmable. For example, they can be used to simplify more facilities according to the principle one button per facility or for single button access. The latter function in combination with inter alia the loudspeaking function makes it possible, using digital system telephones, to replace a separate intercom system. The programmable function buttons can also be programmed so that the telephone becomes an advanced aid for answering calls to extension groups (line pickup group function) and for diversion of calls to secretaries.

The digital system telephone also serves as connection point for data terminals. MD 110 permits concurrent and mutually independent voice and data communications on the digital extension lines and onwards through the system, fig. 16. The data communication facilities, the digital system telephones and their use will be described in detail in separate articles in a later number of Ericsson Review.

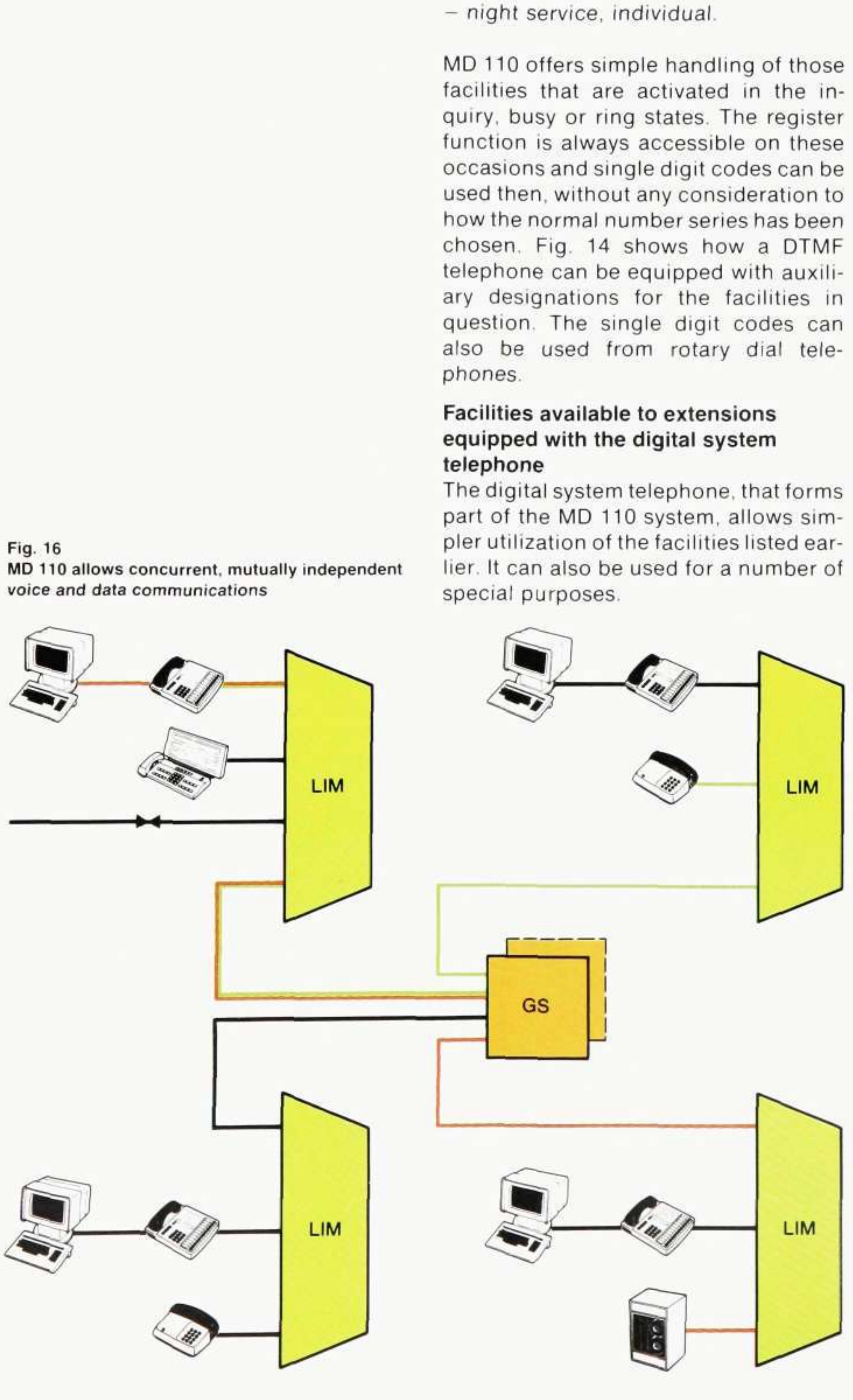
Operator facilities

MD 110 possesses a compact operator console, consisting of separate keyboard and display units, fig. 11. The console is connected to the PABX by two wires only in the same manner as a digital system telephone.

The operator console utilizes liquid crystal displays, LCDs, to show digit and symbol information needed by the operator to handle calls in an effective manner.

The operators can perform all normal extending facilities via the consoles, and by these also gain access to facilities and functions such as:

Fig. 16
MD 110 allows concurrent, mutually independent voice and data communications



- specific call supervision
- abbreviated dialling
- last number redial
- queue indication
- alarm indication
- time indication.

The specific call supervision function is a development of the normal parking, extending and recall functions. The operator console has four buttons that can be used when the operator parks or extends a call that needs supervising for some reason. The operator regains immediate access to the call by pressing the associated button again. One of the four supervisory links is also combined with a monitoring function.

Operation and maintenance functions

Particular attention has been devoted to the operation and maintenance functions in MD 110. Advanced functions for the management of on site data as well as for system supervision, alarm sending, fault locating and traffic recording are included in the system. The functions can be utilized on site or from a remotely situated operation and maintenance centre.

To all extents and purposes the operation and maintenance functions have the same design as those in ASB 900, an earlier Ericsson PABX[1].

All communications between operation and maintenance personnel and MD 110 take place via standard type I/O terminals. Up to six terminals can be used concurrently. These can be connected on site or remotely via modems. The man-machine language used is in accordance with CCITT's Man-Machine Language.

Commands exist for programming and reading all types of on-site data. If suitable, the operation and maintenance organization can transfer responsibility for frequent data changes to the customer.

MD 110 is supervised automatically by periodic program controlled function tests and with the aid of data from the ongoing traffic. Control and system functions as well as the function of individual telephony devices are supervised.

On discovery of a fault the alarm is issued on the operator consoles and, if required, on separate lamp panels. There are four different classes of alarm, that are used to indicate the degree of alarm urgency. Complete data concerning alarm origin can be obtained via an I/O terminal that can be situated on site or at an operation and maintenance centre.

With duplicated equipment, switching to the reserve unit takes place automatically on discovery of a fault. Faulty telephony devices are blocked automatically or designated as "last choice" in order to reduce the effect of the fault on traffic.

The alarm data normally contain all the information that is needed to localize a fault to one or a few boards. Program controlled function tests and controlled test connections, primarily intended for repair verification, can also be used for fault locating.

MD 110 contains functions for gathering, assembling and printing out traffic data for all device groups that require to be dimensioned with consideration to traffic. Data for operators and routes have been paid particular attention.

Power equipment

The PABX has a nominal 48 V operating voltage but this may vary within the range 44–54 V. The PABX is fed from the mains via an external rectifier. In order to ensure disturbance-free operation the rectifier is usually combined with batteries and charging equipment.

Conversion from 48 V to lower voltages and to ring voltage is achieved locally, per magazine. Boards containing d.c./d.c. converters and ring generators respectively exist for this purpose.

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1. Mörlinger, R.: *Operation and maintenance functions in ASB 100 and ASB 900*. Ericsson Rev. *57* (1980):4, pp 149–155.

Digital Signal Processing in System MD 110

Jonas Reinius, Bengt Svensson and Sven-Olof Åkerlund

MD 110 is a digital stored program control PABX whose general functions are described in the preceding article.
In this article the authors describe in greater detail the digital signal processing for various functional units in the PABX such as switches, multiparty conference units, tone transmitters and receivers and the working of digital extension lines.

UDC 621.395.2:
621.391.8

MD 110 is a digital PABX system built up from a number of line interface modules, LIMs, that are interconnected in a star configuration via PCM links to a group switch, GS[1], fig. 1. Each LIM can contain hardware for all functions of the PABX. In addition to adaptation units towards different line types LIM also contains a digital, non-blocking switch, LS, and a control processor, LP.

MD 110 is a fully digital system and uses PCM in accordance with CCITT recommendations, table 1. Voice and tone signals must therefore be converted into digital form in an analog environment. This analog to digital conversion takes place on an extension line circuit board, ELU-A, for extension lines, and on a bothway trunk line board, TLU-A, for public (exchange) and private (tie) trunk lines, fig. 2. The boards are equipped with individual codec and filters for each line.

Digital telephones with special functions, system telephones, can be connected to the PABX via normal two-wire extension lines. Board ELU-D is used for these. Speech is converted into digital form in the system telephone. Similarly, all signals corresponding e.g. to button depressions during dialling are converted into digital signal information. Digital voice and signal information are transmitted concurrently between the system telephone and PABX via the extension line that also has capacity for transmission of data.

Operator consoles are connected to the PABX via digital extension lines in the same manner as system telephones.

Digital public and private trunk lines are connected to the PABX via boards, TLU-D, with a capacity of 30 channels. MD 110 allows use of channel-associated signalling and also signalling via common channels.

Table 1
Characteristics of first order PCM

Voice frequency band	0.3 to 3.4 kHz
Sampling frequency	8 kHz
Bits per sample	8
Time slots per frame	32
Voice channels per frame	30
Transmission speed	2.048 Mbit/s
Companding	A-law

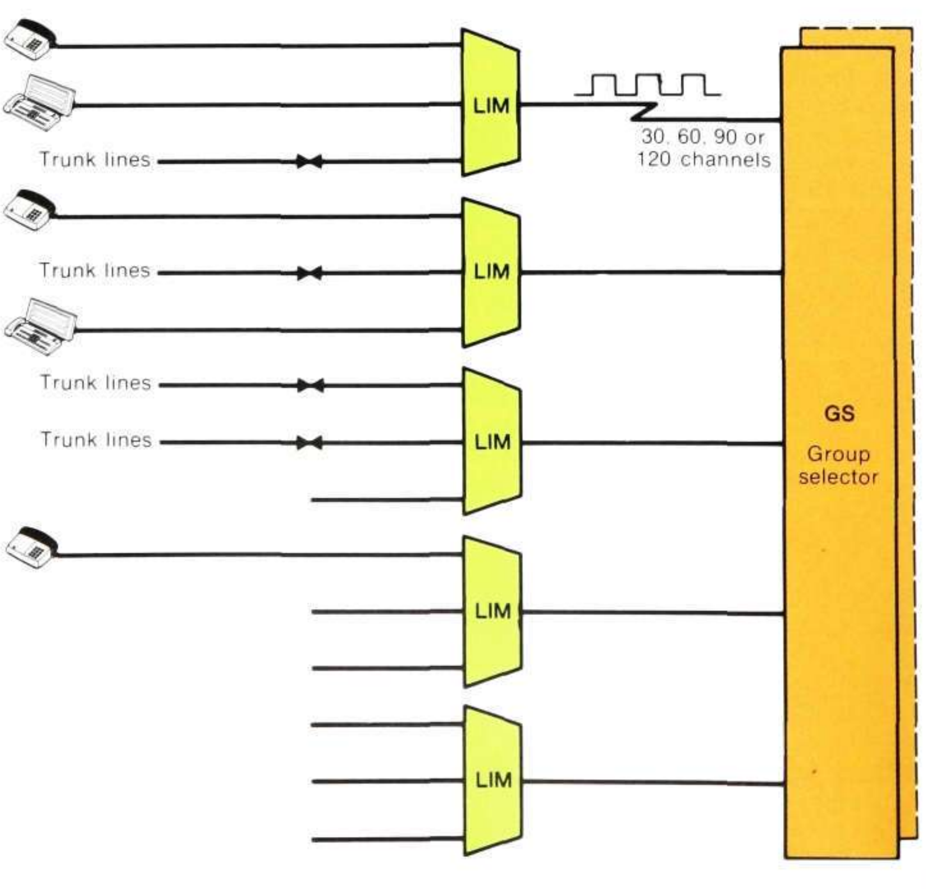

Fig. 1
System structure
The Line Interface Module, LIM, contains telephony devices, switch and control unit. Each LIM is independent of other LIMs and GS for internal connections. PCM links in accordance with CCITT recommendations are used between LIM and GS

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Digital switch

The digital switch in MD 110 consists of a time switch in LIM and a time switch in GS. Path selection through the PABX also includes selection of links in the PCM connections between LIM and GS.

The LIM switch has 512 positions in the voice memory and thereby facilitates 256 concurrent bothway connections without congestion. The switch contains a basic board, BSU, that contains voice memory and control memory. These memories are parity-checked, to provide immediate fault detection and prevent wrong connection.

The parity check is also used to supervise through-connection. Inverted parity is sent on the newly established connection and provides a fault indication. This "incorrect" indication for a correct connection at the correct time is used to indicate a successful through-connection.

The BSU board also contains PROMs for digital attenuation/amplification. If attenuation is to be inserted in a connection a section of the PROM will be addressed by an incoming PCM word. The new PCM word that is read corresponds to an attenuated/amplified signal. Eight sections exist for different attenuation values. These sections are normally programmed for an attenuation from −6 to +15 dB in 3 dB stages.

Attenuation in a connection is inserted in both the voice directions and mutually independently so that a bothway connection can be provided with different attenuations for the forward and backward directions.

The LIM switch also has two switching boards, SSU. The digital information is transmitted in serial form from the device boards to SSU. Each device board position has unique addresses corresponding to maximum eight device individuals. These eight individuals corre-

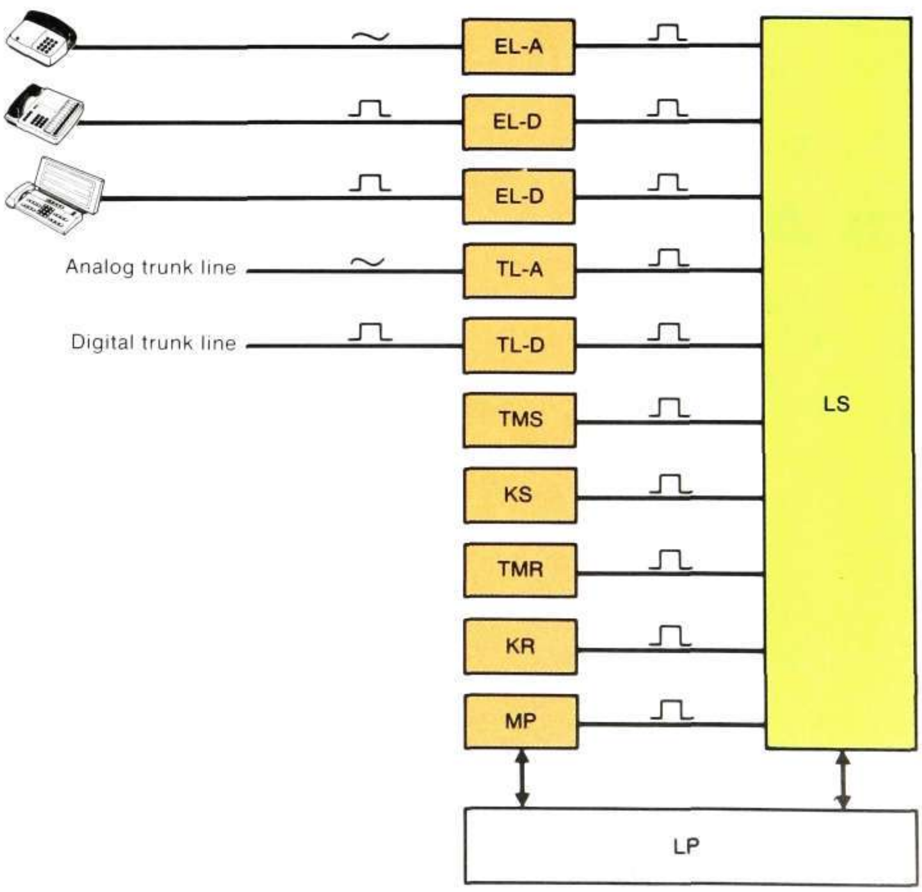


Fig. 2
A line interface module, LIM, comprises control system, switch and devices

LP LIM processor
LS LIM switch
EL-A Analog extension line circuit
EL-D Digital extension line circuit
TL-A Analog trunk line circuit
TL-D Digital trunk line circuit
TMS Tone message sender
KS Tone sender for DTMF signalling
TMR Tone message receiver
KR Tone receiver for DTMF signalling
MP Multiparty conference equipment

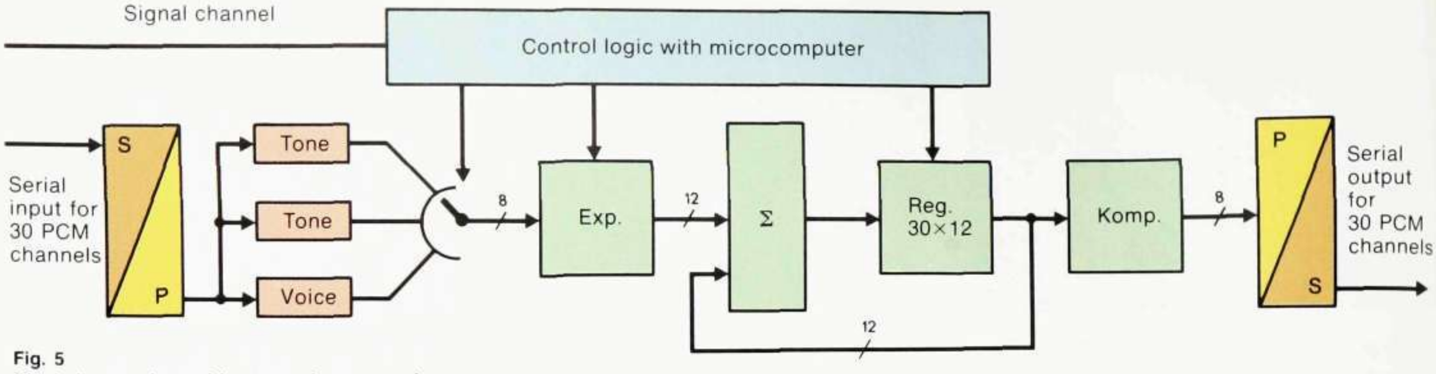


Fig. 5
Block diagram for multiparty conference equipment
The companded PCM code is expanded to a linear 12 bit format before the addition. Any necessary adjustment of the level of the voice signal is undertaken at the same time

S/P	Serial/parallel conversion
P/S	Parallel/serial conversion
Tone	Holding register for tone signals
Voice	Holding register for voice signals
Exp.	Expanding and level adjustment to linear code
Σ	Adder circuit
Reg	Intermediate storage register
Komp.	companding to 8 bits

spond to eight positions in the time switch.

If only every second board position is equipped it is also possible to use the addresses of the adjacent board and the device board can have 16 individuals. In the corresponding manner each fourth board position can be equipped with a board for 32 individuals.

The group switch, GS, comprises time switch modules with 930 switch positions corresponding to 31 PCM links. The modules are arranged in a matrix with maximum 8×8 modules. GS thus permits connection of maximum 8×31 = 248 PCM links, each possessing 30 voice connections.

The PCM links between LIM and GS consist of 2.048 MHz digital connections in accordance with CCITT recommendations. Each link comprises 30 channels for voice connection, T1–T15 and T17–T31, one channel, T0, for synchronization and supervision and one signal channel, T16, that is used for signalling between the processors in interworking LIMs.

Multiparty conference calls

Multiparty conference calls are possible in MD 110, e.g. on operator call extending and conference calls. Fig. 3 shows the traffic pattern for operator call extending. The operator has access to bothway voice connection with an external or iternal party or with both simultaneously. A special warning tone is issued if more than two parties are interconnected. Attenuation of the voice signal in the multiparty conference unit can be selected individually from eight levels for each voice path and voice direction. Up to eight participants can be connected concurrently to a multiparty conference call.

The voice signals are added linearly in the multiparty conference unit. The sum of the voice information of the other participants is added for each participant. With multiparty conference units of this type some risk of instability exists when there are many participants due to reflections in hybrid circuits. To lessen this risk the voice signal from every second channel is inverted so that disturbance signals in odd and even

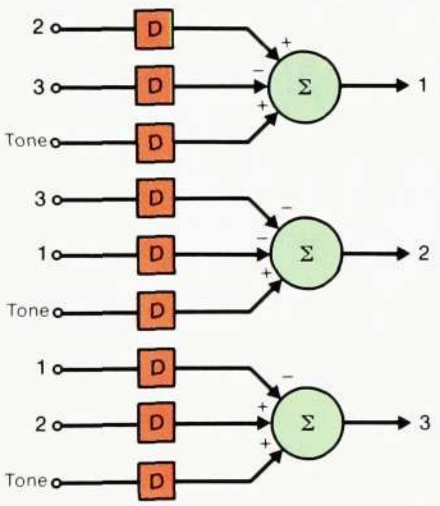


Fig. 4
Addition principle for multiparty calls. Each party receives the sum of the voice signals of the other parties and warning tone. Odd voice channels are subtracted

D	Digital attenuation
Σ	Adder circuit
1, 2, 3	Voice signals

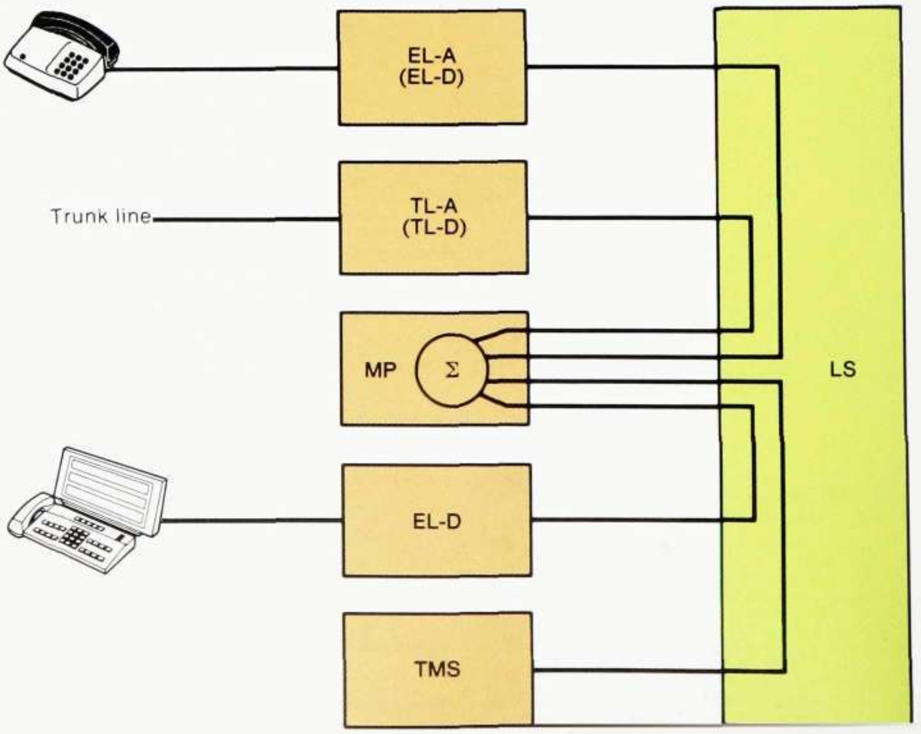


Fig. 3
Connection of operator

LS	LIM switch
TL-A	Analog trunk line circuit
TL-D	Digital trunk line circuit
EL-A	Analog extension line circuit
EL-D	Digital extension line circuit
MP	Multiparty conference equipment
TMS	Tone message sender for warning tone

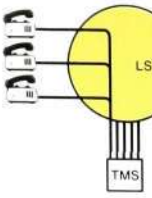
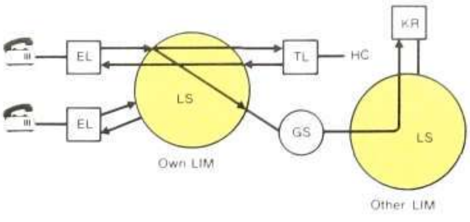


Fig. 6a
The tone message sender is connected by a fan arrangement

Fig. 6f
Inquiry position with "refer back" possibility. A tone receiver, KR, e.g. in another LIM, is connected in parallel in order to detect transmission of a DTMF digit that signifies a refer back request



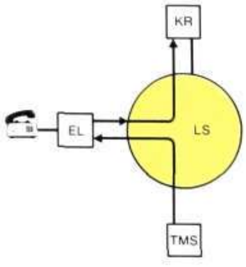


Fig. 6b
Register position with dial tone

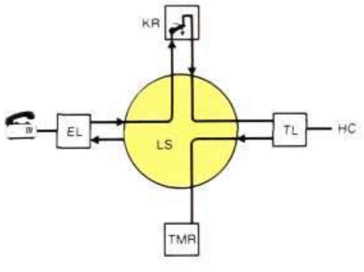


Fig. 6c
Waiting for dial tone from public exchange

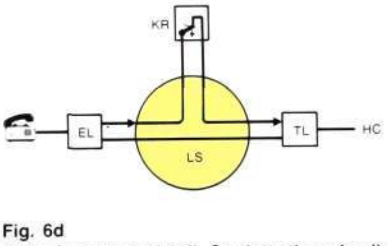


Fig. 6d
Outgoing external call. On detection of a digit in the DTMF receiver, KR, the voice path is disconnected in KR to prevent the public exchange from comprehending the digit

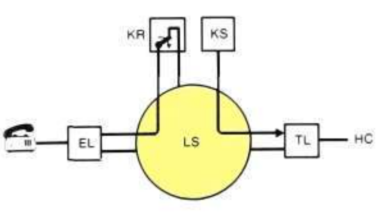


Fig. 6e
Retransmission of DTMF digits. The DTMF sender, KS, is released and the voice path is re-established if no digit exists to send

channels are subtracted from one another, fig. 4.

The multiparty conference unit comprises one board, MPU. The board has two entries for warning tones and 30 voice channel connections. The control logic on the board handles the connection and disconnection of parties and interruptions of warning tones. Several concurrent multiparty connections are possible with up to four different tone cadences, fig. 5.

Tone sending and tone receiving

Tone senders and tone receivers are fully digital auxiliary devices. They are connected towards the line with the LIM switch, LS, when needed, fig. 6.

Traffic cases

The tone message sender is connected without risk for congestion as all users of a tone message are connected to the same exit. Dial tone receivers, DTMF (Dual Tone MultiFrequency) senders, DTMF receivers, MFC senders and MFC receivers are individual devices and congestion can occur if all devices of a particular type are busy.

To distribute the load when devices are lacking, a LIM (about 200 extensions) can request connection of a device in an adjacent LIM. If no device is available in this LIM the request will be queued in the own LIM if a queue position exists, fig. 6f.

To prevent pulses longer than 20 ms initiated via DTMF telephones from being issued towards the external line, but nevertheless permit voice connection during call establishment, the DTMF receiver is connected in series with the voice path for outgoing external calls. In this manner it is possible to lead normal speech via the DTMF receiver, but it will be delayed 13 ms. Transmission of an approved DTMF digit disconnects the voice path. The voice transmission is re-established after an approved pause, fig. 6d.

The DTMF receiver is connected in parallel with the extension in the inquiry and conference states in order to facilitate "refer back" for parties with DTMF telephones. In this state the voice reproduction protection has been increased in that the period for approved button depression is increased to about 100 ms.

Auxiliary devices

Tone senders for up to 24 tone messages and eight DTMF senders are contained on one board. One such board is placed in each LIM in order to provide non-blocking transmission of tone messages, fig. 7.

Four tone receivers for dial tone from external lines and eight DTMF receivers are gathered into one unit comprising two boards. One such unit is placed in each LIM for those PABXs that require one or both of these functions.

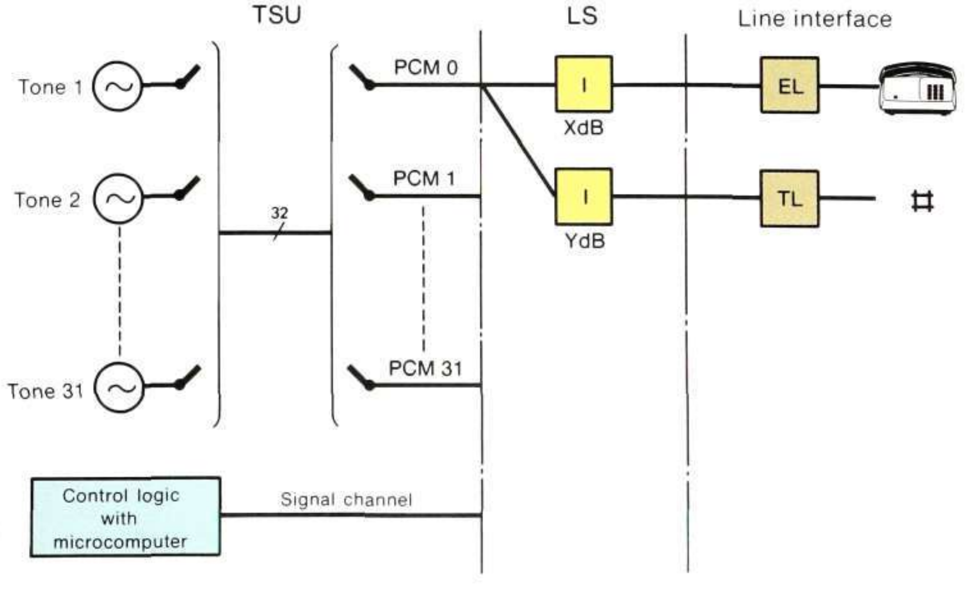
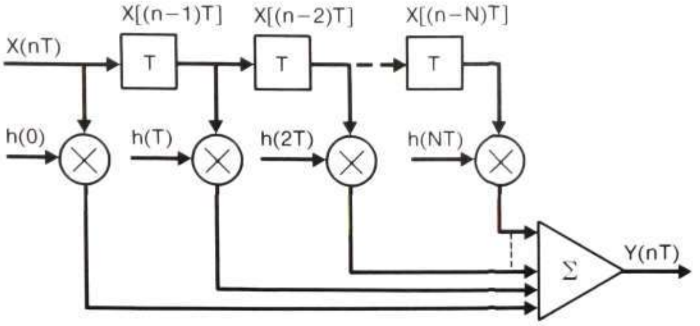


Fig. 7
The tone sender unit, TSU, has a number of tone generators that can be connected to any PCM channel. An unlimited number of users can be connected to each PCM channel and be provided with individual attenuation in the switch, LS

$$Y(nT) = \sum_{k=0}^{N} h(kT) \cdot X[(n-k)T]$$



$$Y(nT) = \sum_{i=0}^{\frac{N-1}{2}} h(iT) \cdot \left\{ X[(n-i)T](\pm)X[(n-(N-i))T] \right\}$$

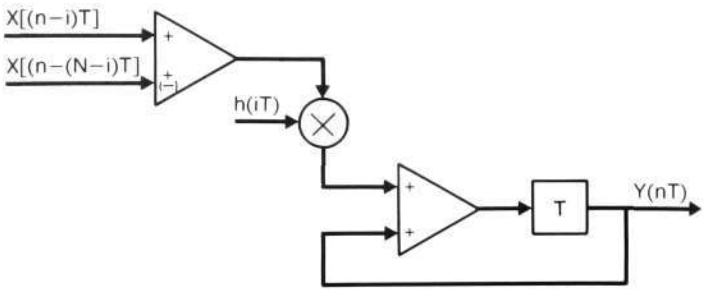


Fig. 9
Calculation formula and block diagram for a normal, non-recursive filter

Fig. 10
Calculation formula and block diagram that demonstrate the use of the symmetry characteristics and sub-sampling of the filters

A board with eight senders and two boards with together four receivers exist for MFC signalling. The MFC receiver can be programmed to detect either forward or backward signals on connection to an external line. An external line can request connection of MFC equipment, irrespective of whether its own LIM possesses these devices or not.

The functions of MFC senders and receivers are similar to DTMF senders and receivers and consequently are not described specifically.

The quantity of tone senders and receivers can be duplicated when a greater device capacity is required or requirements exist for better reliability. Senders and receivers are supervised on a routine basis by interconnection towards one another.

Tone sender

The tone sender has a tone generator part and a control part. The tone generator part generates continuous tones that are synchronized with the PCM rate. The control part connects a certain tone to a certain PCM channel and switches between tone and silence in a tone message. The tone generator can generate 32 tones, that can be connected to any number of the 32 PCM channels, fig. 7.

The tones are stored in a PROM as a number of full waves of the required wave form. The sampling speed is 8 kHz and each tone is described with 240 alternatively 480 eight bit PCM samples. For a full wave this provides a lowest frequency of 33 1/3 Hz for 240 samples and 16 2/3 Hz for 480 samples. As the tones must be stored as a number of full waves of the wave form the tones become multiples of 33 1/3 Hz and 16 2/3 Hz respectively. The PROM has capacity for thirtytwo 240 sample tones or for sixteen 480 sample tones or a combination. The DTMF tones for digits 0−9 and the star and square buttons occupy half of the memory, fig. 8.

The control part assembles complete tone messages and connects these to PCM channels on the order of central software on PABX start or on start of processors on boards e.g. after board replacement. This gives great flexibility and the number of market-adapted boards that needs to be manufactured is reduced considerably. The tone levels can be adapted on connection to a specific line by using the attenuation in the LIM switch, fig. 7.

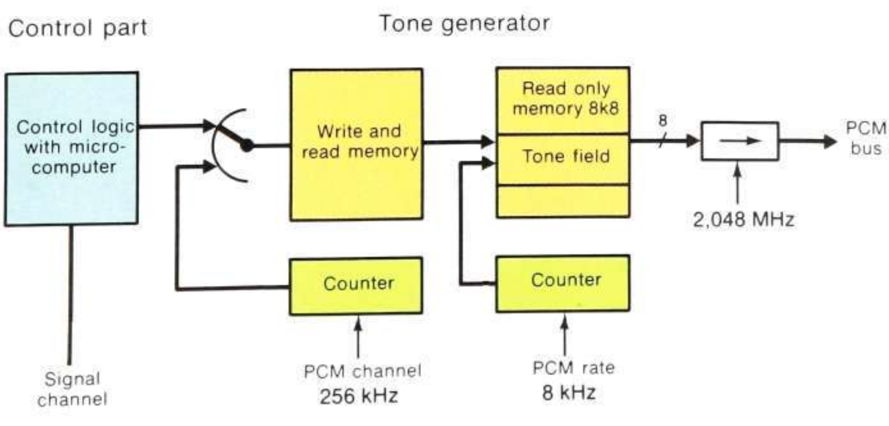
The control part also supervises the repeated sequences for switching between tone and pause e.g. in busy message and switching between the three tones and pause in the diversion message. DTMF transmission is a special case of a non-repeated signal sequence.

Tone receiver

The tone receiver uses digital FIR (Finite Impulse Response) filters. Two filters with a mutual phase displacement of 90° for each frequency are used to detect the amplitude of the signal, irrespective of the phase position. The output signals are added vectorially and the absolute value of the total provides the amplitude value of the signal.

A total of 112 filters is required for eight code receivers and four tone receivers. 18 filters for 2×4 channel detectors and one detector for frequencies in the range 2 to 4 kHz exist for each code receiver individual. The receiver will be blocked if voice frequencies are detected in this range thereby increasing the voice immunity of the receiver. Two

Fig. 8
Block diagram for tone sender, TSU.
The control part achieves connection of the tone generator by writing into the read/write memory the address of the field with tone sample that is to be transmitted on a PCM channel. The read/write memory is addressed on reading by a counter clocked with 256 kHz corresponding to the PCM byte rate. The start address of the field in the read memory containing the sample for the tone that is to be transmitted on PCM channel 0 is written at address 0 in the memory, and so on. The read only memory is also addressed by a counter clocked with 8 kHz corresponding to the PCM sample rate for consecutive addressing of the 240 (480) samples to the shift register. This is clocked on to the PCM bus at a rate of 2.048 MHz



filters for each of the two types of dial tone that the receiver can detect exist for each dial tone receiver.

The calculation formula and block diagram for a normal, nonrecursive digital filter are shown in fig. 9.

By using the symmetry characteristics of sine and cosine filters and by using the same accumulators and multiplicators in time division multiplexing, the formula and block diagram for each diagram can be described as in fig. 10. The calculation is undertaken in an arithmetic processor. The latter is built up with discrete standard MSI/LSI circuits and the function in these is controlled by a microprogram comprising 1024 words each 32 bits long. The clock frequency for the program memory is 8 MHz and this gives a cycle time of 125 μs, corresponding to an 8 kHz sampling speed.

The arithmetic processor interworks with a standard microprocessor via a common result memory. The vectorial addition of the filter results and the analysis of the mutual amplitude levels of the channel detectors that are required for approval of the DTMF code are carried out in the microprocessor. A DTMF code shall consist of one frequency each from the low group and high group for a predetermined period followed by a pause with a predetermined minimum length.

The fact that the amplitude of the input signal is more or less constant during the greater part of the reception period has been used to reduce the number of calculations per filter. It is thus unnecessary to calculate the amplitude of each incoming PCM sample, merely once every 104th sample, which is equivalent to once every 13 ms approximately.

The filters function on the linear principle and their coefficients lie pair by pair, symmetrically around the centre point of the filters. The coefficients in a pair can have the same or opposite sign. To reduce the number of multiplications the symmetry is used so that samples corresponding to symmetrically situated coefficients are added/subtracted before the multiplication is undertaken.

The highest frequency for DTMF signalling is 1633 Hz. A 4 kHz sampling speed is therefore adequate if the effect in the band 2 to 4 kHz is low enough. Thus, only every second sample is used in the calculation for the channel filters. Every sample is used for the other filters.

The amplitude in the 2 to 4 kHz detector is compared with the amplitude of the strongest signal in the low group. If the difference is great enough the foldover distortion around 2 kHz is so little that 4 kHz sampling can be permitted.

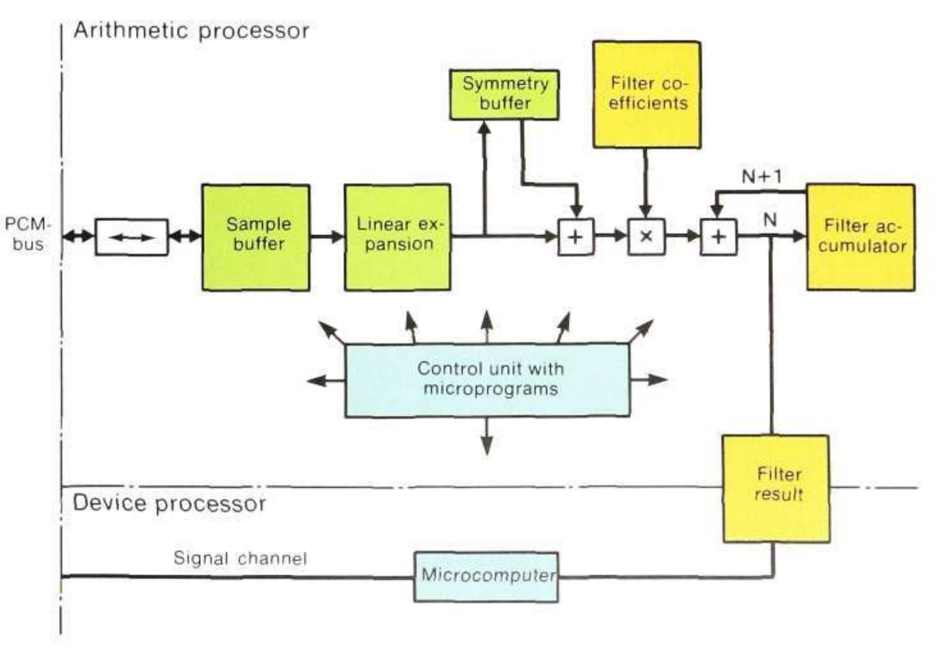
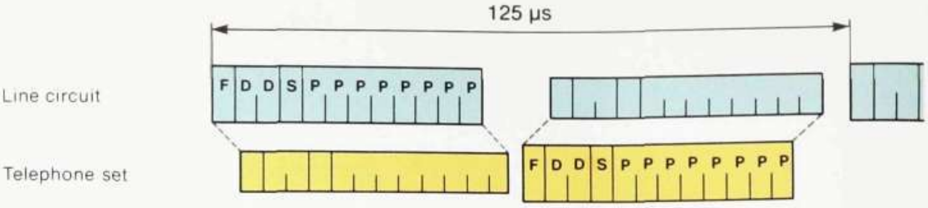


Fig. 11
Block diagram for the tone receiver, TRU.
The tone receiver consists of an arithmetic processor for filter calculations and a microcomputer for analysis of the filter results.
104 PCM samples are stored for 13 ms in the **sample buffer. After linearization they are added to the symmetrical samples of the centre point of the filter before they are multiplied by the corresponding coefficient. The sum of the 52 products is then formed. After the last addition the sum is read into the filter result memory. The microcomputer is** ***informed*** **that new filter results exist and starts the analysis**

Fig. 12
Principle for burst signalling
Each burst comprises 12 bits of biphase coded information. The bit frequency is 256 kbit/s

F Frame synchronization bit
D Data information
S Signal information
P Voice information (PCM)

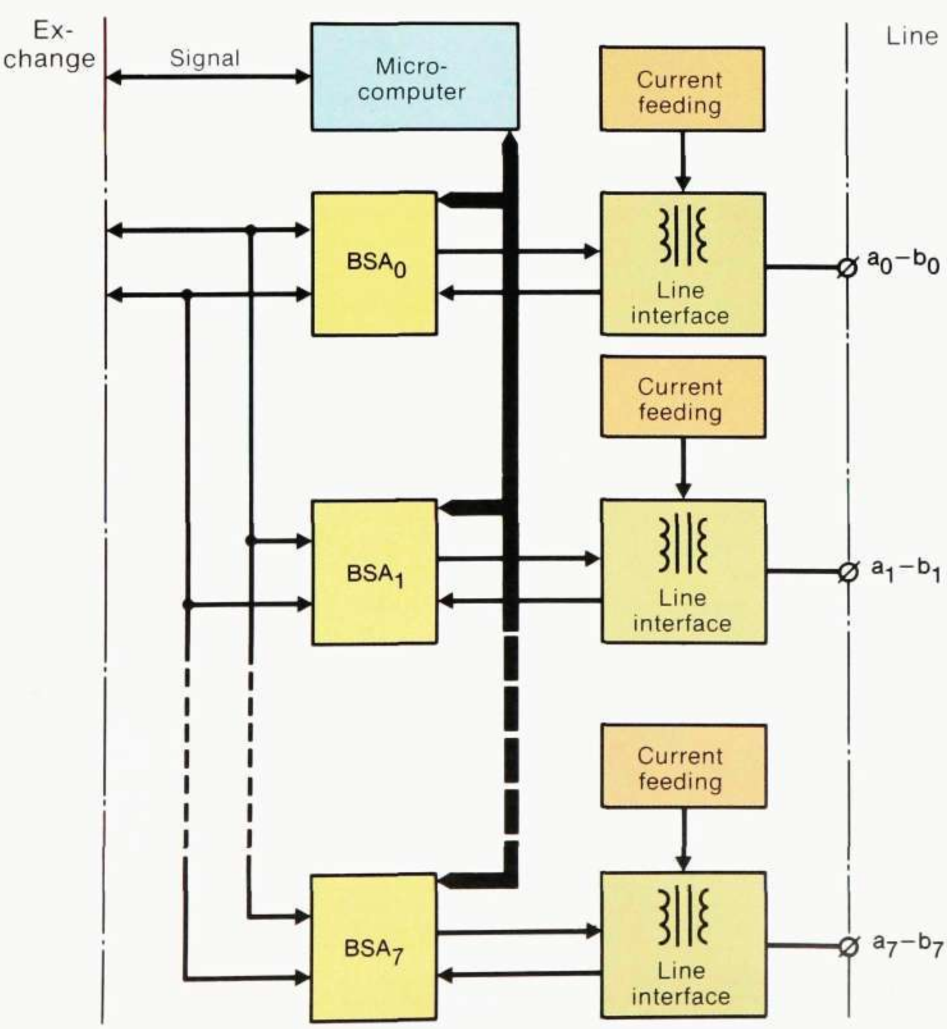


It has been possible to reduce the number of filter calculations for each DTMF receiver from 18×104 for each DTMF receiver every 125th µs to 5×104 for each receiver every 13th ms, fig. 11. The number of calculations in respect of the FIR filters is otherwise so great that it would not have been possible to build such a compact receiver without this reduction in the number of calculations per PCM sample.

The threshold levels for a normal line and a long line are set on start of the PABX. The applicable level and the detection times for tone and pause in multiples of 13 ms intervals are determined each time a DTMF receiver individual is selected.

The threshold level and detection time for the dial tone receiver are set in conjunction with start of the PABX.

Fig. 13
Digital extension line unit board for eight extensions, TLU-I



Digital extension line

The 2-wire digital extension line in MD 110 provides several advantages from the transmission and system aspects:
- 4-wire transmission through the entire connection guarantees voice transmission free from return loss
- The levels of the voice channel are well defined as they are independent of the length of the extension line
- Voice communication, data communication and signalling take place on separate channels and can progress individually of one another
- The signal transmission complies with highly stringent requirements as regards flexibility and capacity
- The 2-wire connection facilitates the use of existing network equipment for analog systems after digitalization.

Signalling

Time Division Multiplex, TDM, is used for line transmission. Information is interchanged alternately between the PABX extension line unit board and the digital system telephone in the form of bursts, fig. 12. The extension line circuit transmits a burst every 125th µs. The telephone is synchronized with this rate so that it sends its burst immediately after a burst has been received.

From the figure it will be seen that the moment of reception on the extension line board is dependent on the line delay. This delay, together with length and repetition frequency of the burst, sets a theoretical maximum transmission length for the TDM method. The limit in this case lies at a line length of about 2 km. Other factors such as current feed, attenuation and signal to noise ratio limit the range to 1 to 2 km.

The line transmission containes three mutually independent channels:
- PCM channel, with capacity 64 kbit/s
- signal channel, for transmission of control information with 8 kbit/s
- data channel, that serves a data terminal interface for 16 kbit/s which is connected to the telephone.

The signal bits, one from each burst, are assembled to a word format consisting of eight signal bits, synchronization bits and parity bit. In their turn these signals

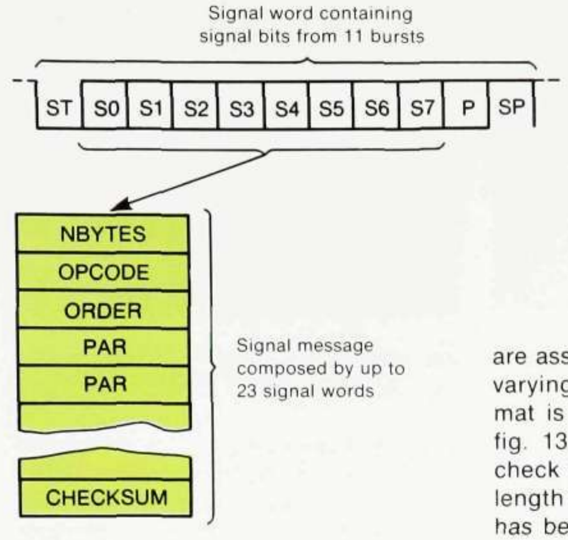


Fig. 14
Signal format

ST, SP	Word synchronization
S0-S7	Signal bits
P	Parity bit
NBYTES	Message length
OPCODE	Priority level
ORDER	Order code, e.g. start flashing
PAR	Order parameter, e.g. LED number
CHECKSUM	Check sum for checking contents of the signal

are assembled into signal messages of varying lengths. The same message format is used in both signal directions, fig. 13. The signal message contains check information in the form of stated length and a check sum. After a signal has been approved by the receiver an acknowledgement is issued to the sender in the form of a predetermined signal word. Both sender and receiver subject the signal transmission to time supervision. In the event of time release a new attempt to send is made. A message can contain several orders, e.g. "extinguish lamp 31 and start the tone ringer".

Design

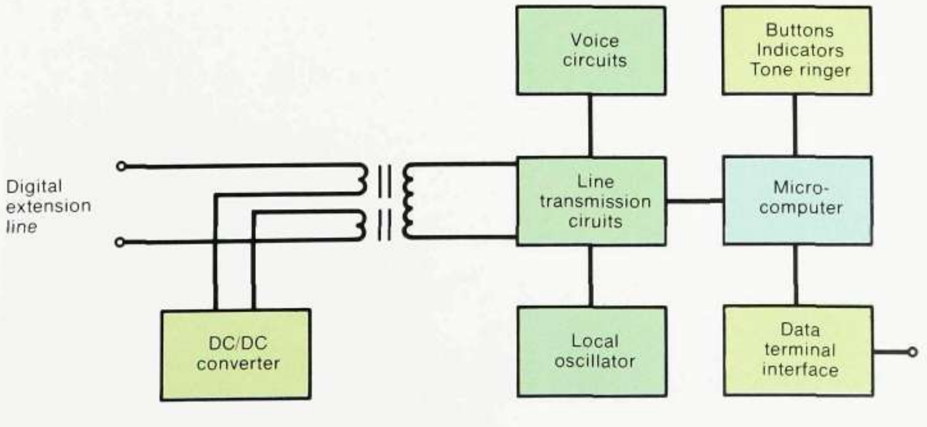
All logic functions required to transmit and receive information via the line and to assemble and disassemble the signal words in accordance with the required format are housed in a custom designed CMOS circuit, BSA (Burst Signalling Adapter). The circuit can be programmed to function as controller on the extension line unit board or as slave in the telephone. Fig. 14 shows a block diagram for the extension line unit board. A microcomputer whose task is to manage the message processing controls eight BSA circuits. The circuits are also connected towards the system switches for voice and data. Line driver circuits and input stages are built into the circuit on the line side. Externally there is a passive bandpass filter, current feed circuits and protection against alien voltages. The current feed is lowohmic as there is no reason to detect any current loop for this application. As a consequence the power losses in the current feed circuits will be relatively small. An electronic protector disconnects the current if line overloading occurs.

Fig. 15 shows the different blocks in a digital system telephone. The line voltage is separated in a line transformer and converted into logic voltages. Incoming burst signals are filtered and decoded in the BSA circuit so that the burst transmission is synchronized with the extension line board. Codec and filter circuits exist for the conversion of PCM signals to analog voice signals and vice versa. A CMOS microcomputer attends to signalling, detection of button signals and the control of lamps and tone ringer. This microcomputer also supervises the auxiliary unit for data terminal connection.

Products

Several products have been developed for the digital extension line:
- the extension line unit board described above, with 8 lines per board
- the operator console with 23 LCD digit indicators and some 30 LCD function indicators
- loudspeaking telephone with 12 or 36 facility buttons and LEDs and an 18 digit LCD display. These telephones can be equipped with an adapter unit for a data terminal interface.

Fig. 15
Block diagram for the digital system telephone in MD 110



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Nordic Public Data Network with AXB 30

Olov Sjöström and Eric Strindlund

The public data networks in Denmark, Finland, Norway and Sweden were taken into full service in 1981. When the need for a public data network was felt in the Nordic countries, the Administrations in the four countries jointly prepared guidelines and specifications for a system that met their requirements. On the basis of these requirements the Ericsson system AXB 30 was developed by ELLEMTEL, the development company owned jointly by Ericsson and the Swedish Telecommunications Administration. The production and a part of the development work was shared by manufacturers in the four countries concerned. The authors describe the events leading to the development of AXB 30 and the structure and properties of the system from the point of view of the user.

UDC 681.327.8:351.817(48)

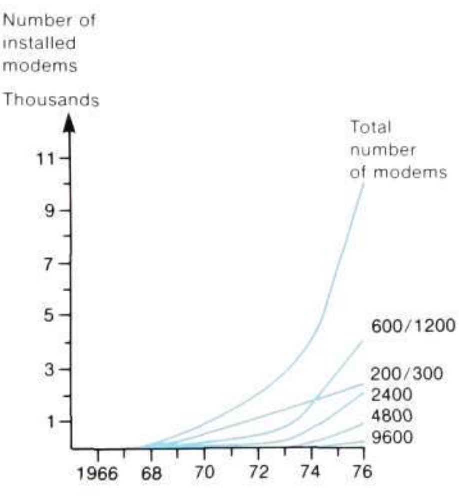

Fig. 1
The increased need of data communication is illustrated by the number of modems installed in Sweden during the years 1966–1976

	Denmark	Finland	Norway	Sweden
1971	644	266	293	1 266
1972	878	606	495	1 677
1973	1 414	859	802	2 333
1974	2 161	1 327	1 042	3 495
1975	2 996	1 727	1 538	5 488
1976	4 600	2 250	2 250	8 500

Table 1
The number of modems installed in the Nordic countries during 1971–1976

The need for a public data network

By the beginning of the 1960s the use of computers was so widespread in the Nordic countries that the need for data communication had arisen. The existing telephone network offered a solution, and the Administrations considered it a matter of urgency to find technical means of utilizing the network.

The introduction of data transmission has not been wholly trouble-free for any telecommunications administration. The high data speeds meant that new transmission demands were made on the telephone lines. The very short connection times that were required could not be achieved in the switched telephone network, and in certain cases point-to-point circuits were therefore the only feasible medium. Such circuits were leased from the administrations.

Eventually the number of terminals for one and the same computer amounted to several hundred. Separate data networks were established for individual users. They were often both complicated and sensitive to operational disturbances. The data transmission required modem equipment for converting digital signals into signals that were suitable for transmission over analog telephone lines. It has been the policy of the Nordic Administrations to provide and maintain modem equipments and lines. The customers are usually responsible for data terminals and computers.

The data service grew very rapidly. During its first decade annual increases of between 50 and 100% were recorded, fig. 1 and table 1. The prognoses indicated continued large growth for several years.

By the beginning of the 1970s it was obvious that something had to be done to rationalize the data traffic and give it a structure similar to that of the telephone traffic. It could not be rational to have data traffic transmitted in several parallel but entirely separate data networks, which were built up of point-to-point circuits leased from the Admin-

Fig. 2
A work position with a display unit (DTE). On the far right the line terminal (DCE) of the data network

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istrations. A general data network was required which met the customers' demands for very short setting-up times, high transmission speed and high operational reliability. In such a network the data traffic can be handled in a similar way to the telephone and telex traffic (circuit switching).

Specification stage

The Nordic countries have cooperated in the field of telecommunications since 1858. Since 1917 this cooperation has been organized in the form of Nordic telecommunication conferences, which are held every few years, now biannually. Committees and working groups maintain the collaboration in the periods between conferences.

The whole complex of problems concerning data communication was discussed during the Nordic telecommunication conference held in Reykjavik (Iceland) in June 1971. A working group was then formed with the task of, among other things, studying and preparing proposals for
- traffic between the Nordic data networks
- standardized interfaces between data networks and subscribers and procedures which were common as far as possible
- joint contributions to CCITT and CEPT in matters regarding data networks.

In 1973 the working group prepared a specification for a hypothetical Nordic data network (HND). The work was based on the following prerequisites:
- HND was to be dimensioned for the needs of the 1980s and 1990s
- the first networks were to be installed by the end of the 1970s
- the specifications were to be optimized for 1985
- applicable CCITT and CEPT recommendations were to be followed.

As a result of the rapid international standardization work the group agreed that the Nordic data network should be a fully synchronized circuit switching network with time division multiplex (TDM) and integrated switching and transmission (IST).

In 1973, according to plan, the group presented the specifications for HND. At that time the group had also contributed to CCITT and CEPT in matters concerning
- multiplex structures
- interfaces
- services and functions
- signalling.

Using the HND report as a specification the Administrations invited budget tenders in order to investigate whether it would be possible to build a network in accordance with this specification and to be able to assess the costs and time requirements.

Fig. 3
The control room for AXB 30 in Copenhagen, Denmark

Purchasing stage

In 1974 the four Administrations decided to let the group prepare tender documents for data networks based on a common specification, since the budget tenders showed that the technology, costs and time requirements were reasonable.

The purchasing stage comprised the years 1974–76, and resulted in Ericsson being given an order for four data network exchanges, in Helsinki, Copenhagen, Oslo and Stockholm, and also

87 concentrators
59 multiplexors
764 modems for 64 kbit/s
11 400 subscriber connection equipments.

The Nordic public data network was designated NPDN, fig. 4. It consists of four separate networks, one in each country, with traffic between the countries and with other countries. The contract also includes options on equipment for the requirements up to 1985.

Fig. 4
The extent of the Nordic data network

Delivery stage

When deciding on the system design for NPDN the four Administrations and Ericsson wanted to utilize the experience already gained, above all in the development of AXE 10 and AXB 20 and from the Nordic trial data network. The system was designed for the same construction practice and the same central processor as AXE 10 and AXB 20. This made the system a member of the AX family, and it was designated AXB 30.

Already during the contract negotiations close collaboration was envisaged between the Administrations and the supplier as regards the system design. The collaboration was carried out in different groups, which included a large number of technicians from the four Administrations. Among other things the technical specifications were made more detailed, and the functions of AXB 30 were given their final form. In general they correspond to the Administrations' requirements and also Ericsson's requirements for system designs.

The trial data network provided some experience concerning concentrators, multiplexors and modems. This experience has been utilized during the very extensive design work, which was carried out by ELLEMTEL during the years 1976–81.

Manufactured locally

Agreements had been made between buyers and supplier regarding manufacture in all the countries and exchange of technical knowhow. In accordance with these agreements manufacturers of data equipment in the purchasing countries were subcontracted for various part deliveries for NPDN.

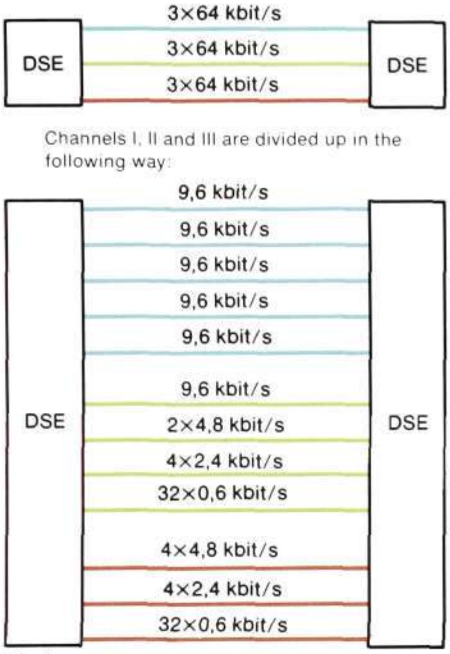


Fig. 5
An example of the distribution of channels between two data network exchanges in accordance with CCITT Rec. X.54

Channel I
Channel II
Channel III

For example, the equipment purchased from Finland includes subscriber connection equipment, containing baseband modems, for delivery to Finland and Denmark, and subscriber line modems for 750 bit/s and 3000 bit/s.

Norway provides concentrators and multiplexors to all contracted data networks, and also subscriber connection equipment for Norway and Sweden.

Denmark has manufactured baseband modems, magazines and exchange cabling. The software for the majority of the concentrator and multiplexor functions has also been produced in Denmark. Furthermore the identity card readers, which are used to check the authorization of the staff to carry out different types of work on the exchange equipment, are made in Denmark.

Great attention has been paid to quality control in the production. For example, the Administrations have prepared routines for supervising the production quality and have ordered continuous reports on the result of the quality testing of the manufactured units.

Installation

Like the other systems in the AX family, AXB 30 has functional modularity, so that functions can easily be changed without any traffic disturbances. This is done by dividing the functions into groups and then introducing these in different stages.

The design work was more extensive than had been expected, which led to delays, and it was considered most suitable to introduce the functions in stages. The two first stages comprised traffic handling functions and essential operation and maintenance functions.

The subsequent stage allowed more subscriber facilities and new types of traffic, and a more extensive maintenance system.

Thanks to the modification methods developed for the AX systems it was possible to introduce the functions in stages and to carry out correction of hardware and software during operation. Current modifications are assembled into logic packages, and each modification package is carefully tested in a test exchange before it is introduced.

When NPDN was taken into service it comprised four data network exchanges, 63 concentrators and 40 multiplexors. The equipment in Sweden, Norway and Finland has been installed by Ericsson. In Denmark Ericsson has been responsible for the installation of the exchange and the concentrators in Copenhagen and Århus. The Danish Administration has carried out the installation and testing of the remaining equipment as well as the connection of lines and subscribers.

Follow-up of the design

Three system test plants (SPA) have been installed at ELLEMTEL and another at LM Ericsson A/S in Copenhagen for verification of the design work. A special unit for operational back-up and final testing of new equipment and modification packages has been set up at MIPSC (Maintenance, Installation and Production Support Centre) in Älvsjö, near Stockholm, Sweden.

The design work on AXB 30 has also included development of test aids, for example a data traffic generator and tester, DCDMT, for concentrators and multiplexors.

The Administrations have followed the design work closely by means of
- discussions within different groups
- correspondence concerning the examination of specifications
- analyses of the design prerequisites for hardware and software units, commands and printouts
- examination of the technical contents and design of the documentation.

Field testing of the system was carried out by the Administrations together with Ericsson. The testing instructions have been prepared jointly and designed to verify specified functions and test the system in various fault situations. Any deviations that have been encountered have continuously been investigated and corrected when necessary. The tests covered both hardware and software, and comprised all types of equipment, including all I/O devices.

In addition each Administration carried out rigorous acceptance tests on the data network installed in the country before it was put into service.

Training
The Administrations' technicians have been trained by means of an extensive program, comprising about 15 000 training days. The program included a general system course on ASB 30 and eight special courses (operation and maintenance, installation etc.).

Documentation
A complete exchange library comprises over 250 A4 binders. The requirements for documentation have been stringent, to the extent of exceeding the accepted AX standard. Every document has been examined by the Administrations as regards technical and linguistic content. The documentation has been prepared in English and in certain parts also in the language of the country in question.

Ericsson has transferred certain documents to microfiche in order to make the document handling easier for the operation and maintenance staff.

Technical characteristics

In accordance with the specifications AXB 30 functions in the following way:
- the network is fully synchronous and designed for circuit switching
- all existing transmission systems, both analog and digital, can be connected
- applicable CCITT recommendations have been followed
- an advanced operation and maintenance system is provided, which meets stringent requirements for availability, operational reliability and low maintenance costs
- the traffic handling ability and transmission quality are good (bit error rate lower than 10^{-6}).

Subscriber terminals for synchronous data circuits and data speeds of 600, 2400, 4800 or 9600 bit/s can be connected to AXB 30. Asynchronous terminals for transmission speeds up to 1200 bit/s can also be connected.

The circuits have bit sequence independence and the transmission can be carried out in full duplex. Both bit and octet

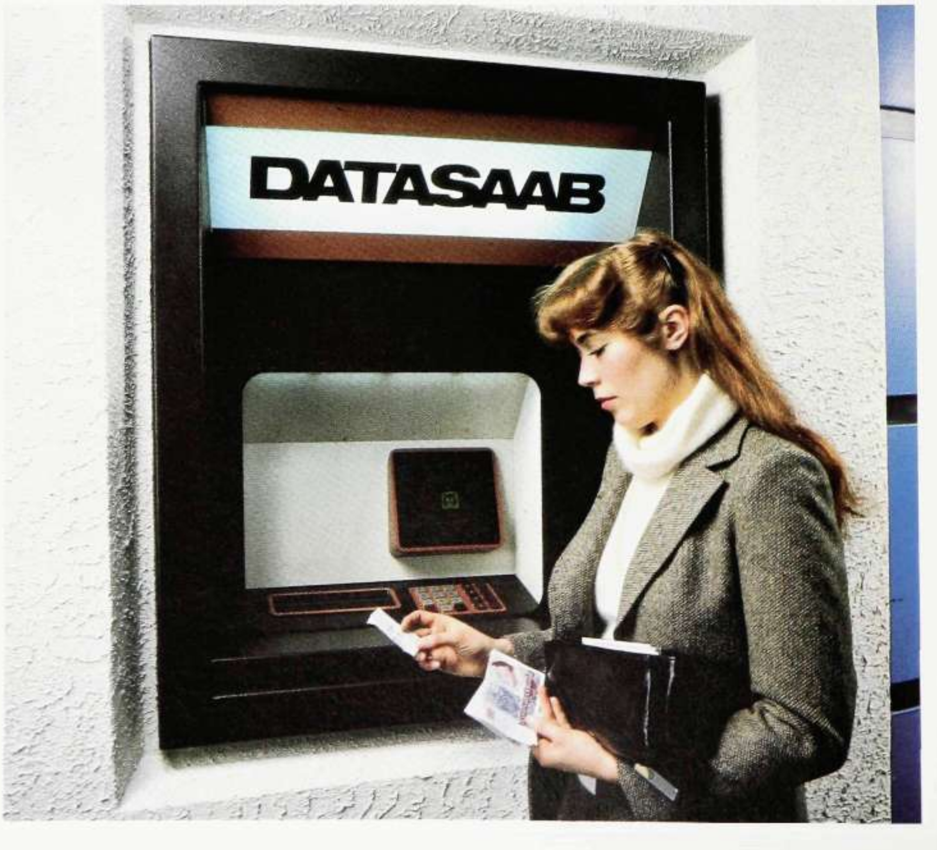


Fig. 6
Bank cash point terminal

Table 2
Status signals in the public data network

No. *Designation* and import

02 *Redirected call.* The called subscriber cannot receive data (e.g. because of a breakdown). The call is redirected to another subscriber. Wait for connection.

03 *Connect when free.* The called subscriber is engaged. The call is put in a queue. Setting-up will be completed when an input becomes available.

20 *Try again.*

21 *Number busy.* Try again later.

22 *Selection signal procedure error.* The selection procedure has not been followed.

23 *Selection signal transmission error.* A fault has occurred in the selection signalling during transmission to the switching exchange.

41 *No authorization,* e.g. because the called subscriber is not a member of the same closed user group.

42 *Number changed.*

43 *Not obtainable.* The called subscriber is no longer connected.

44 *Out of order.* The called subscriber is temporarily out of operation.

45 *Local mode.* The called subscriber operates in the local mode.

46 *DTE fault at called subscriber.* The called subscriber is out of service.

47 *Power failure* in the called DCE.

48 *Invalid facility request.* Attempt at using a facility that is not allowed.

49 *Network fault in local loop.* A fault on the line to the called subscriber.

51 *Contact the Telecommunications Administration.*

52 *Incompatible user class of service.* The called subscriber belongs to another speed category.

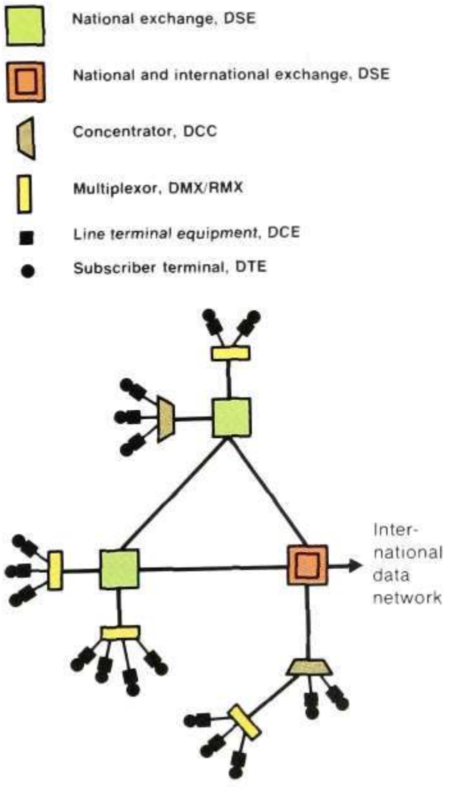
61 *Network congestion.* Temporary congestion in the network.

In addition the following signals are used in tests initiated by subscribers from their DCEs:

69 *The push-button set functions satisfactorily.*

96 *Test result satisfactory.*

Fig. 7
The hierarchic structure of the AXB 30 data network



timing are generated and controlled by the network and transmitted to the connected data terminals.

Data transmission over trunk circuits is carried out in time multiplex at a speed of 64 kbit/s, both between regional units and the data network exchange and between the exchanges. When necessary several such 64 kbit/s circuits in their turn can be multiplexed to higher speeds.

Data circuits are set up and disconnected with the aid of signalling systems recommended by CCITT, and the processes are extremely fast. As regards subscriber signalling, Recommendation X.21 or X.21 bis is used for synchronous terminals and X.20 bis for asynchronous terminals. A channel-associated signalling system in accordance with Recommendation X.71 is used for the signalling between data network exchanges. A signalling system for common channel signalling in accordance with Recommendation X.60 has also been anticipated.

The synchronization between different network components (including PCM) is carried out in accordance with the master-slave method using a national master clock, of the atomic clock type, which is connected to the exchanges in the national network. On international circuits, however, each terminal has its own network timing.

All calls from data equipments are atomatically answered by the network. If the call cannot be set up immediately, the network will indicate the status of the call. For example, if the subscriber tries to call a data equipment which is switched to local mode, the network will inform him why the connection cannot be completed. The different status signals are listed in table 2.

Network structure

The network structure is described by means of an example showing how a subscriber equipment, a bank cash-point terminal in the Stockholm area, is connected.

The bank cash point can be considered as a DTE (Data Terminal Equipment), and it is connected to a DCE (Data Circuit terminating Equipment) via a cable containing about ten wires.

DCE is also placed on the subscriber's premises and constitutes the data network interface towards the subscriber terminal. The data network is fully synchronized from a data network exchange out to each individual DCE.

Two pairs of wires are run from the DCE on the subscriber's premises to the nearest telephone exchange. In the exchange MDF they are strapped to two pairs in a junction cable to the centre of Stockholm. In the MDF in the centre the pairs are strapped to a cable which goes to a data network concentrator, DCC, installed in the same building, where the wire pairs are connected to a modem. DCC contains one modem for each subscriber, selected to suit the line in question.

The concentrator consists of up to ten line modules, each for 50 subscribers, and is connected to the data network exchange, DSE, by means of between two and ten 64 kbit/s circuits.

When a subscriber makes a call, using a speed of, for example, 2400 bit/s, the concentrator has to set up a connection to a 2400 bit/s circuit on a 64 kbit/s line. In the case of a call to the subscriber the concentrator must connect through from a certain 2400 bit/s channel on the 64 kbit/s line to the subscriber in question.

The concentrator is controlled by the parent data network exchange. In the case of subscriber concentrations that are so small that a concentrator is not justified, the subscribers can be connected to an ordinary data multiplexor (DMX) or a remote data multiplexor (RMX), which in its turn is connected via a 64 kbit/s line to a data network exchange or a concentrator.

All lines connected to the data network exchanges are 64 kbit/s lines, and thus the exchanges only handle traffic between subscribers connected to the concentrators and multiplexors throughout the country. The data exchanges of the four Nordic countries are also connected together via 64 kbit/s circuits.

Table 3
Subscriber facilities in AXB 30

Direct call	All calls are connected to the same B-subscriber.
Selective direct calls	The subscriber can have direct calls to up to 8 B-subscribers.
Abbreviated address calling	Dialling is by means of two digits and is used towards a limited number of subscribers. Each A-subscriber has an individual list in the exchange.
Closed user group	Subscribers belonging to a closed user group are marked i the exchange so that calls can only be made to other members of the group. A subscriber can belong to more than one user group.
Outgoing calls barred	Intended for data equipments which are only to be used for incoming traffic and consequently should continuously be available to receive data.
Outgoing international calls barred	Each subscriber in the network has a marking which indicates either blocking or no blocking. A barred subscriber who tries to make an international call receives an indication that the call is not permitted.
Incoming calls barred	Intended for data equipment which is only to be used for outgoing calls and which need not receive incoming calls.
Incoming international calls barred	Each subscriber in the network has a marking which indicates either blocking or no blocking. Any attempt from abroad to call a barred subscriber will be unsuccessful.
Group number	Several subscribers of equal status use the same number.
Connect when free	All calls which come in when a subscriber is engaged are placed in a queue. Queuing calls are connected up as soon as the subscriber is free.
Redirection of calls	Calls are connected up to a certain other subscriber when the called subscriber cannot receive data, e.g. because of a breakdown.
Calling line identification	Before the setting-up of a call is completed, the called subscriber is informed of the number of the caller.
Called line identification	When a call is made the calling terminal operator can check that the network has registered the correct number, in order to ensure that confidential information shall not fall into the wrong hands.
Charge transfer calls	The calls are charged to the B-subscriber.
Charge advice	The system provides the paying subscriber with price information after each call.

Thus a data network of type AXB 30 consists of the following types of components:
- data network exchanges, DSE
- concentrators, DCC
- multiplexors, DMX or RMX
- subscriber equipments DCE.

Fig. 7 shows the network structure. It should be noted that subscribers near the data network exchanges are connected to concentrators or multiplexors in the exchange premises.

The bank which owns the terminal in the example above also has its central computer connected to the data network in a similar way, i.e. via a DCE and concentrator to the data network exchange. Each time a person carries out a transaction at the terminal (e.g. draws cash), the terminal calls the data network exchange. The exchange connects up to the data centre of the correct bank or to a cash terminal centre. Information regarding the bank is provided on the cash card that the customer uses. When the transaction has been completed the circuit is disconnected and the equipment is ready for the next transaction.

A technical description of the data network exchange, the concentrator and the multiplexors will be published in a later issue of Ericsson Review.

Subscriber facilities

The subscriber facilities offered by AXB 30 over and above the basic ones, such as full duplex, automatic status signals in accordance with table 2 and automatic answer, are summarized in table 3 on the next page. Further subscriber facilities will be added to the networks as required.

Charging

The charging in the public data network is based on information regarding each individual call. This informations is recorded in the data network exchange in accordance with two different principles:
- pulse charging with memory cells that correspond to the call meters in the telephone network
- toll ticketing with recording of the setting-up and disconnection times, calling and called subscriber, unsuccessful calls etc.

The use of pulse charging or toll ticketing is therefore dependent on how detailed the subscriber wants his invoices. The information is stored on magnetic tape for subsequent processing in a computer for invoicing.

In the normal case the invoice to the subscriber specifies, among other things:
- fees for subscription and additional facilities
- any non-recurrent fees for new subscriptions and new additional facilities
- traffic charges
- traffic-dependent charges for the use of certain additional facilities.

The subscriber can be given a complete specification of all outgoing calls during a certain period of time.

Calls are normally charged to the calling subscriber. However, it is possible to have calls within the country charged to the called subscriber, but this is an additional facility for which a subscription must be arranged.

The traffic charges vary depending on the transmission speed, duration and distance. A country is divided into charging zones and the distance between the zones affects the charges.

Operational quality and reliability

Compared with data transmission over the public telephone network and leased lines, the major advantages of the public data network are its greater operational reliability and faster fault localization and fault clearing. However, disturbances in the data transmission can occur even in the data network, and test routines and facilities for connecting in standby equipment must be available also in this network.

Service aims

Service aims were laid down in the specification, based on the requirements of the subscribers and the tech-

nical and economical resources. The service aims, which are listed below, refer to a fully built-out network:
- call attempts which are unsuccessful because of congestion in the network <0.5%
- bit error rate for a complete circuit between two subscriber terminals $<10^{-6}$.

The following construction requirements were set in order to reach the maximum availability in the network:
- high-quality components used throughout the network
- duplication of central equipments and the circuits between them
- automatic supervision of equipments and circuits
- changeover to standby units and alternative circuits in the case of faults
- automatic fault limiting and fault localization
- automatic routing of alarms and fault printouts to the correct operation and maintenance centre.

Supervision and fault detection
The equipment connected to the data network includes computers and terminals. The Administrations are responsible for the communication between the data circuit terminating equipments (DCEs). The subscriber or his supplier is responsible for any other equipments connected to the system.

The divided responsibility for maintenance has made it necessary to provide the network with functions that enable the subscriber and the Administration to determine quickly and reliably whether a fault is located in the network or in the subscriber's equipment.

The fault detection functions are designed so that faults which would mean down time for the subscriber are usually detected automatically, and an alarm is sent to the maintenance staff without any assistance from the subscriber.

The subscriber can set up a loop in the interface by means of a button on the DCE and thereby test his own equipment. The subscriber can also initiate testing of the DCE from the network. These function tests, together with the pilot lamps in DCE, help the subscriber to determine whether it is necessary to make a fault report to the Administration.

Automatic fault localization is obtained by means of loop connection of the data circuits at different points in the network between the subscriber and the data network exchange. The various loop connections are remotely controlled from the exchange.

As can be seen from the above, the possibility of automatically detecting

Fig. 8
Petrol station terminal

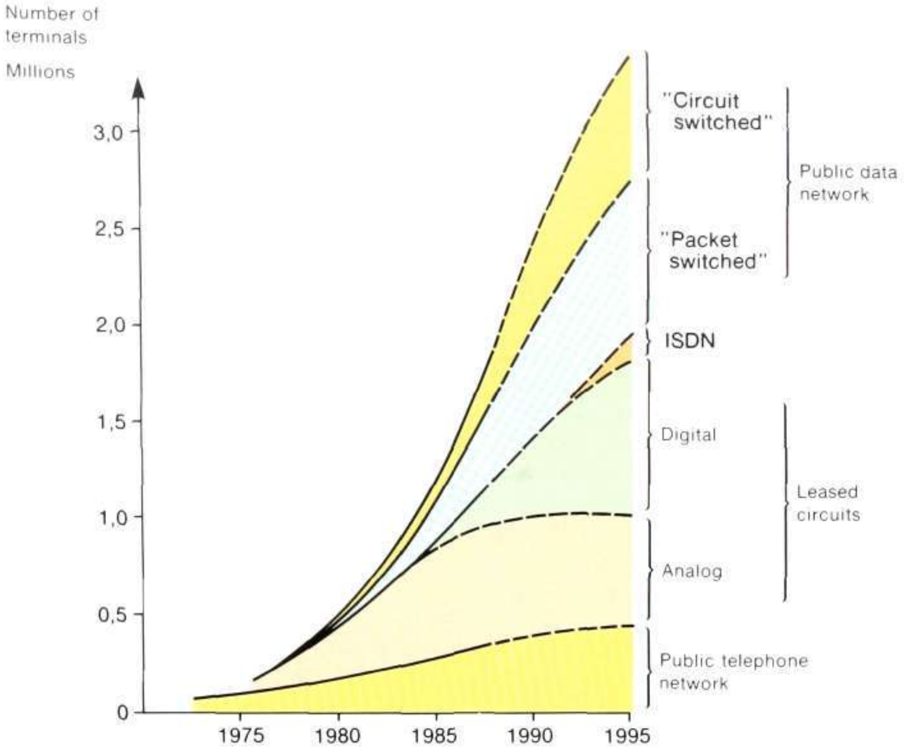


Fig. 9
A prognosis of the public data traffic in Western Europe, distributed over different transmission media

and analysing faults, so that fault-clearing measures can quickly be initiated, are very good. In addition the transmission quality and the traffic functions of the network are supervised, for instance by means of test traffic.

Protection against unauthorized access to data

The high degree of reliability of NPDN does not only apply to the operational quality but also to the protection afforded to the subscribers aginst unauthorized access to stored or transitted information.

The Administrations' responsibility comprises the transmission stage in the data system. This can be protected by providing the subscriber with different checking and blocking facilities. When equipment is connected into the data network the subscriber should consider what degree of security his system requires in the transmission stage, and on the basis of this decide which additional data protection facilities he requires.

NPDN offers various additional facilities which, among other things, offer the subscriber the possibility of creating "his own network". This means that the free switching possibilities that exist in a public network can be limited to the extent desired by the subscriber. This provides the same degree of protection and security as leased point-to-point circuits, and at the same time gives access to the other facilities and advantages of the data network.

The additional facilities described below are all intended to provide greater protection against unauthorized access to data:

- Closed user group enables a group of subscribers to form their own "network within the network". The setting up of connections to and from this network is prevented.
- Blocking of incoming or outgoing calls can be arranged for both national and international circuits.
- Identification of the calling number can in certain cases provide sufficient protection for a subscriber who does not otherwise want to restrict his possibilities of free traffic. The identification is carried out by the network, which means that the caller cannot change the information. The facility for identification of the called number also helps to increase security.
- Specified invoicing enables the subscriber to check what the data equipment has been used for.

The circuits are monitored in order to prevent unauthorized "listening in" or changing of the data being transmitted.

Future extension of NPDN

The plans for NPDN include extension to a considerably greater capacity than the 11 400 subscribers of the initial order.

The Administrations have ordered further equipment. At present data network exchanges are being installed in Århus, Gothenburg and Bergen. In addition the Oslo exchange is to be extended, a new data network exchange built in Malmö and a total of about 60 new concentrators installed. Fig. 9 shows how the number of data terminals connected to the telecommunication networks in Western Europe has grown up to 1980, and gives the estimated growth during the 1980s and beginning of the 1990s, distributed on different types of telecommunication networks. As can be seen, the growth is expected to be rapid.

Operational experience

The Nordic data network has been in trial operation since August 1980 and has successively been in full commercial service since August 1981. A number of Swedish commercial banks have their cash point terminals connected to the public data network. During September approximately 700 000 transactions were carried out from such terminals, about 300 in number.

After a few initial problems were cleared up the network has functioned entirely satisfactorily.

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Control and Supervision System for Railway Traffic

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Ericsson has developed and marketed centralized control and supervision systems for railways since the middle of the 1930s. The latest system, JZA 715, is a standardized system indended for Railway Administrations with varying requirements and rules. A modular program structure enables the system to be adapted for networks with different ambition levels and different geographical conditions.
The author gives a brief history of the need for and development of control and supervision systems, and describes the fuction and structure of the new system.

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The advantages of controlling widespread railway networks from one and the same place were realized quite early on. Such centralized control meant better utilization of resources, not only personnel and rolling stock but above all the capital-demanding ground structure, track network and other fixed installations. History shows several early examples of how one or two tracks in four-track railway lines could be closed down when centralized control of the railway traffic was introduced. Tre traffic capacity was maintained or even increased, and at the same time the cost of track maintenance was reduced considerably. It has also been possible, more recently, to avoid expensive building out of single-track lines to double track lines because of the increased capacity provided by centralized control. On lines with increasingly intensive traffic it has often been possible to manage the greater load without having to engage more staff.

At first the centralization consisted of control of switches and signals over individual lines. The foundation had thereby been laid for CTC (Centralized Traffic Control), which is a form of operation with its special rules and regulations, and not synonymous with remote control. A big step forward was taken when the first real remote control and supervision system was installed on New York Central Railroad in 1927. The route Stanley–Berwick in the state of Ohio, with 32 switches and 102 signals, was controlled via one single pair of wires.

Development by Ericsson

Ericsson developed its first control and supervision system in the 1930s. The system, which operated at a transmission speed of 10 bauds, used step-by-step selectors. It was put into operation in 1938 on the Stockholm-Saltsjön Railway, a route of 16 km, and comprised seven substation equipments.

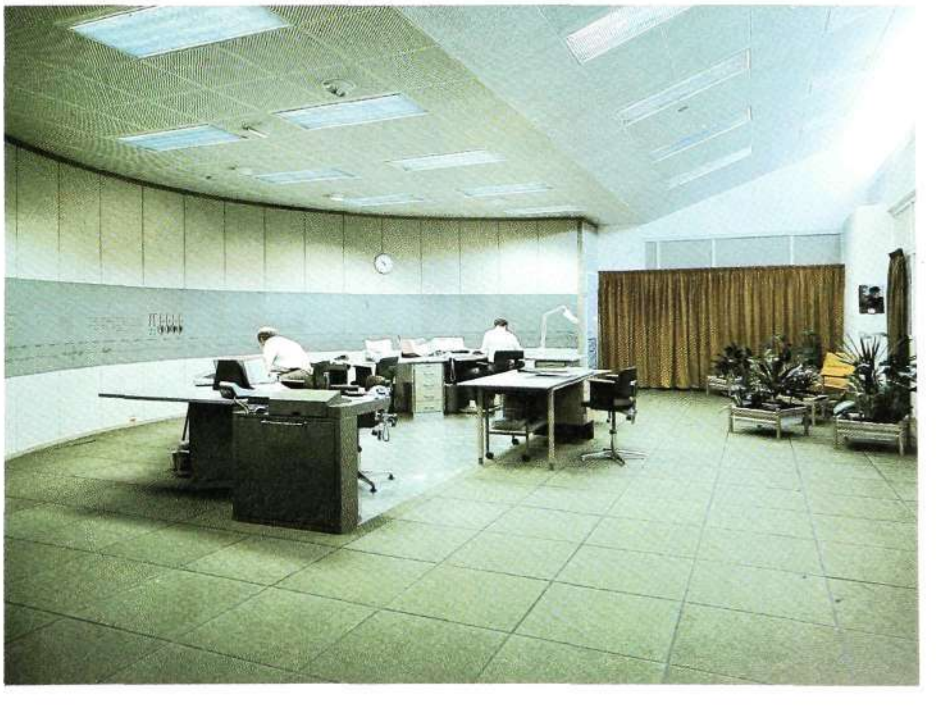


Fig. 1
Right from the start the Swedish State Railways planned centres for large geographical areas. The picture shows the CTC centre in Gothenburg

Fig. 2
To the right a key-set for controlling train traffic and to the left sets for different telephone systems

At the beginning of the 1950s the CTC systems with relays were developed which were to dominate the Ericsson production for two decades. These relay systems were the fastest in the market, since they operated at 25 bauds, and only the changes in the sensed states were transmitted.

Ericsson introduced key-set control already at this early stage. At all CTC centres control had previously been given by means of individual function buttons or switches mounted on an indication panel, which restricted the size of the controlled area. With the new system the operator could control the traffic while sitting back from the panel, thus getting a good view of a much larger area. The Swedish State Railways accepted key-set control at an early stage, and could therefore right from the start plan centres which covered large geographical areas, figs. 1 and 2.

Several electronic systems were designed during the 1960s. The first were experimental systems using germanium technology. These were followed by various systems containing discrete components in silicon technology, with an information transmission speed of 1000 bauds.

Modern systems

The now well-established system family JZA 700 was developed in 1970, when integrated circuit technology had been stabilized. The JZA 700 family consists of a number of printed board assemblies, each of which has a standardized interface towards the JZA 700 bus. These assemblies can be combined in different ways to give several system variants for different capacities and sizes, without any new design work being required. This means, for example, that there is no need to install unused physical capacity. The system contains many interesting technical designs, for example the JZA 700 bus, which without strapping or other programming determines the number of transmitted or received words merely through the number of assemblies inserted in the shelf.

The railway application, unlike normal data transmission with star-shaped networks, means that

- many terminals are placed in series along the railway lines
- there is often a shortage of wires in the cables
- the cables are often of poor quality and heavily loaded, which gives a low upper frequency limit (<2.7 kHz)
- the cable routes often suffer interference caused by longitudinal voltages induced as a result of the traction current.

An optimum design requires that the same number of wire pairs is used along the whole length of the route, and that the number is as low as possible. JZA 700 is designed for half duplex operation, with regeneration of the transmitted signal in each terminal, fig. 3. In addition the two line sides are galvanically separated and floating relative earth, which automatically gives great immunity against induced voltages.

Right from the start JZA 700 was designed with the control and transmission system integrated to form a common control and supervision system, which made the system very economical. It has been sold to many countries and is also manufactured by Ericsson subsidiaries in Australia and Italy.

Introduction of computers

During the late 1950s and early 1960s Ericsson worked on ways of simplifying the work of the operators and thus making even more efficient use of the control centres, i.e. heavier traffic, control of larger areas etc. Auxiliary systems based on relays were developed, such as train identification, train describer and destination tagging systems. The

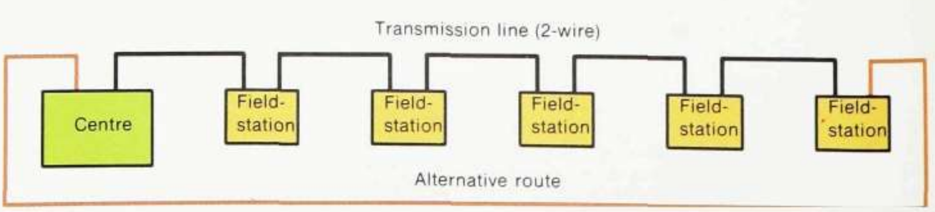


Fig. 3
Alternative circuits between a CTC centre and substations

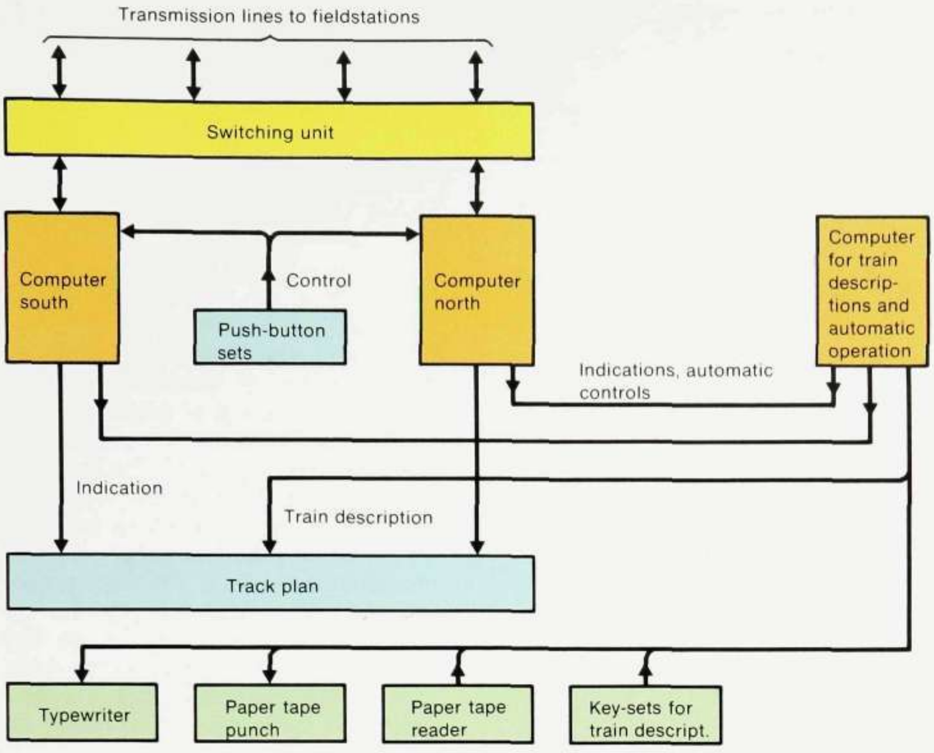
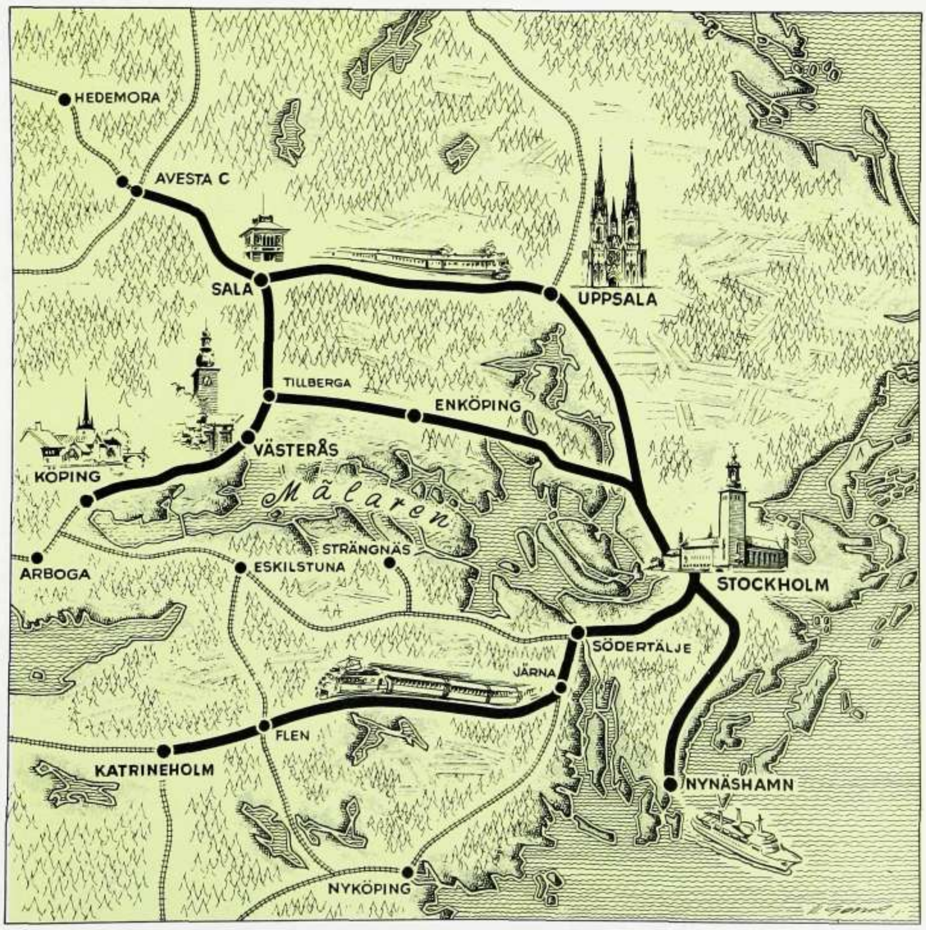


Fig. 4
Hardware configuration for the UAC 1605 systems

Fig. 5
The remotely controlled interlocking system in Stockholm was one of the first computer-controlled systems in the world



last system, for example, gave the different trains a destination tag, after which the control system guided the trains to the correct destinations by setting up appropriate routes at the correct moments. This system was developed in close collaboration with the Danish State Railways.

These types of systems tended to become increasingly voluminous. At the same time computer technology was becoming more practically applicable with the development of silicon components. There were as yet no usable minicomputers on the market, and Ericsson therefore developed its computer UAC 1605. It was intended particularly for traffic control and used parts from the Ericsson AKE systems.

During the latter part of the 1960s and the beginning of the 1970s a program system was developed for UAC 1605 having the following functions:
- transmission to approximately 100 terminals (substations) over four transmission lines, control and indication function with a key-set and panel (JZA 410/411)
- transfer, manual input, automatic input, remote input, search, display and remote display of train descriptions (JZF 20/21)
- automatic setting up of train routes, timetable comparison and platform sign control based on a timetable, using a traffic plan, time schedule and sign plan, which can be entered up to a year in advance (JZK 20/21).

Two computers are used for the first group of functions above, and a third computer is used for the other groups, fig. 4. The first two computers each serve one half of the geographical area. In case of a fault in one computer the other will automatically take over the whole area. The third computer contains only auxiliary functions and there is no standby. If a fault occurs the operation is taken over by the operators, who control the traffic manually and without train descriptions. Such computer systems were installed in two centres, one for controlling all train traffic in the Stockholm area, fig. 5, and one for controlling the local traffic on the S-railway in Copenhagen, Denmark. The Stockholm centre was one of the first computer-controlled systems in the world. The Copenhagen centre is one of the most advanced, with fully automatic operation. The traffic statistics from that centre are very good, with less than 3% of the trains delayed by more than two minutes.

In addition to the actual control systems Ericsson has also developed application aids in the form of systems for generating installation data and traffic plans.

Present-day systems

Prerequisites

A railway system has to meet certain traffic requirements, fig. 6.

The purpose of the control and supervision system is to help regulate the railway traffic so that the traffic requirements are met.

The railway traffic process includes the interlocking process. For safety and technical reasons the interlocking limits the traffic flow over and above the limitations set by the track network, rolling stock and regulations. For optimum traffic control it must be possible to predict the restrictive effect of the interlocking process. Access to information regarding the conditions for the interlocking process is therefore a prerequisite for such prediction, fig. 7.

As regards the fuction of JZA 715, the system can be considered as consisting of two main parts:

- the basic function, which controls the subordinate interlockings
- superior functions, which supervise and control the traffic process, and which therefore reduce the operators' work load considerably.

The earlier computer systems were controlled by means of tables in order to adapt them towards the actual track layouts of different railways, but extensive individual programming was still required for each Administration. Differences in traffic rules and other requirements meant considerable work, even if certain parts of previously prepared programs could be used. Hence, when the systems were to be modified for use with more modern computers it was decided to design a new program system which would be suitable for any Administration. Thus the new system also had to be table-controlled for adjustment to different traffic rules and other individual requirements. Additional programming should only be necessary for very special individual requirements.

The system should also have a modular structure, so that demands for different ambition levels could be met by building up a system using only the required standard function units, plus any specially developed units. Moreover, the interfacing towards different types of safety equipment by means of individually constructed relay sets in the substation equipments was to be replaced by table-controlled software in the central unit.

Discussions held with different Administrations and comparative field tests have shown that conventional panels can with advantage be replaced by colour visual display units, CVDUs, as long as these are used in an appropriate way.

JZA 715 includes two types of CVDUs for:

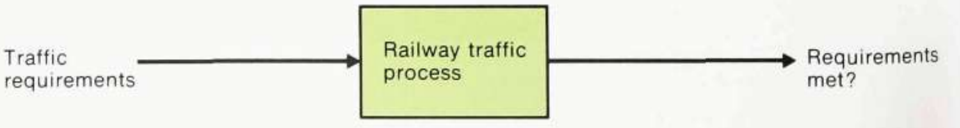


Fig. 6
A Railway system has to meet the traffic requirements

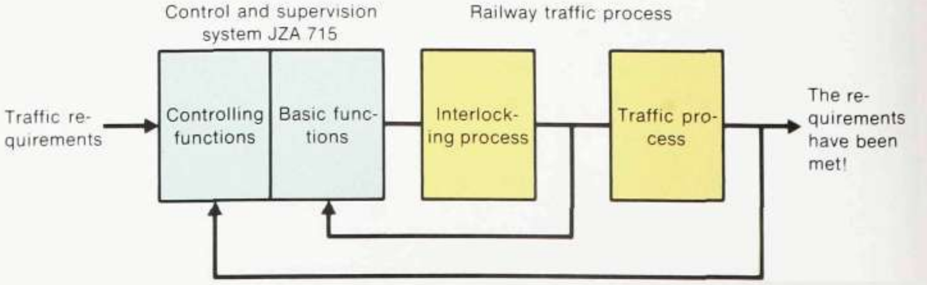
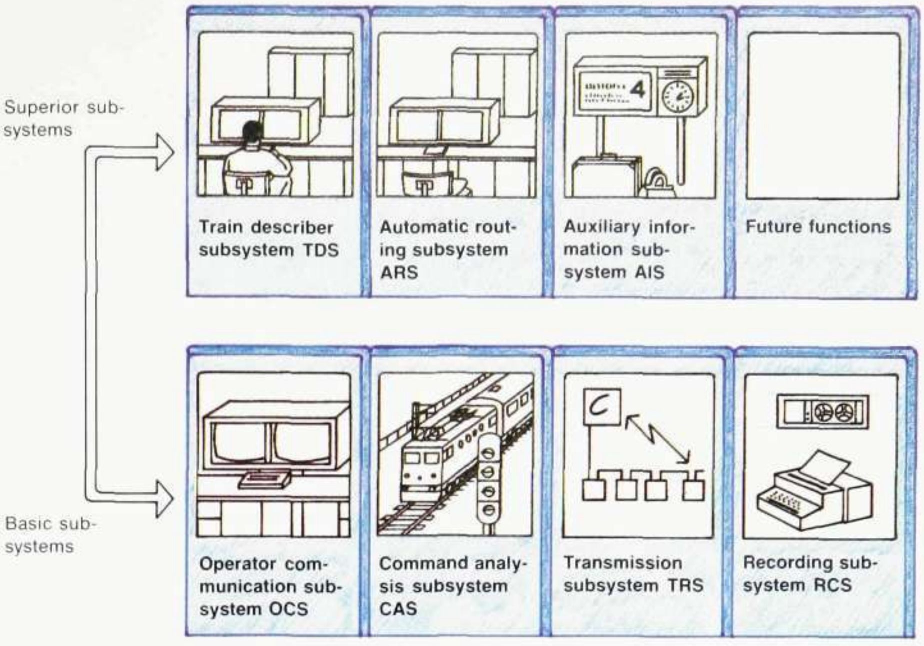


Fig. 7
JZA 715 consists of two main functional parts, which together regulate the traffic so that the requirements are met

Fig. 10
Subsystems in JZA 715



- Overview display, which gives an overall picture of the traffic situation, with little signalling information. It shows mainly the track layout and train positions, fig. 8.
- Detail or area display, which gives a more detailed information, showing signalling as well as traffic information, fig. 9.

The CVDUs can also give other types of information, such as alarm lists and route plans. When necessary the operator can be provided with a larger number of displays, fig. 15.

The controls are usually given from a keyboard similar to that of a typewriter. From this, each command is fed in in the form of a three-digit mnemonic code, followed by the number of the object. By meas of an auxiliary function the most common commands can also be isssued from a panel containing individual key-sets for the objects and for certain commands.

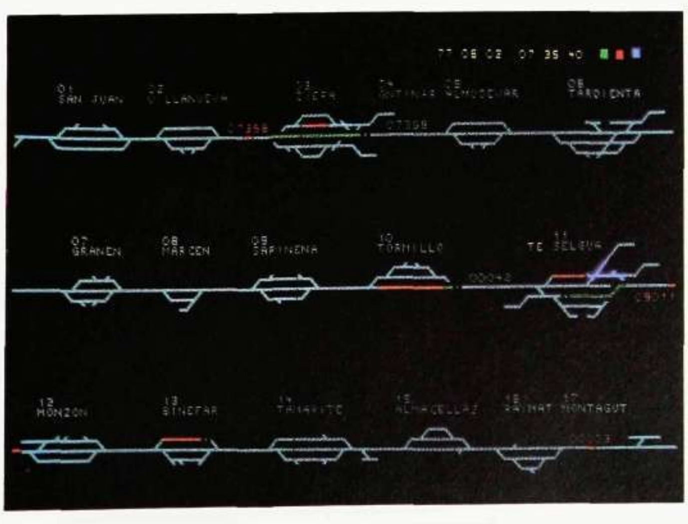
Basic fuctions

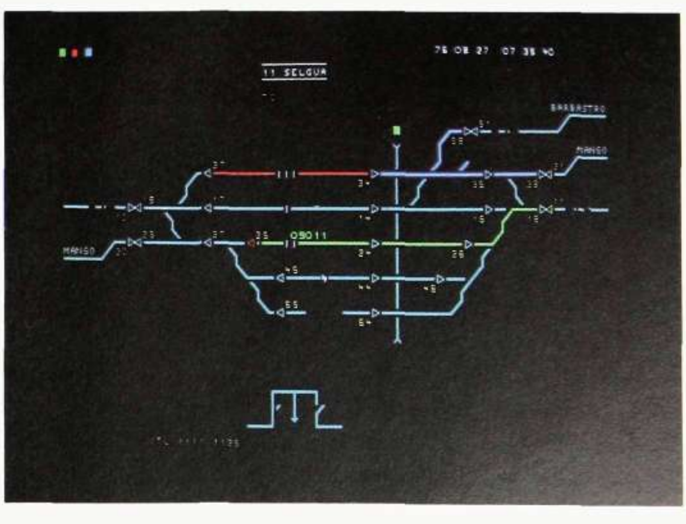
The basic functions are provided by four subsystems which together form the actual control system, fig. 10:

- Operator communication subsystem (OCS), which handles the man-machine communication, i.e. all manual input of commands and all display of information to the operators.
- Control and indication analysis subsystem (CAS), which analyses outgoing commands and incoming indications. It includes static and dynamic checking of the feasibility of the commands, a breakdown of the commands and logic conversion of the incoming "raw" indications into such a form that it can be displayed.
- Transmission subsystem (TRS), which handles the transmission of messages to and from the interlockings and provides the interface towards them. This subsystem, together with CAS, provides the necessary logic matching towards the different types of interlocking systems that are encountered.
- Recording subsystem (RCS), which records occurrences in the controlled process that concern the traffic and also the maintenance. The sub-

Fig. 8
Key display–the railway traffic in general (left)

Fig. 9
Sub-area display–detailed information for the operator





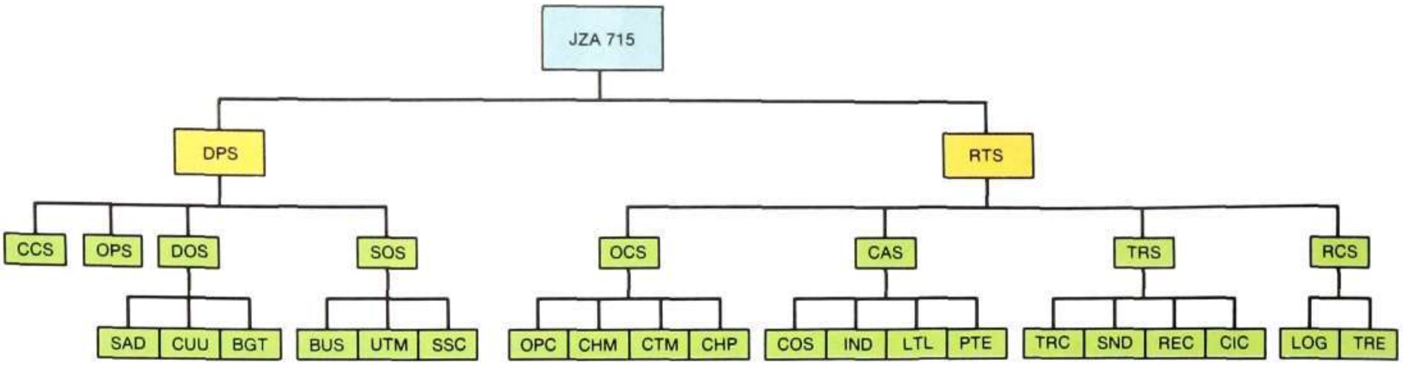


Fig. 11
The structure of JZA 715, the basic version

system also prints reports of preselected types of occurrences during a specified period of time.

In addition to the operational functions (RTS) there are a number of functions that are necessary for operating the system itself. These are provided by subsystems in the data processing system (DPS), fig. 11:
- Computer subsystem (CCS), which consists of the necessary hardware.
- Operating subsystem (OPS), which consists of the standardized software associated with CCS.
- Dual oparation subsystem (DOS), which consists of a specially developed standby system with a continuously updated standby computer.
- System operation subsystem (SOS), which consists of a specially developed communication system, for communication between programs within a system as well as between systems that control different geographical areas.

All the basic functions described above are included in the basic version of JZA 715, which can handle all ordinary control and supervisory functions for the controlled interlockings, regardless of type. This applies both when JZA 715 is used as the control system for a large, local computer-controlled or conventional interlocking, and when it works as the centre in a remotely controlled area with many subordinate interlockings, or both, fig. 12.

JZA 715 can also be used as a modern centre in an older, remote control system. The new centre need not necessarily be situated in the same place as the old one.

Controlling functions

The basic functions can be supplemented by several functions provided by superior subsystems, namely:
- Train describer subsystem (TDS), which provides central input, transfer and display of the descriptions of the supervised trains, and the possibility of remote input and remote indication of train descriptions, fig. 13. The last two facilities, which are usually used for communication to and from manually controlled border station fringe boxes, present the train descriptions to the operators just as if they were coming from a local system. TDS also includes facilities for tracing a certain train, listing all trains and automatic input of train descriptions.
- Automatic train routing subsystem (ARS), which makes possible automatic setting of train routes on the basis of train descriptions and the time table, and with the possibility of taking into consideration different conditions such as the location of other train movements. The system also includes a simpler, programmable automatic function intended primarily for installations that lack train descriptions.
- Auxiliaryinformationsubsystem(AIS),

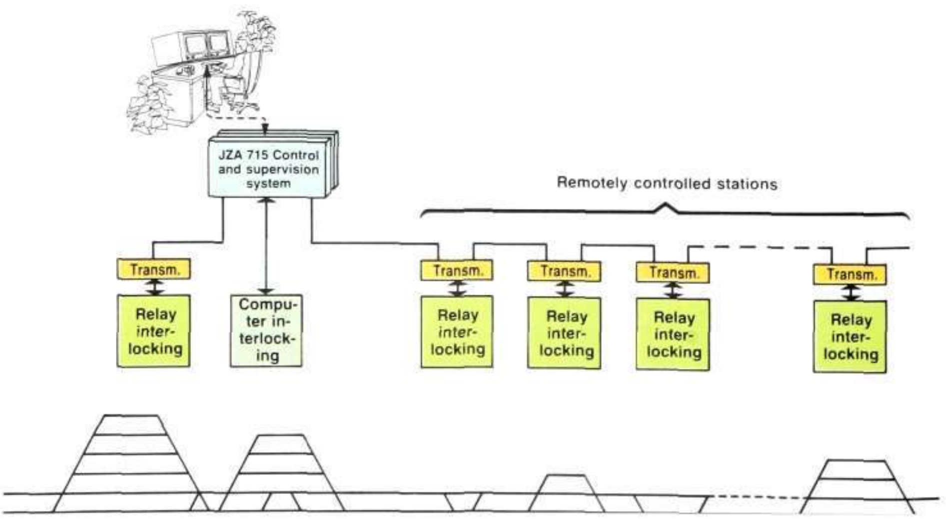
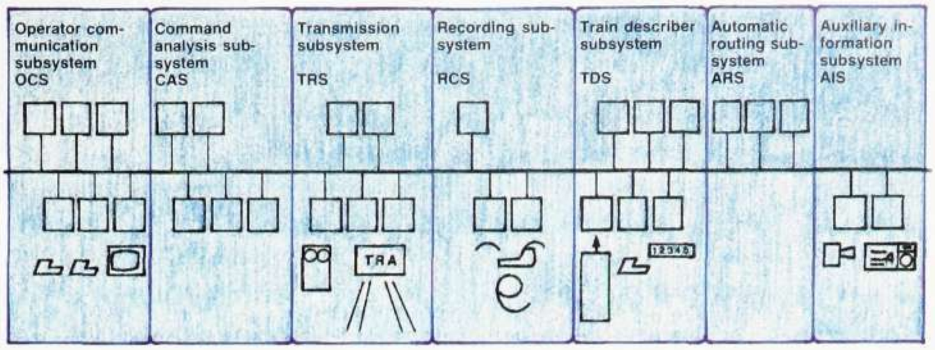


Fig. 12
Different controlled interlockings

Fig. 14
Bus-oriented program system



which provides automatic control of platform signs on the basis of train descriptions, automatic timetable checks when trains pass by, with reporting of any deviations, and display of selected timetable information. The control of platform signs, for example, means that the system automatically provides correct information to connected local systems for platform displays, regardless of any delays or changes in the order of trains. Simple sign systems can be controlled directly. AIS also gives an automatic blanking signal when the train in question has left the platform.

Most of the controlling functions are based on the train running plan (timetable). Temporary alterations may have to be made in the case of severe traffic disturbances. Such alterations are made with the aid of an editing function (DTG). The alterations are automatically erased when they have been used.

Realization

All functions are realized in subsystems which consist of a number of function blocks. These in their turn consist of a number of programs. (Of course, an installation need not be equipped with blocks that are not used.) The blocks in the system communicate with each other via the program data bus (BUS), a function in SOS, fig. 14. This communication system means that new blocks can easily get access to the process information that passes through the system.

The geographical aspects and interlocking characteristics of the controlled area, as well as the current states and characteristics of the objects in the area, have been compiled to form a common data base, "the geographical field", which is used for several functions. Yard objects and interlocking characteristics are represented in this field by different modules, which are geographically connected to each other on the basis of the track configuration. The system contains some twenty different types of modules, fig. 16. Areas with interlocking systems which are not in accordance with the geographical interlocking method are also represented in this way in JZA 715. Inconsistencies in existing interlockings have sometimes been discovered during the site adaptation of JZA 715, thanks to this stringent representation method!

Fig. 15
A control position with four colour video displays and one black and white display for information in the form of text

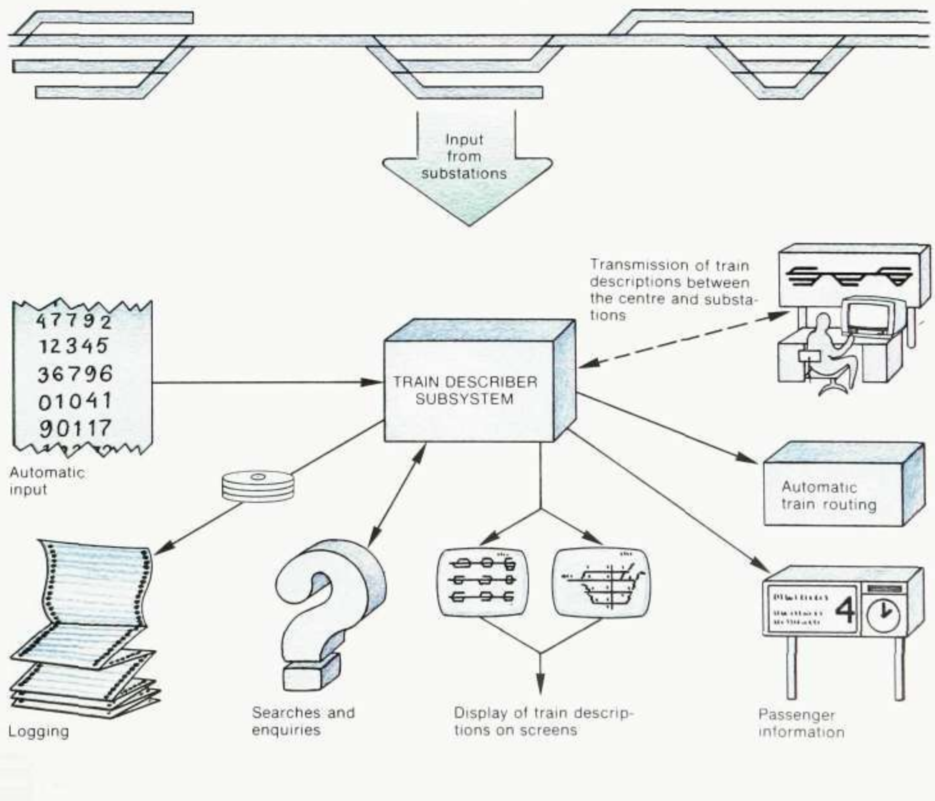


Fig. 13
Train describer subsystem, TDS

Fig. 16
The geographical features and interlocking characteristics of the controlled area are represented within JZA 715 by the geographical field

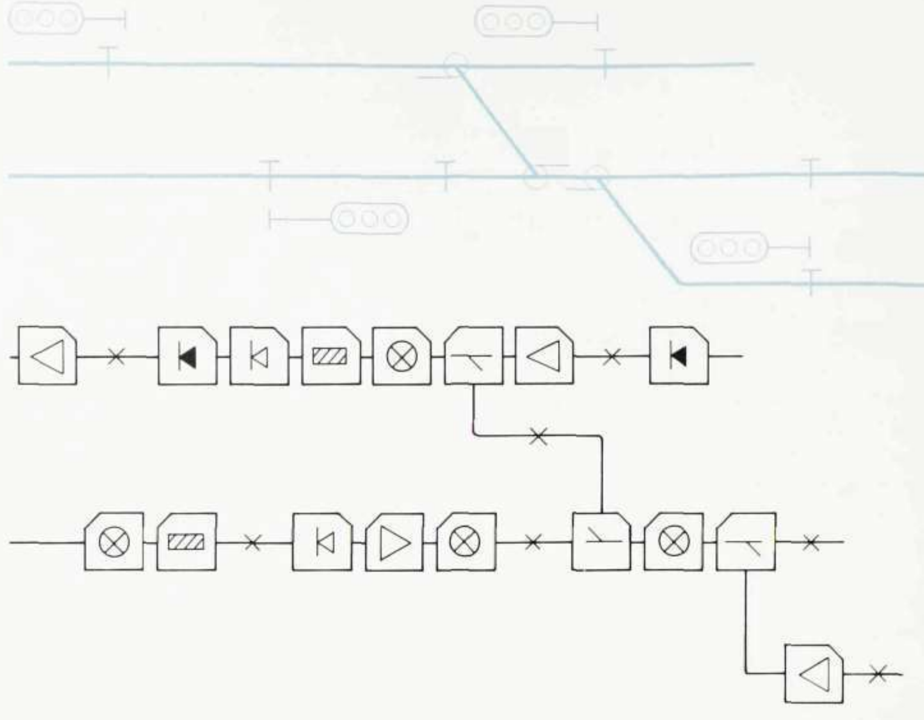
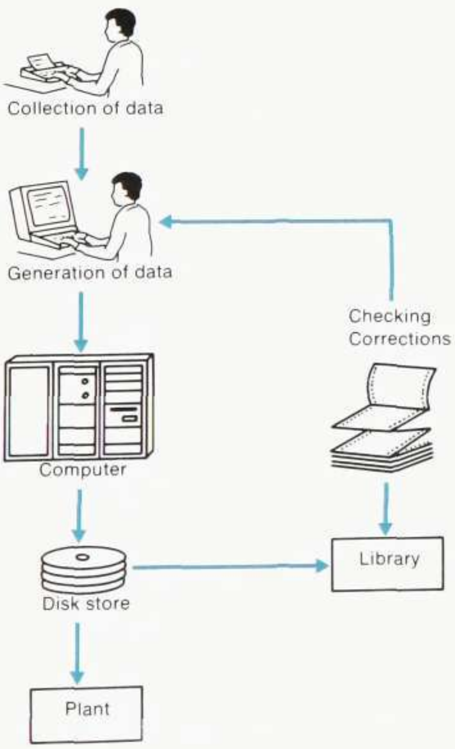


Fig. 17
Project planning aid, GEN 715



Project planning

A special data generating system has been developed which simplifies considerably the project planning of systems, fig. 17. The generating system produces the majority of the necessary individual system data, which includes the display pictures and geographical field. The installation testing is also simplified, since many errors in the input data are detected during the automatic checks carried out during the project planning stage.

A special train plan generating system has also been developed for generating train plans (route plans, time plans, sign plans) for the superior subsystems. It is intended both for new installations and for modifications. The Administrations themselves will also be able to use it, for example in connection with a change of timetable. The basic information used for new installations consists of the Administration's normal basic data for the preparation of timetables. It is fed in, either manually or automatically from the Administration's ADP system, and is then supplemented by additional information, which is entered interactively.

Development tendencies for the control and supervision systems

Present-day JZA 715 systems are already equipped with functions that enable adjacent installations to exchange certain information about train descriptions, delays etc. concerning the trains that pass the boundary. It is expected that in future this information will need to be extended, for example by

- enquiries regarding delayed trains which have not yet reached the system's own area
- transmission of temporary timetables for special trains.

The systems may also need to act as a digital information collector and provide other controlling systems, such as a locomotive dispatching system, goods waggon administration system, timetable system and statistics system, with the real-time information concerning the traffic process that the control and supervision system already contains.

The control and supervision centres are usually placed at large railway junctions. Each JZA 715 system contains a geographical representation of the track network, with the correct distances and all other information concerning the traffic in the area supervised from the junction. It seems natural that in future the centres will be connected together and form a decentralized system. This would avoid the vulnerability of central controlling supersystems. In the case of a fault at a junction, whether it is in the equipment or in the track, the railway traffic itself could find alternative routes.

The linkage envisaged above could in the future depending on the amount of information and processing required, be arranged either directly between different JZA 715 systems or indirectly, by means of a decentralized superior system, such as an information system. Whichever the alternative it is likely to need a data network for communication. Such data networks could either be public or exclusive to the Railway

Table 1
Computer-controlled CTC centres, in operation or on order

Place	Put in operation	Number of objects	Computers, number, type	System type
Stockholm, Sweden	1971	2350	3×UAC 1605	JZA 410
Copenhagen, Denmark	1972	1450	3×UAC 1605	JZA 411
Gothenburg, Sweden	1978	560	2×PDP 11/05	JZM 750
Oslo S, Norway	1979	1200	2×PDP 11/34	JZA 715
Arlöv, (Malmö), Sweden	1981	397	2×PDP 11/34	JZM 750
Melbourne, Australia	1982	2950	4×PDP 11/70	JZA 715
Doboj, Jugoslavia		1000	2×PDP 11/44	JZA 715
Olskroken, (Göteborg), Sweden		1200	2×PDP 11/44	JZA 715
Broadmedows, Australia		1250	2×PDP 11/70	JZA 715
Barcelona, Spain		400	2×PDP 11/34	JZA 715
Oslo F, Norway		1000	2×PDP 11/44	JZA 715

The junctions may then be equipped with different algorithms for optimizing the resources, such as:
- optimizing and allocating the transport route
- reorganizing the traffic during work on the track, derailments and other disturbances.

The energy crisis is likely to lead to more intensive railway traffic, which will mean greater demands on the operators to make correct decisions very quickly. In order to prevent the work load from becoming impossible to master, the control centres must in future be provided with auxiliary functions, for example:
- passing and by-passing optimization algorithms, which gives the operator the necessary data to show the consequences of shifting the passing of trains to alternative stations. The operator then chooses one of the alternatives by making temporary changes in the train running plan.
- optimization of the choice of track, which provides the operator with an alternative track arrangement in the case of disturbances. The operator can then carry out the alternative plan by changing the destination labelling of the train plans concerned.

Functions can also be added which enable these traffic changes to be carried out automatically as soon as the operator has chosen an alternative.

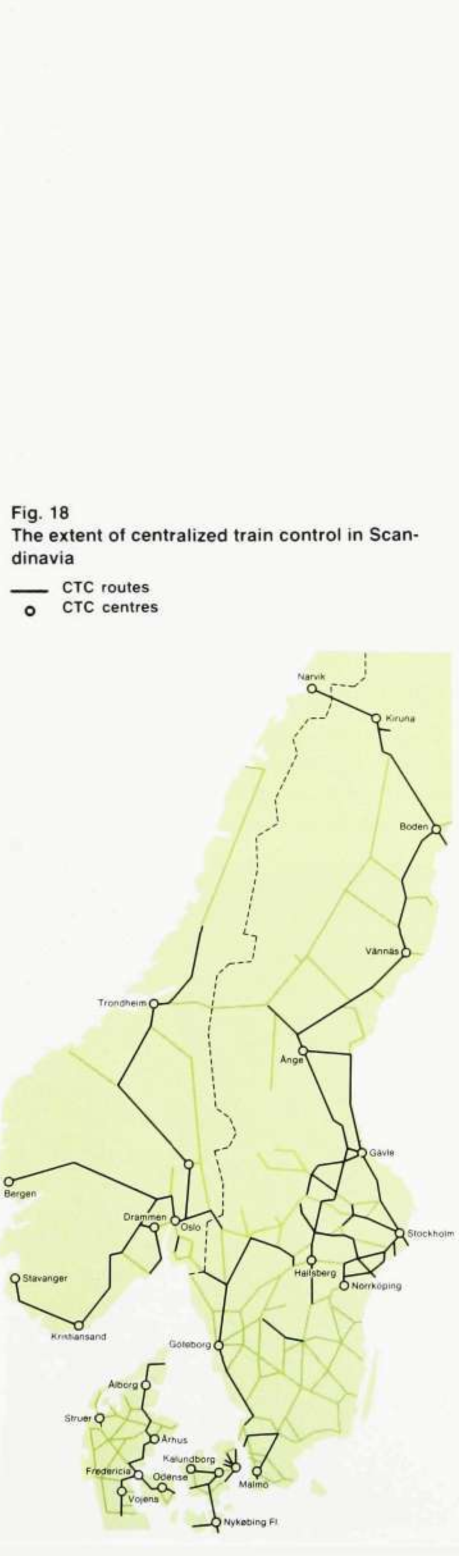
Summary

Centralized control gives better utilization of resources, as regards personnel as well as other resources. It is therefore possible to increase the capacity considerably, with only the limited investment for a control and supervisory system. Large centres with heavy traffic can be regulated with the aid of controlling functions to ensure a reasonably uniform traffic intensity. System JZA 715 contains a whole range of functions for controlling railway traffic. Such functions have previously only been available in separate systems, which have often been manufactured by different manufacturers and realized in older technology.

Up to now Ericsson has delivered equipment for 64 control and supervision centres for 1459 substations along 10 419 track kilometres. Fig. 18 shows the parts of the railway network in Scandinavia that are equipped with centralized train control. Table 1 shows the computer-controlled centres which have been supplied by or ordered from Ericsson.

Fig. 18
The extent of centralized train control in Scandinavia

—— CTC routes
o CTC centres



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ERICSSON SUNWIND

John Åkerlund

Ericsson has developed a power supply system that uses solar and wind power as energy sources. It is intended to be used as a source of power for remotely-located telecommunications equipment, and to meet the stringent reliability requirements imposed by Telephone Administrations. The system contains a maintenance-free wind generator, solar cell panels, a mini-diesel generator set, specifically designed for intermittent operation, batteries, and microprocessor-controlled power processing equipment for optimal power output control. The system is also well-suited for powering lighthouses and other navigational aids, as well as monitoring systems and meteorological reporting stations.

UDC 621.311.6
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There is a growing demand for telecommunication services in small rural communities. The high costs involved in arranging power supplies for remote radio link repeaters, just to take one example, has, up to now, often prevented telephone network growth in such areas.

ERICSSON SUNWIND, fig. 1, is a power supply system which can supply remotely located telecommunications equipment with up to 500 W power for 24 or 48 V d.c. The most important characteristics of the system are a high degree of reliability and very low maintenance requirements.

ERICSSON SUNWIND is a combined solar and wind power system consisting of solar cell panels, a wind generator, control equipment, batteries and a mini-diesel generator. The combination uses the complementary nature of sun and wind to smooth out the variations in the power supply from the separate sources. A lower level of the peak power installed can be selected as compared with a non-combined system. Also the battery can be smaller. The amount of solar and wind energy available varies continually in an irregular pattern. There can be major deviations from year to year and between locations in the same area. When there is an abnormal shortage of solar and wind energy the mini-diesel can be started up to charge the plant battery. The availability of this reserve reduces the need for exact meteorological data, making it easier to design the system.

Depending on the power requirements, as well as meteorological and geographical conditions, combinations of the ERICSSON SUNWIND system components can be selected. The wind turbine, solar cells and the mini-diesel can also be used by themselves or in pairs. Solar cells and the mini-diesel are frequently selected where there is a low power requirement in an area with a high level of solar radiation.

The wind generator

A wind generator which is to supply power to a telecommunications system must have at least the same high degree of reliability as the other system compo-

Fig. 1
ERICSSON SUNWIND
The solar cells are mounted under the wind generator

JOHN ÅKERLUND
Power Supply Department
Telefonaktiebolaget LM Ericsson

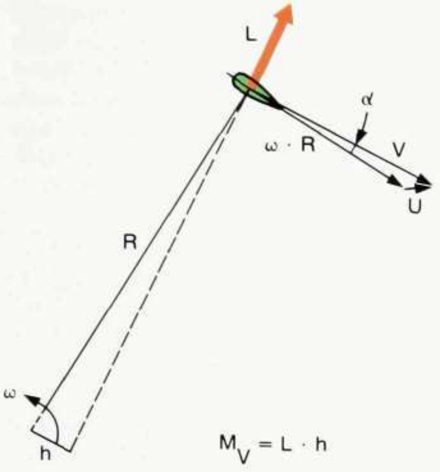


Fig. 3
How the Darrieus turbine functions
The wing section is actuated by an air flow V. It is the resultant of the wind velocity U and the tangential speed ω R. V is displaced relative to ω R at an angle of attack α. The air flow with the speed vector V generates air power L perpendicular to V. As V has the angle of attack α, L will have a moment arm h through the turbine centre. With the moment arm h the air power L generates the driving torque $M_v = L\,h$

nents. Ericsson, in cooperation with the Aerospace Division of SAAB-SCANIA has developed a wind generator which meets the following stringent conditions. It

- is maintenance-free
- can be mounted on top of radio link masts and has at least the same mechanical strength to withstand high wind velocities as the mast and the parabolic antennas
- has high efficiency at low wind speeds.

The wind turbine

The wind turbine is of the Darrieus type with straight, vertical airfoil blade sections, fig. 2. It has a vertical axis which makes it independent of the wind direction. The turbine starts when the wind velocity reaches a value of about 3 m/s. Good start-up characteristics have been obtained by means of a Savonius turbine having the same axis as the Darrieus turbine.

From the mechanical and strength point of view the turbine has been designed for wind velocities of up to 50 m/s with a margin to withstand gusts in excess of this value. The heavy-duty steel hub is mounted on a generously proportioned bearing. The arms and turbine blades are manufactured from extruded aluminium sections; the Savonius turbine is made of aluminium plate.

The aerodynamic air force L is formed by the air flow around the wing section, fig. 3, and perpendicular to it. The rotation speed together with the wind velocity causes the air force L to produce the moment arm h through the centre of the turbine. The driving torque will be $M_v = L\,h$.

In order to limit the rpm at high wind speeds the turbine has been equipped with a maintenance-free, highly-reliable mechanism which turns the blades so that they stall and lose their thrust. When the stall occurs the air force L is reduced, and the braking force D_s increases, fig. 4. Together with the arm radius R as a moment arm, D_s forms a braking torque $M_b = D_s\,R$, which is so powerful that the turbine is slowed effetively down to a constant maximum speed of about 300 rpm. A patent has been obtained for this design.

Because the turbine has a vertical axis the generator can be placed under the turbine attachment plate inside the

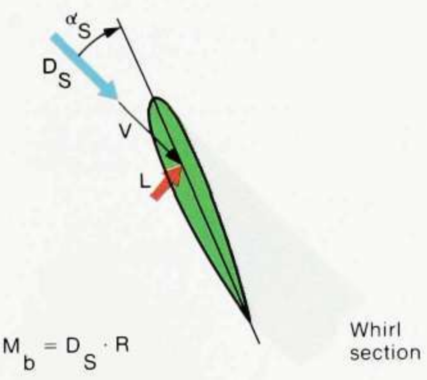


Fig. 4
At high wind velocities the turbine blades are twisted so that they stall. The resulting air power L is reduced and a resistive force D_s increases. The braking moment will be $M_b = D_s\,R$

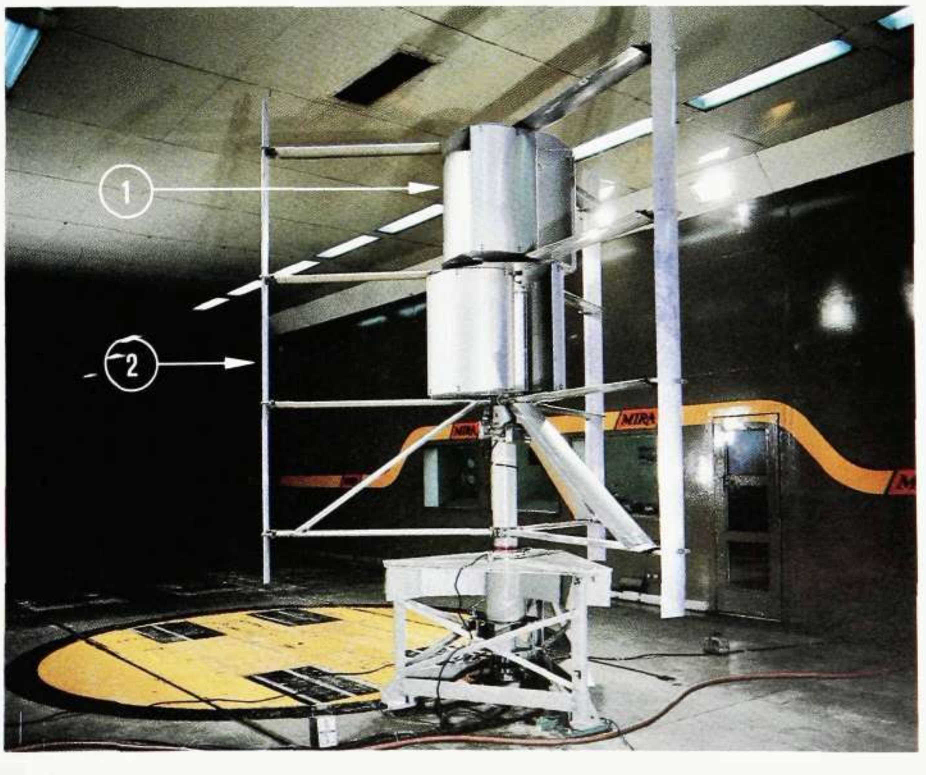
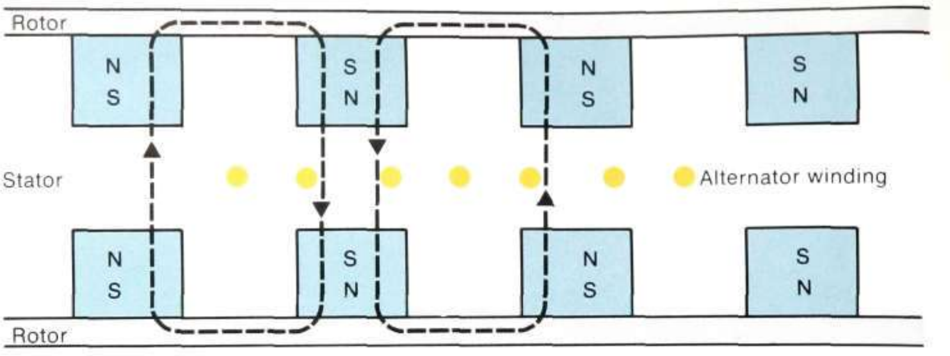


Fig. 2
The wind generator has an estimated average power output at 5 m/s of 150 W. The maximum power output is 900 W for wind velocities in excess of 10 m/s

① Savonius turbine
② Darrieus turbine

Fig. 5
The alternator is of the "pancake" type with 56 pairs of permanent magnets placed along the periphery. The stator winding is encased in a layer of composite material placed between the two rotor discs which are rotated by the turbine shaft



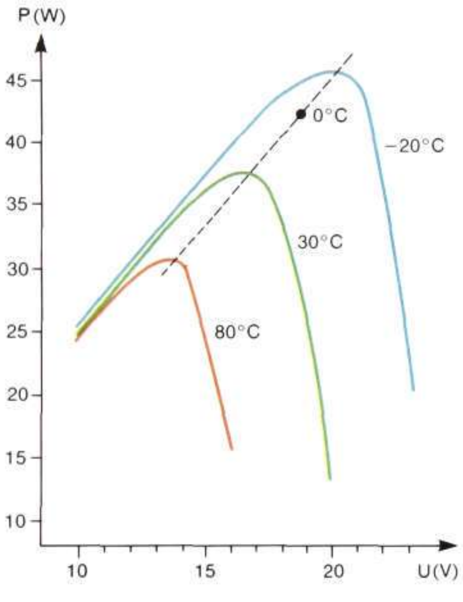


Fig. 7
The output from a 12 V solar cell panel with constant solar radiation of 1000 W/m² for various panel temperatures

mast. No slip rings are required to transfer the electrical power and the turbine axis bearing is simple compared with that of a conventional wind generator. The annual inspection can be carried out from inside the radio link mast.

The alternator

The alternator is a new type specially designed for wind power application. It has a multi-pole design so that it can be connected directly, without any gearbox, to the slowly rotating turbine. It is a brushless, three-phase, permanently magnetized alternator and does not suffer from magnetization losses. The windings are located in a self-supporting stator of iron-free composite material. As the windings do not have iron cores the permanent magnets cannot lock the wind turbine during start-up. This is of great importance as the wind generator must function at the most frequently occurring low wind velocities in order to produce the maximum amount of energy. Only the bearing friction in the turbine and generator needs to be overcome during start-up. Fig. 5 gives a general idea of how the generator works.

Solar cells

Ericsson examines the solar cells which are available on the market. Tests are carried out as normal component testing with the emphasis on examining the quality of encapsulation. Additional long-term system tests are carried out with solar cells, electronic components and batteries. In this way a considerable amount of knowledge concerning solar cells has been accumulated in the Company. It is important to have system know-how about the component parts of the solar and wind systems in order to be able to combine them in an optimal way.

The results from these solar cell tests form the basis on which suppliers are selected and this also guarantees that the solar cell panels delivered by Ericson are of the highest quality.

In ERICSSON SUNWIND the solar cell panels can either be connected directly to the battery or via d.c./d.c. converters in the power processing equipment, fig. 10. If the solar cell panels are connected via the d.c./d.c. converters their output voltage can be controlled and the power output maximized. This also avoids a voltage increase across the battery.

The solar cells have the characteristic that voltage and peak power increase as the temperature decreases. The solar cell panels are designed with such a large number of cells in series that the

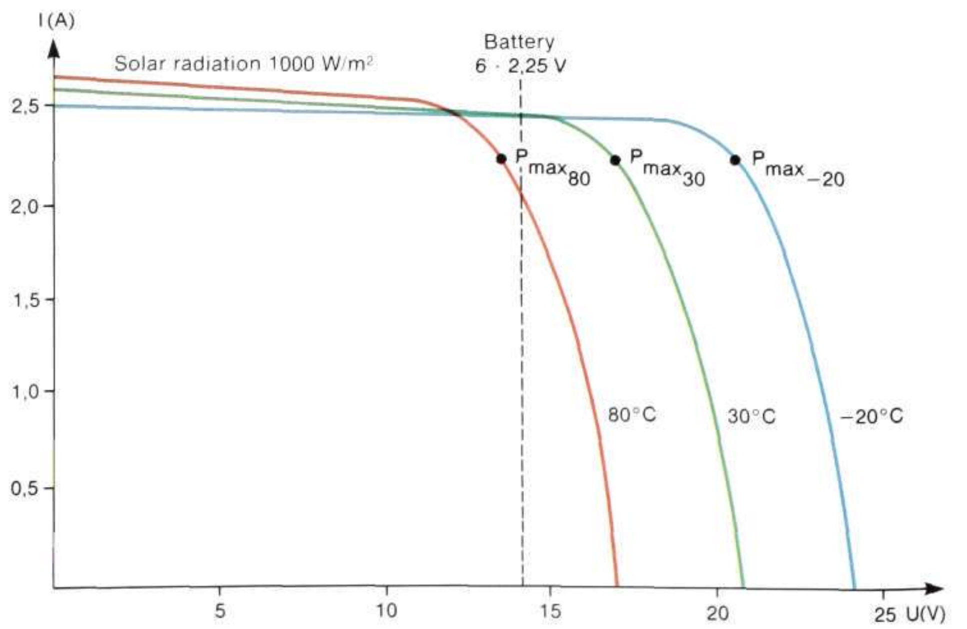


Fig. 6
The figure shows the current-voltage characteristic for a 12 V solar cell panel at different temperatures. The verical line marks the voltage necessary to charge a battery with 6 cells to 2.25 V per cell

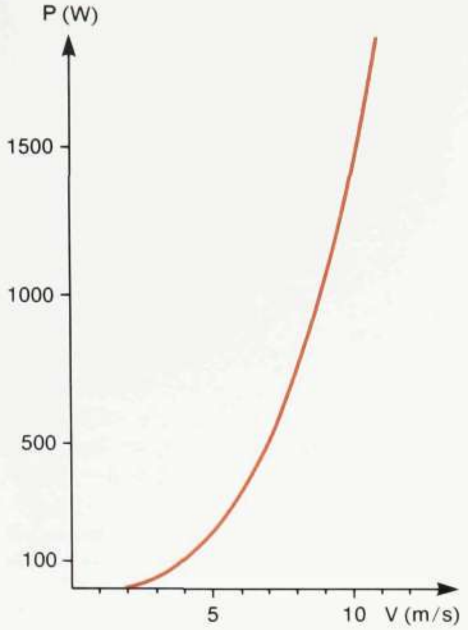


Fig. 8
The wind energy power increases by the cube of the wind speed

$P = k A \eta_T V^3$
P the turbine shaft power for various wind velocities
k constant which varies in particular with the density of the atmosphere
A the frontal area of the turbine against the wind
η_T turbine efficiency
V wind velocity

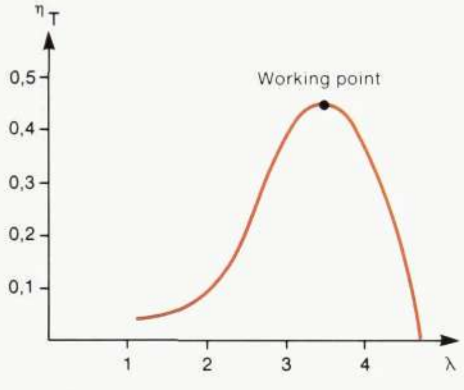


Fig. 9
Efficiency of a triple-bladed Darrieus turbine

$\lambda = \frac{R\omega}{V}$
λ ratio of blade speed to wind speed
R turbine radius
ω rotation speed (angle velocity)
V wind velocity

voltage at the maximum panel temperature, ~ 80°C, will be the same as the charging voltage for the battery, fig. 6. With an average temperature of 0°C during the winter it will then be possible, using d.c./d.c. converters, to increase the average power output by at least 25%, fig. 7.

The solar cell panels usually have 12 V nominal output voltage and about 30 W nominal peak power at 80°C. In a power supply system using 48 V system voltage, four 12 V panels are connected in series. The required peak power for the plant is obtained by connecting a number of these panel groups in parallel.

During the cold and dark part of the year the maximum power of the solar cells increases when it is most needed and compensates for the reduced solar radiation. A power-maximizing control system smoothes out seasonal variations; this means that fewer solar cell panels and lower battery capacity are required.

In small systems in hot climates the solar cells can be connected directly to the battery. However, power-maximizing control equipment is motivated in large systems and in cold climates due to savings in the number of solar cell panels and the size of the battery.

The power processing equipment

The power processing equipment is intended to maximize power output and regulate the charging voltage from the wind generator. It is used also to maximize power output and regulate the charging voltage from the solar cells, as has been previously described.

Wind energy varies greatly with the wind velocity, fig. 8; it is important that the power output from the wind generator carefully follows these variations, so that maximum power is always obtained.

The aerodynamic efficiency of the wind turbine is also dependent on the relationship of the rpm of the turbine to the wind velocity, fig. 9. The turbine is at its most efficient when this relationship is about 3.5.

The power processing system regulates the power output from the turbine so that the relationship is maintained regardless of variations in the wind velocity. This is especially important at the lower wind speeds which most often occur.

The power processing equipment, fig. 10 and 11, consists of a rectifier, a num-

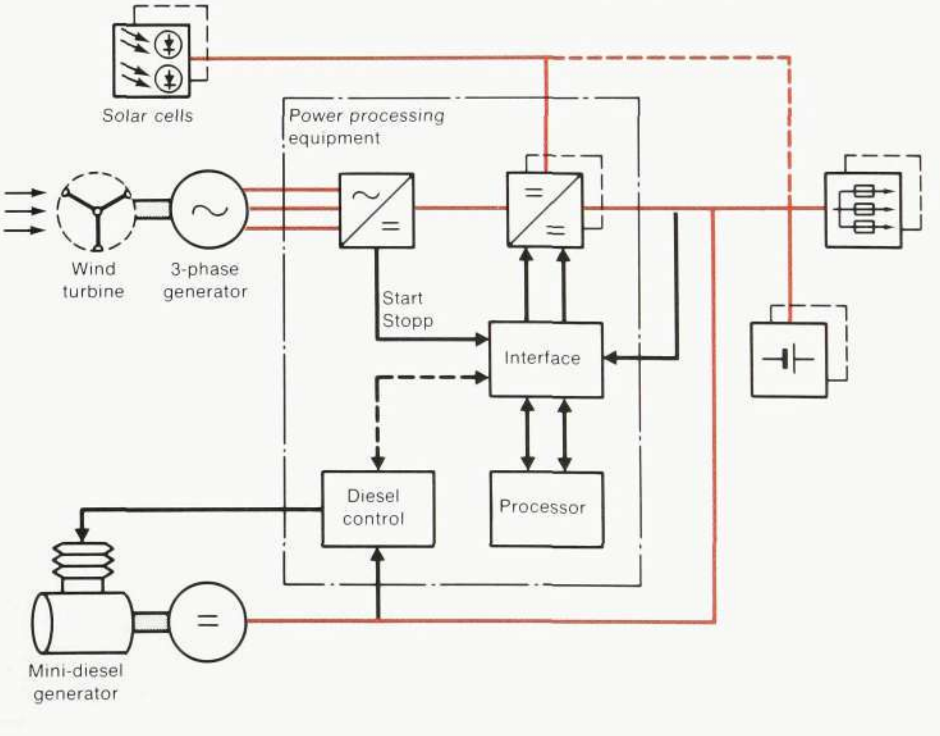


Fig. 10
Block diagram of ERICSSON SUNWIND

Fig. 11
Control equipment for the wind generator

ber of d.c./d.c. converters and a microprocessor with interface circuits. The diagram also shows a mini-diesel generator set which will be described later. Its control system is a stand-alone unit but it can be connected to the main system for interchange of information.

The three-phase, unregulated rectifier has an output voltage which varies between 50 and 375 V d.c. It also contains a voltage monitor and a frequency monitor. The voltage monitor measures the generator idle voltage and is used to decide when, from the energy point of view, it would be profitable to start up the system. The generator idle voltage is a direct measure of the wind velocity and the available wind energy. The frequency monitor records the alternating voltage frequency which is the measure of the rotational speed and is used in the microprocessor to analyse the variations in available generated power.

The d.c./d.c. converters are choppers. They convert the very varied voltage from the rectifier to a constant voltage for charging the system battery. The microprocessor continually calculates the maximum available power from the varying energy flows of the wind and sun and controls the d.c./d.c. converters. This can take place in two ways, by means of voltage control or power control.

Voltage control is used when the system battery is fully charged. The battery voltage is then held constant, the charging current is minimal and the energy sources are only loaded with the power which the load consumes.

Power control is used for most of the time. It is utilized when the battery is not fully charged. The power control value is continually being calculated by the microprocessor and information about the maximum available power value is constantly sent to the d.c./d.c. converters. These then regulate the power output from the wind and solar energy sources so that they work most efficiently.

The capacity of the microprocessor is not wholly utilized for managing the control equipment. It can be used for other duties such as operation and maintenance routines, for operational statistics, for the collection and processing of meteorological readings, or for supervision and control of other equipment installed in the plant.

Batteries

In solar-wind plants the batteries work under special conditions. Daily low-rate charging and discharging are combined with a slower monthly and seasonal charging and discharging on a major scale. The battery current will be very small in relation to the capacity of the battery. There is an increased risk for sulphatization and a subsequent degradation in capacity if the battery is allowed to stay in a half-charged state for long periods.

The characteristics which are regarded as being of most value for batteries in solar-wind plants are:
- high charging efficiency
- low self-discharge
- minimum water consumption

Fig. 12
Turbine testing in a wind tunnel

Fig. 13
Control equipment for the diesel generator

- ruggedness during transportation
- low purchase price
- long life.

The characteristics "high charging efficiency" and "low self-discharge" are important for the energy economy of the whole system.

In the lead-acid battery range a selection can be made from low antimony-alloyed, calcium-alloyed and pure lead. The low antimony-alloyed can be delivered dry, which makes transportation easier.

Nickel-cadmium batteries have good values except for charging efficiency. They perform very well at low temperatures and in fact this is the only battery type which can be used in conditions of extreme cold. However they are three to four times more expensive than lead batteries and so for economic reasons cannot be used in large-scale solar-wind plants.

As the charging currents are small in relation to the battery size, water consumption will be low. There is no need to inspect the battery more than once a year.

Charging on a regular basis several times per year is recommended to prevent sulphatization of the battery. In systems combined with a mini-diesel such charging takes place at regular intervals, thus providing secure full-charging and extended life length. The batteries are usually sized to permit a reserve time of one to two weeks.

Mini-diesel generator

With a mini-diesel generator in a solar-wind energy system the size of the energy sources and the battery can be reduced. Dimensioning of the system is simplified, and years which are poor in solar radioation and winds as well as unforeseen variations in load cause no interruptions of operation. The battery is charged regularly which counteracts sulphatization. In addition the battery can be charged during installation and also when the annual battery inspection is carried out. The mini-diesel generator can be used as a power source during maintenance and installation work.

The most telling arguments against conventional diesel generators in remote locations are the reliability problem and the high cost of operation and maintenance. These problems can be greatly reduced by the use of a simple auxiliary system and by limiting the running time.

A diesel generator must be very reliable in start-up in order to be used in solar-wind systems. There must be almost no margin of doubt about its ability to switch in as a final source of reserve power if solar and wind power have not been sufficient, over a long period, to charge the plant battery. Analyses of start failures in conventional diesel generators have shown that the majority of missed starts have been caused by faults in the auxiliary system; these have included missed start motor engagement, faults in the start motor, badly maintained or faulty starting battery, faults in the start battery charger or faults in the motor's supervision circuits. In order to eliminate the majority of these causes and to obtain a very reliable mini-diesel generator, both the separate starting motor and the motor supervision system have been excluded.

Fig. 14
The diesel generator unit

Power rating	1.5 or 3 kW
Output voltage	48 V (or 24 V)
Motor power	2.1 or 4.2 kW at 3600 rpm

The diesel generator unit consists of an air-cooled, single-cylinder diesel motor and a d.c. generator, fig. 14. The diesel motor and generator are belt-connected in order to facilitate service. The generator is equipped with a starter winding and is also used as the starting motor. Consequently no separate starting motor with associated starter battery and charging equipment is required. The plant battery serves simultaneous as a start-up battery.

A special unit has been developed to control the mini-diesel generator in solar-wind power systems. The main objective has been to limit the diesel generator running time. This control unit is based on a microprocessor; its software contains the following functions:
- supervision of the battery charging state
- clock
- voltage monitors
- start-up
- shut-down.

The battery charging state is supervised by means of a digital counter which continually registers the current to and from the battery. In this way the charging state can be monitored with great precision and the mini-diesel can be started up at a predetermined minimum remaining charge level. This is to prevent the diesel starting up an unnecessary number of times.

The clock function permits periodic start-up of the mini-diesel so that the motor's oil film is retained intact. It also permits different running times.

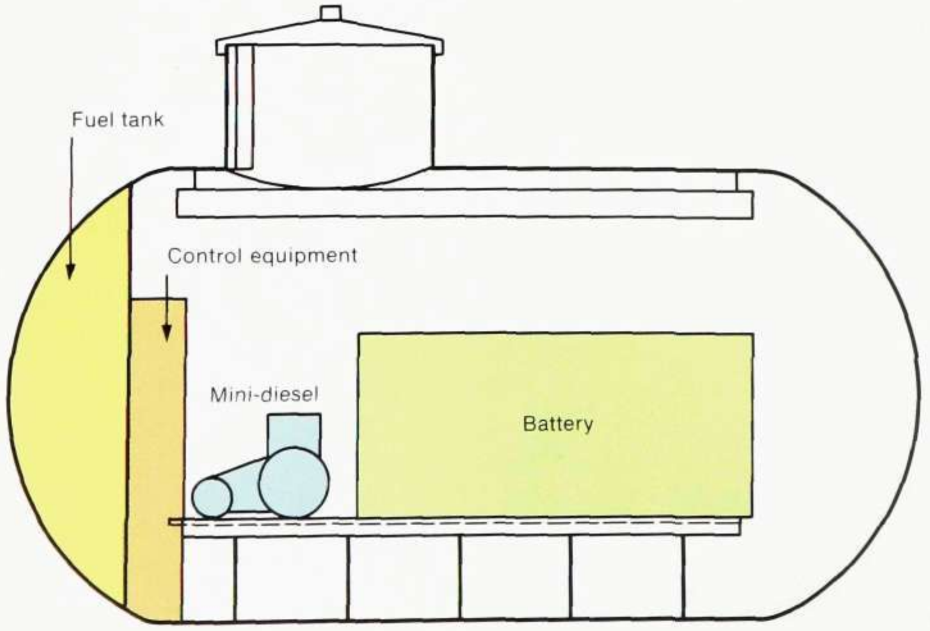
The voltage monitors are used to supervise over- and under-voltages and for the charging voltage level.

The mini-diesel is started up periodically at programmed intervals and when the charge of the battery falls below an acceptable minimum value or if the under-voltage monitor has issued an alarm.

Shut-down is initiated both when the programmed charging voltage level has been reached and the battery is fully charged and also if the overvoltage monitor has issued an alarm.

As the running time for the mini-diesel set has been limited with the help of the charging monitor, fuel consumption and repair requirements have also been reduced. The fuel and lubrication oil containers hold a 5-year supply assuming a running time up to 200 hours per year.

Fig. 15
Plastic tank for underground storage of ERICSSON SUNWIND equipment



Installation, operation and maintenance

The wind turbine is designed to be installed at the top of a mast, as the average wind velocity greatly increases with the height above ground level. If a microwave radio link is to be powered, then it is natural to utilize the mast which also carries the microwave antennas, fig. 16. Because of the height above ground level, the masts will often be exposed to lightning strikes. To prevent damage to the wind turbine, the generator and control equipment, these units have been given lightning protectors which lead the lightning generated currents to the earthed mast.

The solar cell panels can be mounted on the mast itself. In countries where the panel angle should be less than that permitted using mast mounting, special racks must be used.

Fig. 16
Mounting of the wind turbine

One very important point is that the ground equipment, i.e. the control unit, the batteries and the diesel generator, be placed in a controlled environment. In hot climates it is important that the average temperature is not too high because of the effect this will have on the life expectancy of the batteries and the electronic equipment. On the other hand, in cold-climate locations it is vital that the equipment be protected from sub-zero temperatures as these will reduce battery capacity and the diesel motor will be difficult to start.

Because the price of energy from a solar-wind plant is very high, it is not economically reasonable to provide a controlled environment for the equipment by means of cooling or heating as this will consume power. Instead, the equipment is placed in a shelter cooled by passive means or in an underground plastic tank, fig. 15. A separate storage area has been reserved for fuel for the diesel generator. Using these methods a controlled environment is guaranteed both in hot and cold climates.

When installing in remote sites it is very important that maintenance is kept to a minimum, because of the costs involved. The solar-wind system only requires one inspection per year. During this visit the battery must be checked, the wind turbine inspected and the solar cell panels washed. The mini-diesel must be functionally tested and the fuel tank filled, if it has not already been filled for several years' operation.

Summary

Solar and wind power is always available, although, by its very nature, it varies considerably in potential power capabilities because of changes in the weather. One day may be cloudy and windy, the next sunny and calm. These continual variations in climate have both short and long-term effects from wind gusts lasting for seconds up to daily, weekly and seasonal variations. During the year the sun and the wind supplement each other. This complementary nature smoothes out seasonal variations allowing the installed maximum power in the sun and wind energy-capture devices and the battery size to be reduced as compared with a non-combined system.

ERICSSON SUNWIND is designed to use this continually varying flow of energy in the best possible way; with microprocessor control the solar and wind energy converters can be guided to their optimal working points and at each moment extract the maximum amount of power; the vertical-axis turbine with its patented blade-pitch-adjustment offers secure operation even in extremely high wind velocities and is maintenance-free and highly reliable; and the mini-diesel generator permits a reduction in the size of the energy sources and battery while extending battery service life.

AXE 10 in the Stockholm Telecommunication Area

Sten Rimbléus

The Swedish Telecommunications Administration has decided to replace all telephone exchanges in the country by AXE 10 exchanges by the year 2020. In the case of the Stockholm telecommunication area, which comprises one million subscribers, the conversion is to be completed by about 2010.
The author describes the modernization plans and the systematic work methods that simplify the introduction of AXE 10. He also describes the two methods used for connecting the subscribers to the new exchanges, namely simultaneous or successive changeover. The operational experience from the first AXE 10 exchange in Stockholm is also reported.

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The Stockholm telecommunication area

Sweden is divided into 20 telecommunication areas and the Stockholm telecommunication area, with over 8000 employees, is the largest. The area is led by a director supported by a staff unit. The activities are divided into three geographical areas, Telephone areas North, Central and South, each with full responsibility for the results of its operation, fig. 1. They handle all installation as well as operation and sales within the area.

A number of facilities and activities are handled centrally for the whole telecommunication area:
- telex and data communications
- interception service and operator-assisted traffic
- mobile telephone traffic
- administration of buildings
- information
- training.

Approximately 75% of the one million subscribers within the telecommunication area are concentrated in less than 20% of the area. The subscribers are served by 218 telephone exchanges equipped with crossbar switches, 51 with 500-line selectors, one with code selectors and six AXE 10 exchanges. In addition there are a number of tandem, trunk and international exchanges the majority of which are equipped with 500-line selectors. Fig. 2 shows the present exchange positions and tandem area structure in the Stockholm trunk code area (08 zone).

The switching equipment consists mainly of machine-driven 500-line se-

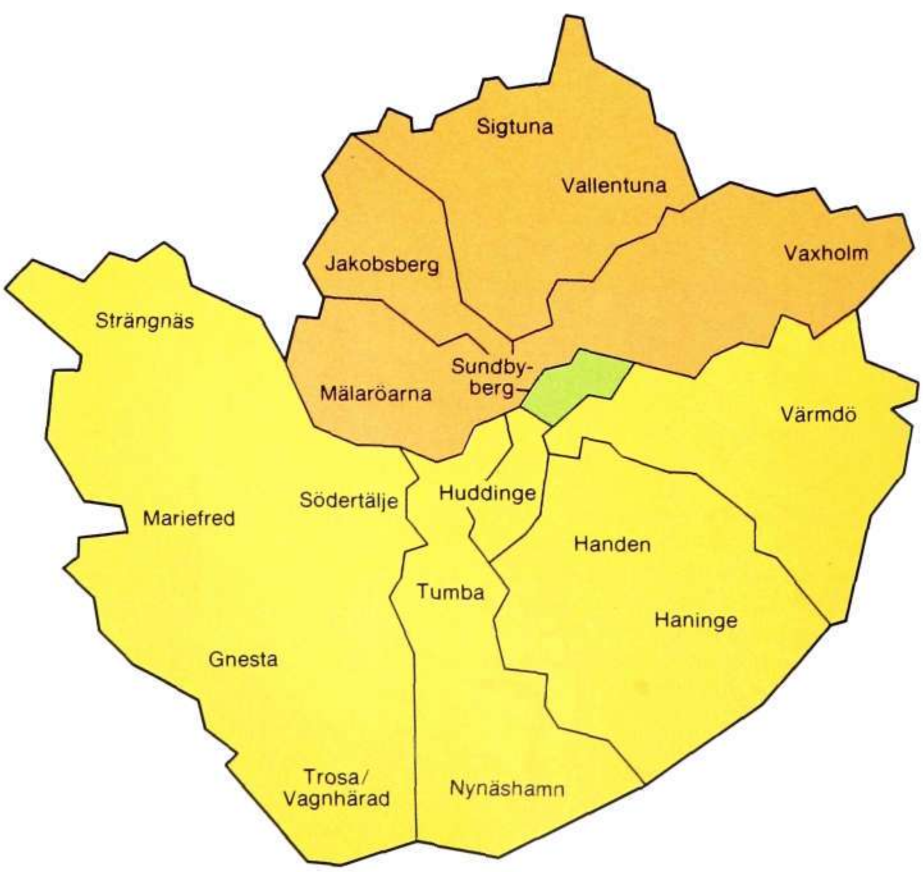


Fig. 1
The Stockholm telecommunication area and its division into telephone areas and sections

Telephone area North
Telephone area Central
Telephone area South

STEN RIMBLEUS
Swedish Telecommunications Administration
Telephone area Stockholm Central

lectors, a system which is becoming old and which does not meet present-day demands for functions and services. There are also problems as regards wear, spares and maintenance.

Modernization

The Telecommunications Administration has started an extensive modernization program, comprising the replacing of all 500-line selector exchanges in Sweden by the year 2000 and all crossbar exchanges by about 2020. Until then equipment that has been recovered from dismantled exchanges will to some extent be used for extensions caused by growth of the number of subscribers[1].

The central planning was started at the beginning of the 1970s and the project was designated MOTE (MOdernization of the TElephone network). In 1974 the Central Administration decided that all 500-line selector exchanges in the country should be replaced by AXE 10, and that detailed planning work should be started. It was later decided that the AXE 10 exchanges should be equipped with digital group selectors and that in the long run the whole network should be converted to digital operation. The advantages of this are, for example, that the trunk and subscriber networks will be less expensive, the transmission quality will be improved and it will be possible to integrate digital PABXs into the network.

The first 500-line selector exchange in the Stockholm telecommunication area was put in operation at Norra Vasa in 1924. It comprised 5000 lines. Since then a total of about fifty more 500-line selector exchanges have been installed. These exchanges are still being built out to a limited extent. They are now to be replaced by AXE 10 in approximately a third of the time required for the original building up of the network. This illustrates the scope of the modernization work now in progress.

Future network structure

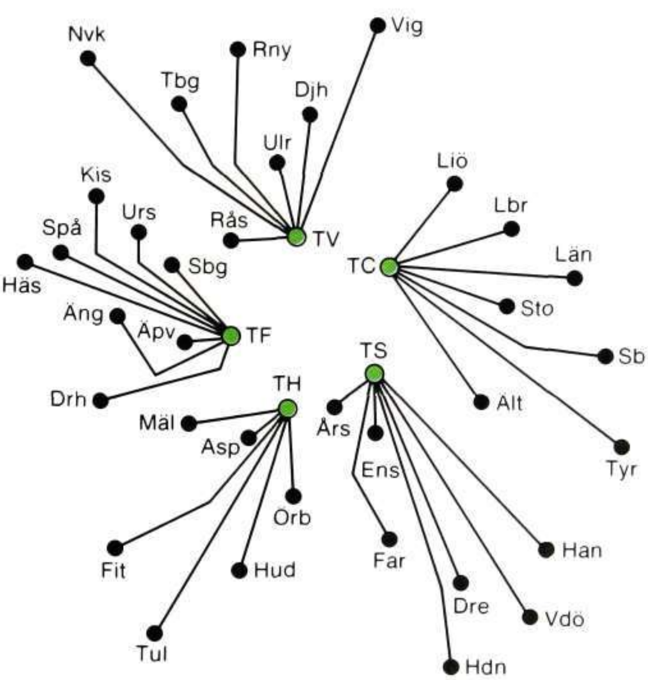
In order to obtain a suitable design for the new digital telephone network it is important to start the project planning process with strategic network planning. Such planning means investigating different ways of obtaining an envisaged target network 15 years ahead. For the Stockholm area different exchange structures have been studied, and also how the trunk traffic should be handled in future. Other studies included different circuit structures, local network configurations etc.

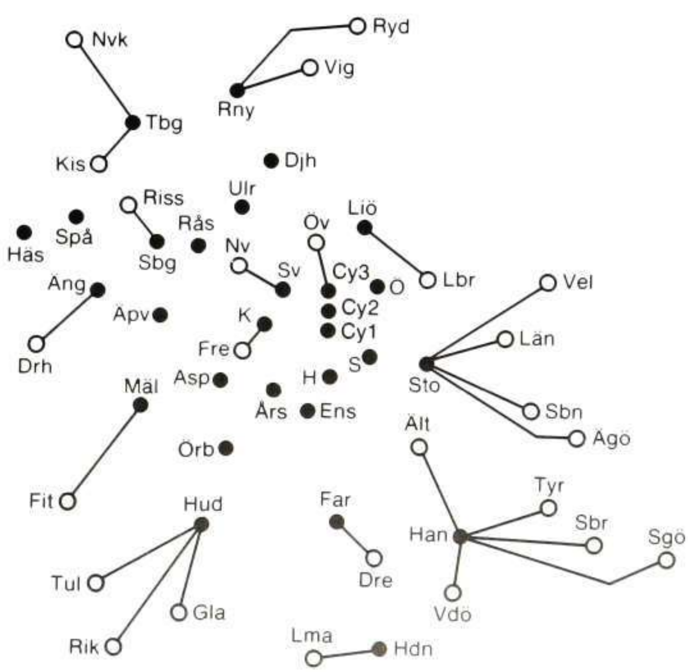
Fig. 2, left
Present exchange positions and tandem area structure in the 08 zone

- Tandem exchange
- Local exchange

Fig. 3, right
Exchange structure plan for the 08 zone

- Parent exchange
- Remote subscriber stage





At the present stage the structure plan for the 08 zone indicates that the best economic result will be obtained by limiting the number of exchanges with a central processor and digital group selector to approximately 30. These will then form parent exchanges, to which over 20 remote digital subscriber stages will be connected, fig. 3.

The size of the parent exchanges will normally vary between 20 000 and 50 000 subscribers. There will be three combined local/tandem exchanges, Råsunda, City and Örby.

The trunk traffic from the local exchanges will be transmitted via direct high usage routes or via the trunk exchanges Fredhäll (AKE 13), Hammarby and City. The last two are new AXE 10 exchanges. Each local exchange is connected to at least two trunk exchanges in order to obtain good availability and high reliability, fig. 4.

The circuits will normally be of the PCM type via cable or radio relay link. Fig. 5 shows a rough scetch of an envisaged future network between AXE 10 exchanges.

Digital subscriber stages will be introduced later on[2]. A 4-wire network that is fully built out all the way to the remote subscriber stages would given an improvement of approximately 10 dB as regards attenuation, compared with earlier analog networks. In certain cases coaxial cables having smaller cores than before can then be used in new installations and to replace old cables that need changing. Cables with optical fibres will also be used[3].

Modernization plans—strategy

The main purpose of the modernization is to replace the oldest exchanges. The dismantled equipment will be used for extensions elsewhere as required. Beyond this basic aim further modernization will also be carried out if the market demands are sufficient to cover the necessary capital outlay. The modernization investments are concentrated to those areas in the telecommunication field that are most interesting from the point of view of marketing, namely city areas and certain suburbs. In addition PCM will be introduced on circuits between exchanges to the greatest possible extent.

The modernization plan is based on an order of priority, where the main factors have been

- market
- growth rate
- need for replacement (age)
- prospects of reuse
- premises
- technical aspects.

According to plan, all 500-line selector

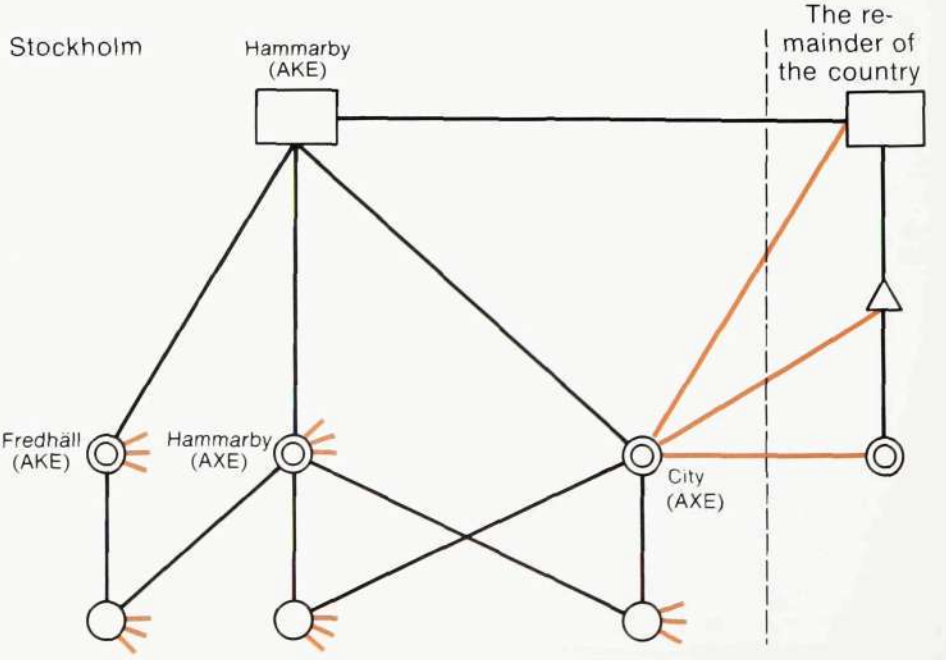
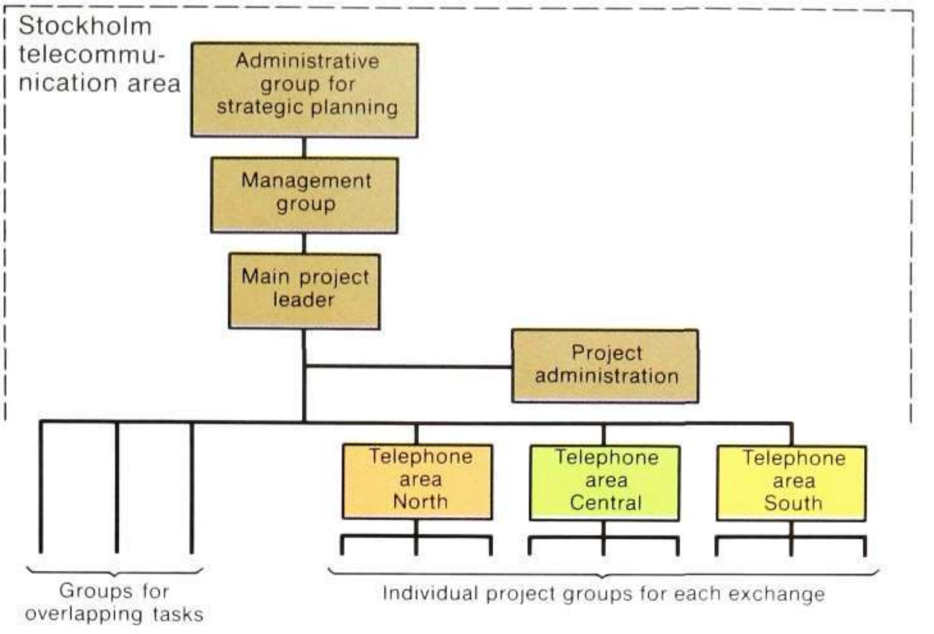


Fig. 4
The routing of trunk traffic from local exchanges in the Stockholm trunk code area (08 zone)

Fig. 6
The project organization for modernization of the telephone network in the Stockholm telecommunication area (MOTE-S)



exchanges in the Stockholm area are to be replaced before the year 2000, and all crossbar exchanges by around 2010.

Tandem and trunk exchanges have been given priority so that digital operation can be introduced at the earliest possible stage.

Simplifying the modernization work

A project group within the Stockholm telecommunication area has been working on the implementing of the modernization plans since 1978.

The main aim of the project is to coordinate the changeover to AXE 10, the introduction of PCM and the adaptation work required in the network when digital operation is introduced.

The project comprises planning, coordination, follow up, method studies, information and training.

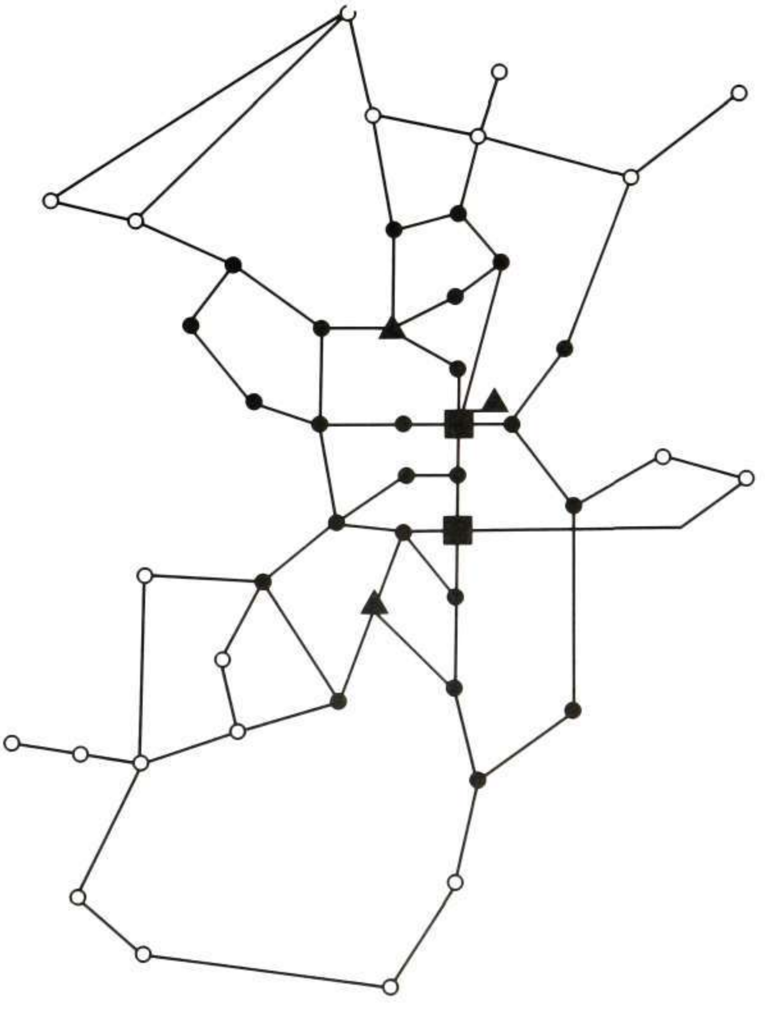


Fig. 5
Future network between the AXE 10 exchanges in Stockholm

○ Exchanges in the 07 zone
● Exchanges in the 08 zone
▲ Tandem exchanges
■ Trunk exchanges

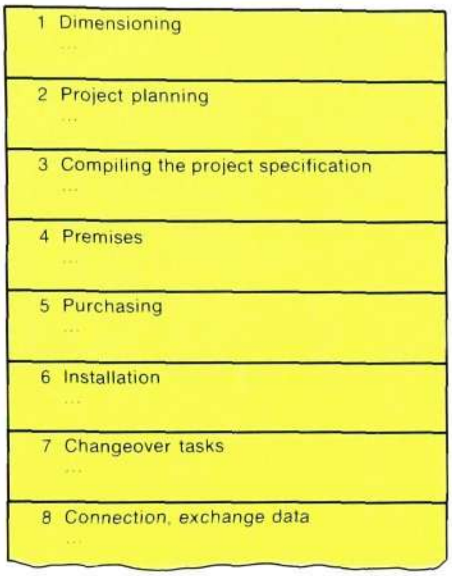


Fig. 7a
Planning schedule for the replacement of a 500-line selector exchange by AXE 10

The work has been very carefully planned and organized. Efficient work methods have been developed and great care taken in training the personnel in the new technology.

Organization and distribution of responsibility
Fig. 6 shows the project organization for the Stockhom area (MOTE-S). The tasks of the project groups are primarily as follows:

- *The administration group* has the *stratetic responsibility* for MOTE-S and makes decisions mainly concerning the effects of circumstances and problems outside the project. The decisions can concern finances and the allocation of resources. The group contains the three heads of the telephone areas.
- The *maganement group* has a *tactical and operative responsibility* for the project. This means the distribution of resources, responsibility for the current project activties and also technical principles and conditions. The group contains technical chiefs from the telecommunication area.
- *Investigation groups* are appointed for fundamental matters that concern the whole area, for example the development of a structural plan for the exchanges, the introduction of mini-MDFs etc.
- *The project groups* (BAXE, Preparation groups for AXE 10), one for each exchange which is to be replaced, are responsible for the work until the new exchange has been taken into service. Each group is led by a project leader and consists mainly of staff from within the telephone area concerned. A special sub-project leader with special responsibility and authority joins the group during the stage when the exchange is being put into operation.

The main project leader handles the coordination with the management group and ensures that important questions are referred to the administrative group.

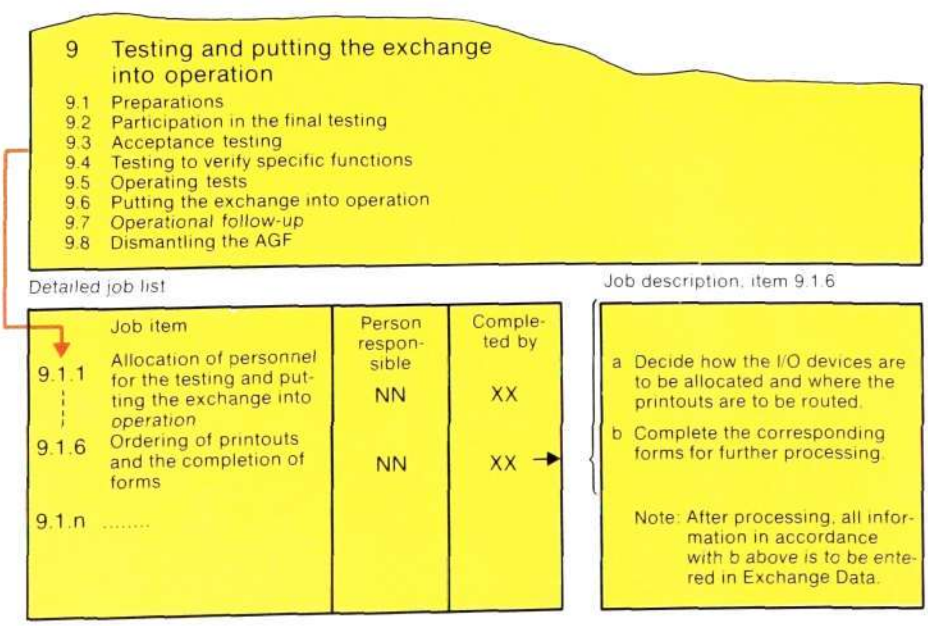
Systematic work methods
In view of the extensive modernization program and the limited personnel resources, great efforts have been taken in the initial stage to develop systematic work methods. A main schedule has been prepared for the various projects. Access to exchange premises has often been the factor that determined the starting dates of the individual projects.

All stages in the work are specified in chronological order on a standardized planning schedule. Reference is made to detailed job lists, which in their turn refer to job descriptions, fig. 7. Experience shows that these schedules and lists simplify the work considerably and ensure that the necessary preparations are made well in advance. The main activities listed in the planning schedule are:

Dimensioning
The dimensioning outlines the overall framework for the task. Initially, rough dimensioning is carried out which provides the basic data for the requirements as regards premises and power supply equipment. Traffic measurement results and the results of various types of prognoses are then compiled and detailed dimensioning is carried out, which forms the basis for decisions concerning the changing-over method and traffic routes.

Surveys are made in the network, for example of the number of PBXs, for the subsequent planning work. The power requirements for the fully built out exchange are calculated.

Fig. 7b
Testing and putting an exchange into operation



Project planning
On the basis of the dimensioning data, documents are prepared for a fully extended AXE 10 exchange with a control room, MDF, power supply and cable vaults. Documents for the adaption of the network are also prepared. In most cases PCM equipment is introduced in the junction network in connection with the replacement of an exchange.

Compiling the project specification
A preliminary project specification is compiled by the traffic engineers and the exchange planner. The Central Administration at Farsta and the supplier are then contacted and a final project specification is prepared.

Premises
A specification is made of the requirements as regards premises. A building schedule and building documents are prepared. After the agreement with the contractor has been approved the building work is started according to schedule.

Purchasing
The AXE 10 equipment is purchased as soon as the final project specification has been accepted. Other equipment is purchased separately.

Installation
In this section of the schedule the distribution of responsibility for and the documentation of the installation work are specified. The supplier installs the purchased equipment.

Changeoever tasks
These tasks consist mainly of the preparation and changeover work that has to be carried out in order to be able to test the exchange and put it into operation.

Connection, exchange data
The detailed planning of the connection work is carried out in accordance with a specially prepared schedule. Exchange and subscriber data are compiled with the aid of computer-based systems. These systems produce cassettes which are used to load the data stores

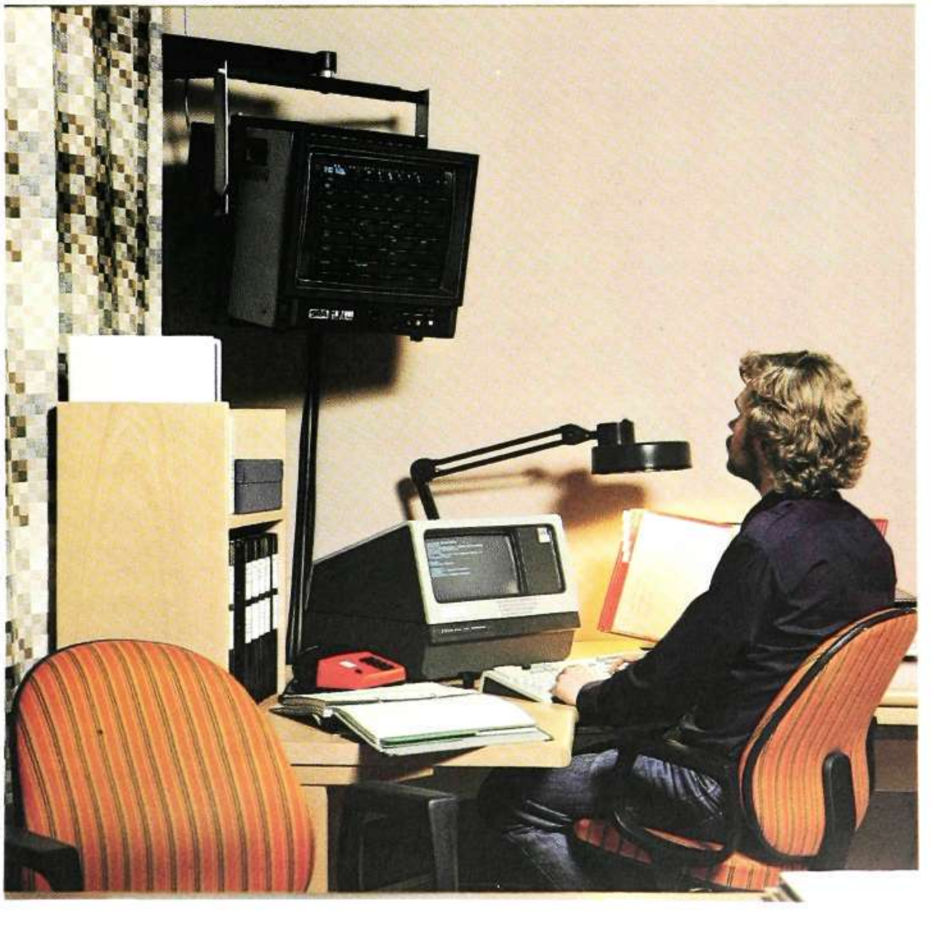


Fig. 8
A part of the maintenance centre for Telephone area South

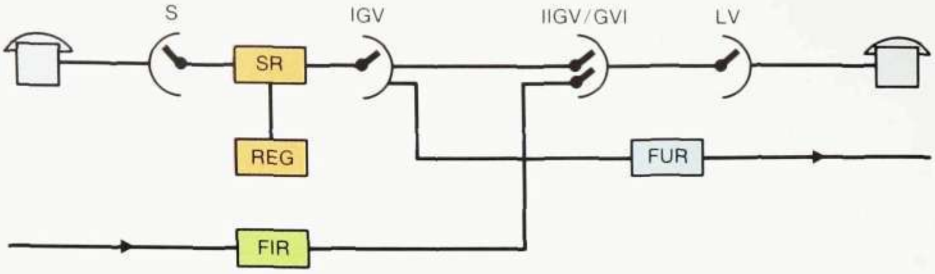


Fig. 9a
Trunking diagram for a 500-line selector exchange in Stockholm

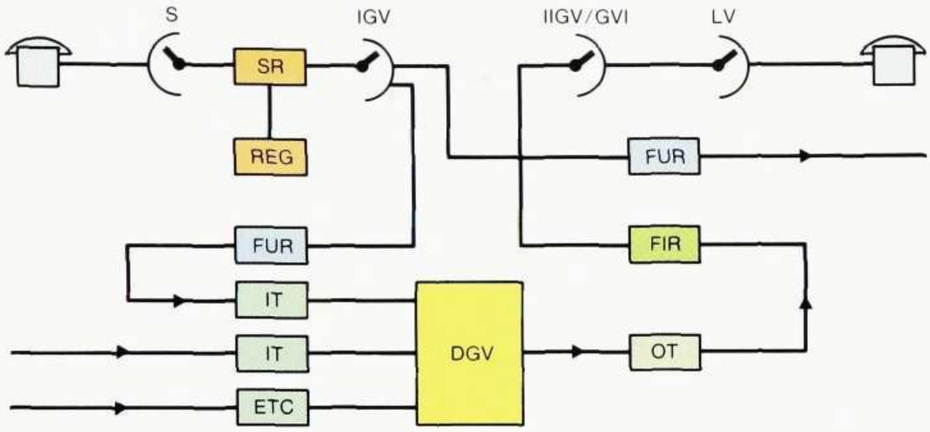


Fig. 9b
When the first changeover stage has been completed the internal and incoming traffic to the 500-line selector exchange is switched by the digital group selector, DGV, in AXE 10.
Order of changeover:
1 A route to IIGV/GVI is wired up from DGV
2 Incoming routes and one or more routes for internal traffic are wired to DGV
3 At the end of the first stage all incoming traffic goes via DGV

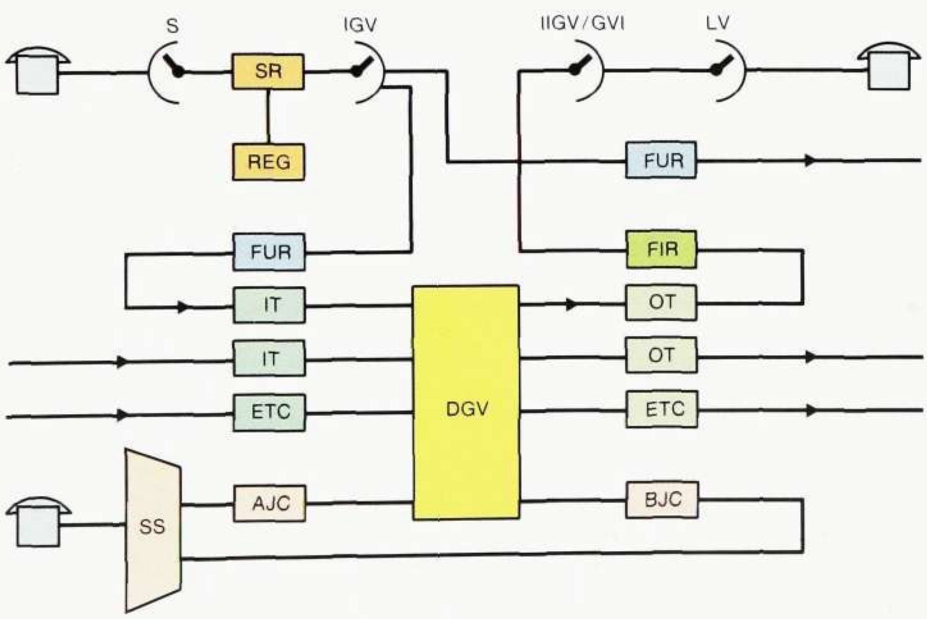


Fig. 9c
The trunking diagram for the second stage, when the AXE 10 subscriber stage, SS, is opened for traffic and the changeover of subscribers from the 500-line selector exchange is done successively.
1 The routes to IIGV/GVI are reallocated as routes for internal traffic to SS through DGV and BJC in step with the changeover of subscribers to AXE 10.
2 The outgoing routes from the IGV of the 500-line selector exchange are reallocated to traffic via outgoing routes, OT, via DGV.

of the exchange. The input data are then modified by means of a command cassette or directly with commands from a terminal.

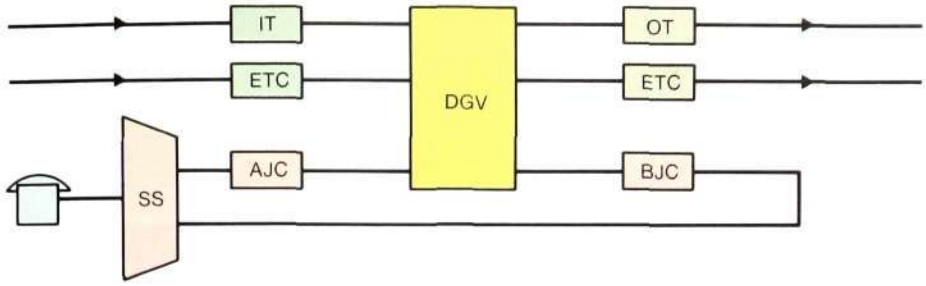
Testing, putting into operation
After the preparations for putting the exchange into operation the staff of the telecommunication area participate in the final tests carried out by the supplier. After that the area staff carry out their own tests, which include acceptance tests, tests that verify specific functions and operational tests. When all tests have been completed and the delivery has been accepted the exchange can be taken into service.

Changeover methods
The following two changeover methods are used in order to make the transition to AXE 10 as efficient and free from disturbances as possible.

Simultaneous changeover, which is mainly used for small local exchanges. With this method no extra equipment is required for traffic between the old and the new exchange. However, a considerable amount of concentrated work is necessary both during ordinary and inconvenient working hours. Wiring changes must also be carried out in surrounding exchanges and PBXs at the same time. For these reasons this method should be used mainly for local exchanges with a capacity of less than 10 000 lines.

Successive changeover, which is used for large exchanges and is carried out in stages, can be adapted to the available resources. This method is particularly advantageous when the exchange building cannot hold both the complete new AXE 10 and the old electromechanical exchange. The AXE 10 control system and the digital group selector are then immediately built out to the required capacity. The rest of the switching equipment is built out in stages according to the conditions of each individual case. A certain amount of temporary equipment is required for the traffic between the old and the new exchange, but this equipment can be reused. Fig. 9 a–d shows the main stages of the successive changeover from a 500-line selector exchange to AXE 10. This method has the following

Fig. 9d
The trunking diagram for the AXE 10 exchange when all subscribers have been changed over.



advantages:

- the subscribers can be connected in in any optional order, for example per primary cable or per 500-group
- the risk of disturbing the subscribers is minimized
- the wiring changes in adjacent exchanges, PBXs and the network can also be carried out in stages, which means that most of the work can be done during ordinary working hours.

However, the actual changeover work must be extended over a relatively long period of time.

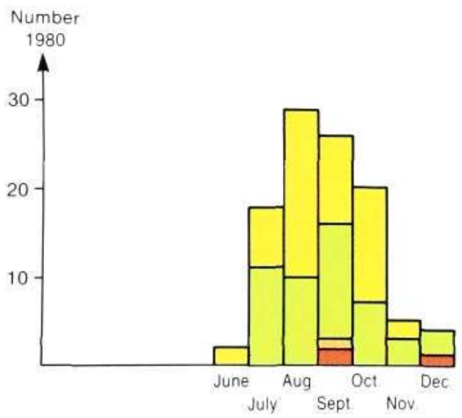
Fig. 10
Operational statistics from the AXE 10 exchange at Ulriksdal

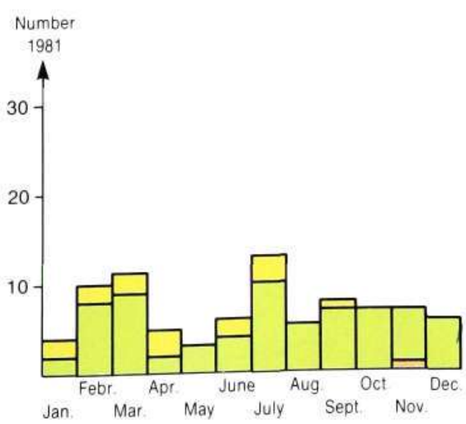
Minor system restart is usually caused by a software fault and takes 7 seconds. No disturbance of calls in progress.

Minor, manually initiated restart is made because of a software fault which has led to blocking of a device or individual subscribers. It takes 7 seconds and calls in progress are not disturbed.

Major automatic restart is usually caused by software faults and takes 7 seconds. Calls in progress are disconnected.

Major automatic restart which includes reloading of the programs takes 10 minutes. All traffic is interrupted.





Operational experience

The first AXE 10 exchange with a digital group selector was taken into service at Ulriksdal on 12 June, 1980. The previous 500-line selector exchange served 7 300 subscribers, all of which have been connected over to the new exchange. The AXE 10 exchange is dimensioned for 11 000 subscribers. The exchange was taken into service and the subscribers connected in in accordance with the successive changeover method. The work took two months. The operation of the exchange has been followed very closely and the operational experience has been extremely satisfactory, as can be seen from the following.

Hardware faults

Two modifications have had to be introduced since the exchange was put into operation. One raised the tone level by 5 dB and the other adjusted the interface between the PCM equipment and the digital group selector. A number of printed board assemblies with faulty components have also been replaced.

Software faults

A number of software faults were detected during the first months. Five software modifications have successively been introduced in order to eliminate certain traffic disturbances, and about 30 in order to improve the handling.

System restarts

System restarts are carried out automatically when incongruous data are detected or a program handling fault occurs. Restarts can alse be ordered manually. Fig. 10 shows the system restarts that have occurred since the exchange was taken into service.

Breakdown

The only breakdown that has occurred hitherto took place after six months of operation. It lasted approximately 24 minutes and occurred when a large program package was being replaced. The staff were aware that such a major job could cause disturbances and it was therefore carried out when the traffic was very low, after midnight. The effects of this breakdown were thus not very serious.

Traffic quality

Test traffic has been generated by traffic route testers and the percentage of calls lost because of congestion or technical faults has been as low as 0.12%.

Traffic recording

The traffic recording functions that are built into AXE 10 have been used regularly. The results show that all routes are correctly dimensioned and that the traffic handling capability is good.

Manning

Since the exchange was put into operation it has only been manned during ordinary working hours, i.e. Monday-Friday 0730–1600 hours, fig. 11. Administrative staff have on occasion been placed at the exchange in order to follow events there and to develop the operation and maintenance routines further. When not manned, the exchange has been supervised from a maintenance centre which is common for the whole telecommunication area. During the first year a total of 25 alarm calls where received which resulted in a repairman being sent to the exchange. All these faults could be cleared with the aid of the available operating instructions.

Training

The Telecommunications Administration has long provided further specialized training for personnel working with AXE 10. For this purpose a number of independent part courses have been prepared, which can be combined to suit different training requirements. The following example shows a course for

Fig. 11
The control room in the AXE 10 exchange at Ulriksdal near Stockholm when the exchange was being put into operation.

AXE 10 technicians. It lasts 19 weeks and consists of the following six part courses:
- 1 week, basic course on SPC technology
- 2 weeks, basic course on AXE 10 technology
- 1 week, basic course on operation and maintenance methods
- 4 weeks, digital exchange and PCM technology
- 7 weeks, detailed AXE 10 course
- 4 weeks, detailed course on operation and maintenance.

The above-mentioned part courses will later be supplemented by revision courses.

A large part of the training is carried out locally, but the Administration also maintains a Training Centre for centralized training. The 4-week course on operation and maintenance is held at the Training Centre, where the students have access to a complete training exchange in which faults can be simulated. This gives valuable practical experience together with the theoretical training.

Centralized operation and maintenance

System AOM 101 for operation and maintenance of the AXE 10 exchanges will be introduced gradually[4]. Operation and maintenance centres will be set up in each of the three telephone areas. Each centre will be equipped with display units, printers and cassette tape recorders connected to the central AOM 101 equipment. These three centres will only be manned during ordinary working hours. At other times the supervision will be carried out at an operation centre that will be common for the whole of the Stockholm telecommunication area.

A field group for the operation and maintenance work will be set up in each of the three telephone areas. The field groups will also assist in certain parts of the preparation work and when the AXE 10 exchanges are put into operation. This will enable the technicians in the field groups to retain their knowledge of the system. At a later stage, when the exchanges and maintenance methods are completed and working well, the technicians in the field groups will only have to visit the exchanges when manual intervention is necessary. The size of the field groups will be calculated on the basis of one technician per 30000 exchange numbers.

However, large exchanges will have a certain number of permanent staff for work in the MDF, work pertaining to the premises etc.

Conclusion

The work on replacing old 500-line selector exchanges and crossbar exchanges by modern AXE 10 exchanges proceeds according to plan. Six of the new exchanges have been taken into service, and at the time of going to press planning or installation work for about ten more exchanges is in progress.

The experience gained hitherto shows that the AXE 10 system and the methods developed for the realization of this huge modernization project have certainly come up to expectations.

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Contents

COVER
Operator's position with equipment for word and text processing, for communication over external networks using telex and teletex, and with access to central ADP systems

DIAVOX Courier 700, Digital System Telephone for MD 110

Jonas Reinius and Olof Sandström

MD 110 is a digital SPC-PABX for 100 to 10000 extensions. A general description of MD 110 and description of the digital signal processing in the system were provided in two articles in the previous number of this publication[1,2]. In addition to analog, conventional telephones it is also possible to connect digital system telephones, designation DIAVOX Courier 700, to MD 110. In this article the authors describe the design of digital system telephone DIAVOX Courier 700, the facilities it provides and some application fields.

UDC 621.395.6.037.37

With a digital telephone like system telephone DIAVOX Courier 700 it is possible to simplify some of the functions available with a conventional telephone. Numerous new extension facilities can also be introduced. It is thereby possible to attain a heavy traffic handling volume and an excellent overview of ongoing, camped-on and queuing calls. Flexible programming routines permit the functions of DIAVOX Courier 700 to be adapted to the individual requirements of each user. Extension user facilities that previously required several telephone systems e.g. intercom, line pickup, queuing and diversion, can now be incorporated in the same telephone. Data communication facilities, that previously required a parallel network can now be integrated in MD 110 with DIAVOX Courier 700.

DIAVOX Courier 700

DIAVOX Courier 700, digital system telephone for MD 110, is a member of the DIAVOX family and utilizes the new, modular telephone design Courier. The design is modular in the sense that the same basic tools, complemented with add-on modules for different sizes, are used in the manufacture of the mechanical parts. Courier utilizes the same handset as the other telephones in the DIAVOX family.

Initially, digital system telephones will be available in two versions, with one alternatively three rows, each row containing 12 function fields. Each field has a button, an associated light emitting

Fig. 1
Variants of the MD 110 digital system telephone DIAVOX Courier 700 with one and three rows of function fields

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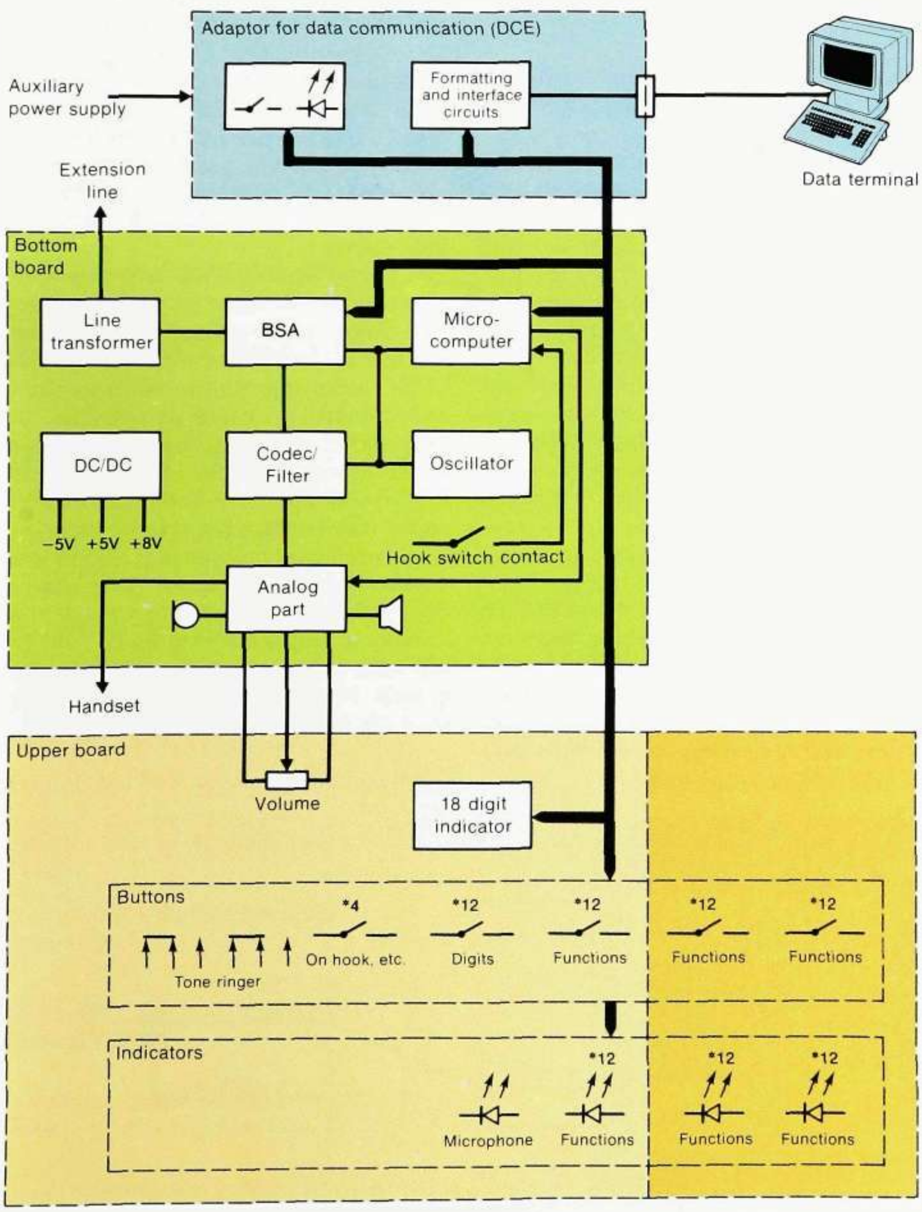
diode, LED, and designation strip. It is a simple matter to remove and replace the designation strip for an entire row, fig. 1.

Digital system telephone functions
The two versions of DIAVOX Courier 700 that will be available initially contain the following:
- handset
- voice controlled loudspeaker function with adjustable volume, connection button and privacy button
- tone ringer with three different characters and two levels, alternatively switched off
- 18 character, 7 segment liquid crystal display, LCD
- pushbutton unit for dialling calls
- function buttons for disconnection, transfer and switching to programming mode
- three function fields for own line, *triple access line*
- 8 alternatively 32 function fields for programmable functions, depending on the size of the telephone
- jack for tape recorder
- jack for "don't disturb" indicator (for doorpost signals)
- jack for connection of Data Circuit terminating Equipment, DCE.

Fig. 2
Block diagram for DIAVOX Courier 700



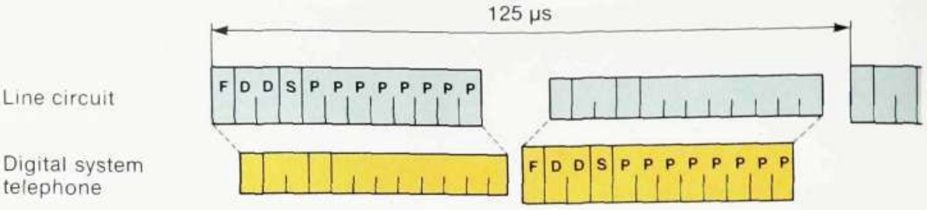
Design
The digital system telephone, whose block diagram can be seen in fig. 2, receives a 48 V low-ohmic feed from the line circuit. Feeding is supervised and disconnected when the line is shortcircuited. The signal on the line is led via a transformer into a specially designed CMOS-circuit. The latter contains all logic required for line transmission and signal word formatting. The circuit is designated BSA *(Burst Signalling Adapter)*. BSA can be programmed as controller on the line circuit or as slave in the telephone. PCM-data is fed from the BSA-circuit directly to codec and filter, which in their turn are connected to the voice part of the digital system telephone.

Connection
Transmission is digital which provides the following advantages:
- voice and data communications as well as line signalling can take place on the same extension line independently of one another
- non-attenuated voice transmission, irrespective of line length
- the transmission characteristics are identical to those for a four wire connection.

The digital system telehone has a two-wire connection, fig. 3, which allows existing cable networks for analog extension telephones to be used on changeover to a digital system telephone, even when the latter has a data terminal connected. This can save time and money.

Fig. 4
Principle for burst signalling. Each burst consists of 12 bits bi-phase-coded information. The bi-phase frequency is 256 kbit/s.

- F Frame synchronization bit
- D Data information
- S Line signalling information
- P Voice information (PCM)



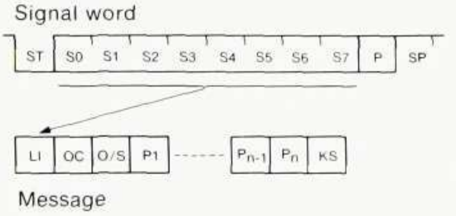


Fig. 5
Principle for line signalling. Each signal word is composed from the line signalling bit from each of 11 consecutive bursts. The message is formed from minimum 4 and maximum 23 signal words. To check transmission the signal word has start/stop format with parity check and the first word of the message is length information and the last word a check sum

Signalling

All signals between MD 110 and the digital system telephone are exchanged in digital form, in bursts, fig. 4. The extension line unit, ELU-D, sends a 12 bit burst at a transmission speed of 256 kbit/s every 125 μs. The telephone is synchronized with this rate so that it sends its burst after receiving a burst. The delay between transmitted and received burst in the line unit of the PABX depends on the line delay. Burst length, repetition frequency and delay combined limit the length of the line to about 2,000 metres for this signalling method. Other factors such as current feed, signal attenuation and disturbances limit the range to about 1,000 metres.

The line transmission function utilizes three mutually independent, bothway channels for:

- voicetransmissionat a rate of64 kbit/s with one PCM-sample per burst
- line signals at a rate of 8 kbit/s with one bit per burst. A signal word is composed of the signal bits from 11 consecutive bursts and consists of 8 bits signal information plus synchronization and parity bits, fig. 5. The signal words are assembled into messages of varying lengths. The message contains check information in the form of message length and check sum. The receiver acknowledges the message after first approving it. Retransmission takes place if a fault exists. A message can be "raised handset" from the telephone or "light LED with call flashing and start tone ringer" to the telephone
- data transmission at a rate of 16 kbit/s with two bits per burst, a data flow that is processed in the auxiliary unit attached to the telephone.

The line transmission also includes synchronization with 8 kbit/s, two half bits per burst.

The line signals proceed word by word between the BSA-circuit and the telephone's CMOS-microcomputer. The latter is responsible for the line signalling i.e. it deals with the signalling protocol, checking the received messages and time supervision for acknowledgement of transmitted messages. The microcomputer also controls the scanning of button and hookswitch signals, cadences for LEDs and character transfer to the 18 character LCD.

The digital system telephone does not possess its own storage function for abbreviated dialling or other facilities; all data for these are stored in MD 110. Only cadences for tone ringers, LED and LCD signals are stored in the telephone. These are programmed on installation.

Indications

The digital system telephone has optical and acoustic indications. Each function field has an associated LED that can display five different states, fig. 6, in order to provide the user with as clear and detailed a picture as possible for each traffic function. The 18 character LCD is normally divided into three fields to provide the user with information about numerous different situations in an unequivocal manner, fig. 7. For example, on an incoming call to the user's own line, *triple access line*, the caller's identity is displayed in the left field of the LCD before the user answers. On answer the caller's number moves to the right field and the left is blanked.

Ring signals are issued by tone ringing

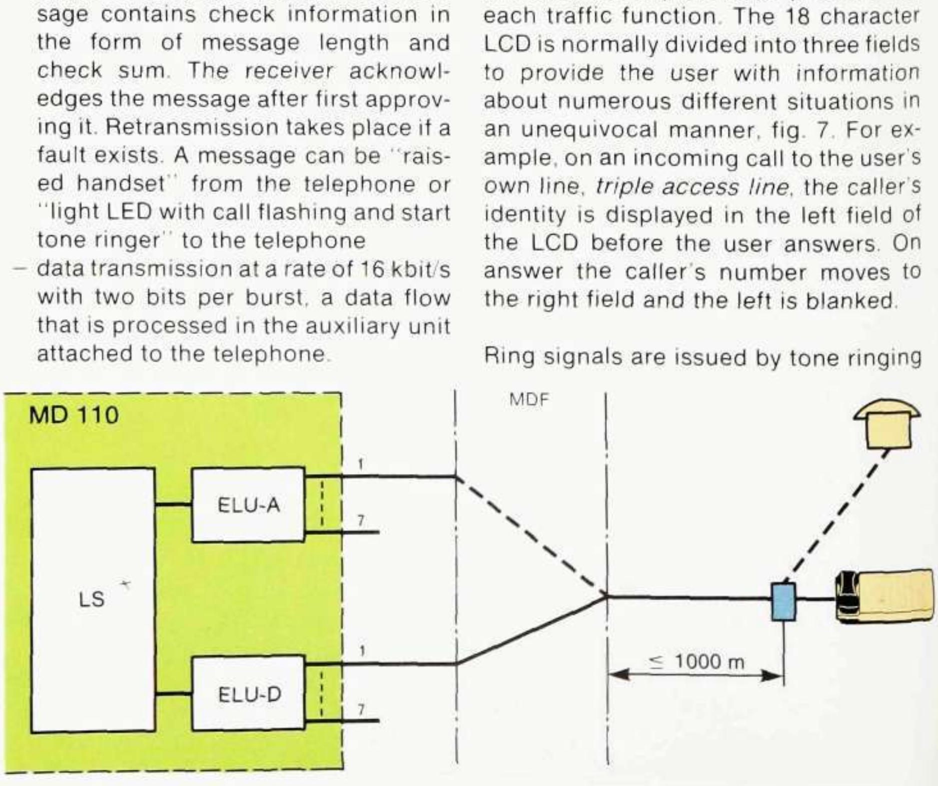
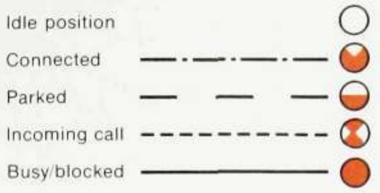


Fig. 3
Connection of DIAVOX Courier 700 via normal extension line



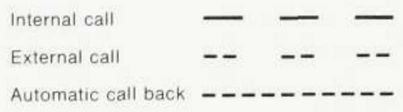


Fig. 6, left
Example showing optical signals for the function field LED

Fig. 8, right
Example showing cadences for tone ringer

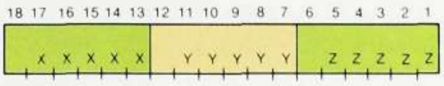


Fig. 7a
The use of the fields of the LCD

X Calling party
Y Diverted or dialled party
Z Called party, connected conversation party

Fig. 7b
Example of what the LCD can show in various traffic cases

Status	Direct traffic	Diverted traffic
	Set No. 2000	Set No. 3000
REGISTER	DIALLED DIGITS	
BUSY, CONGESTION, etc.	1000	1000 6000
OUTGOING CALL, internal	1000	1000 6000
OUTGOING CALL, external	007499239	
INCOMING CALL, internal to triple access line	4000	4000 2000
INCOMING CALL, external to triple access line	E	E 2000
SPEECH CONNECTION, internal	4000	2000 4000
SPEECH CONNECTION, with incoming call to triple access line	3000 4000	3000 2000 4000
EXTERNAL SPEECH CONNECTION, incoming call	E	2000 E
EXTERNAL SPEECH CONNECTION, outgoing call	007499239	
EXTERNAL SPEECH CONNECTION, incoming call to triple access line	3000 E	3000 2000 E

Note: 1000 diverted to 6000, 2000 diverted to 3000

in the telephone loudspeaker. The digital system telephone user can program one of five ring alternatives for each traffic function, namely; no ring signal, ring signal after delay, first ring signal (muted) only and one ring signal (muted) only after delay. Incoming calls during an ongoing conversation are announced by the first ring signal (muted) only.

The tone ringer has three different cadences with which to announce internal calls, external calls and automatic callback, fig. 8. Each extension can select one of three different signal characters on the digital system telephone. This is helpful if there is more than one telephone in the room. It is also possible to choose between a loud signal, muted signal or disconnecting ringing completely.

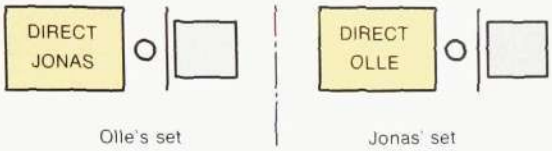
Traffic facilities

A number of new traffic facilities have been defined for the MD 110 digital system telephone. These make possible the inclusion in one system of facilities that previously demanded several parallel communication systems. Furthermore some functions have been simplified, for example a complicated procedure has been reduced to a single button depression.

Outgoing calls can be initiated from a digital system telephone by:
- lifting the handset, waiting for tone and dialling the required number precisely as with a conventional telephone
- pressing the loudspeaker button, waiting for tone from the loudspeaker and dialling the number
- dialling the number directly by pressing the digit buttons without waiting for tone. After the entire number has been dialled a tone message indicating that the called party is free, busy etc. is received via the loudspeaker
- pressing a traffic function button, waiting for tone in the loudspeaker and dialling the number in te normal manner.

When the user lifts the handset the digital system telephone is switched from the loudspeaking to the normal voice mode.

An incoming call is indicated with rapid flashing on a function field LED and can be answered by:
- lifting the handset or pressing the loudspeaker button if the call is to the extension user's own line, *triple access line*
- pressing the traffic function button that indicates the call, accepting the call in the loudspeaker mode and then, if desired, switching to normal voice mode by lifting the handset.



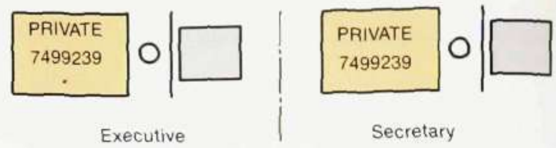


Fig. 13, left
Dedicated intercom link facility

Fig. 14, right
Exclusive external line facility

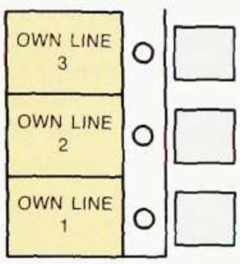


Fig. 9
Triple access line facility

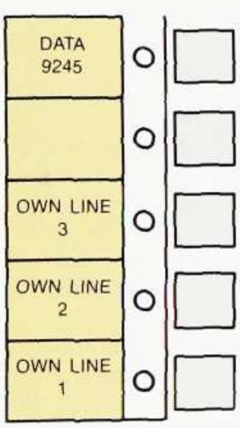


Fig. 10
Data line facility

New traffic facilities

Triple access line corresponds to the extension line of a conventional telephone. Each digital system telephone has one and only one *triple access line*. This line is represented on the system telephone by three function fields, and thus has three separate connection possibilities. An extension can namely be free for new calls despite one connection possibility already being seized. When the new call is answered the ongoing call is parked. Two connection possibilities have now been seized, one for parking and one for conversation. To facilitate transfer of the latter call the third connection possibility, for outgoing calls, is used and three connection possibilities are seized, two for parking and one for conversation. A prerequisite for the digital system telephone to be free for incoming calls is that a free connection possibility (field) exists for inquiry and that no unanswered, incoming call to the *triple access line* already exists, fig. 9.

Data line facilitates establishment of data communication irrespective of ongoing voice communication. All states are indicated optically only, in order not to disturb ongoing conversation. For an outgoing call the *data line* button is pressed whereupon the LED double flashes to indicate dialling state. The B-number, external or internal, is dialled in the normal manner. The *data line* LED on the called party's telephone flashes rapidly to indicate a call. When he answers the switch to data mode ensues and the rapid flashing of the LED changes to a steady glow. If the called party is busy this is indicated to the caller by rapid flashing on the DCE and the caller can disconnect or camp-on, fig. 10.

Extra line. A digital system telephone can have an arbitrary number of *extra lines*, only the quantity of function fields is a limiting factor. The free states of the *extra lines* do not affect and are not affected by the free state of the *triple access line*, fig. 11.

Line pickup is the name of the traffic facility that allows the *triple access line* or *extra line* of a digital system telephone to be represented in the digital system telephone of another extension, e.g. for call diversion. The facility is used for incoming and outgoing calls. A line can be represented for *line pickup* purposes in up to 30 other digital system telephones, fig. 12.

Dedicated intercom link can be regarded as a direct connection between two digital system telephones that cannot encounter busy state although congestion state is possible. The called party knows immediately who is calling as the caller's name is on the designation strip adjacent to the LED, fig. 13. If the extension has programmed the *dedicated intercom link* for immediate voice connection then the same function as for intercom calls is obtained.

If A calls B and receives no answer he can inform B that he has called by causing the appropriate LED on B's telephone to flash in a special manner.

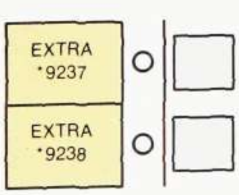


Fig. 11
Extra line facility

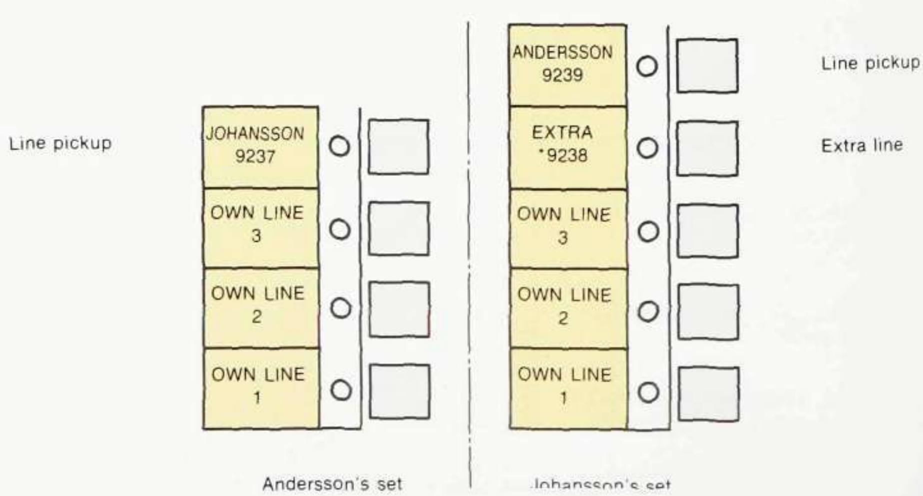
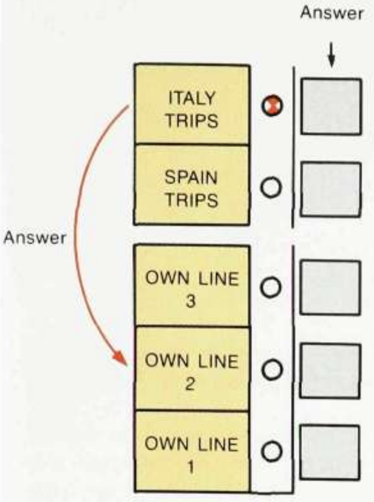


Fig. 12
Line pickup facility



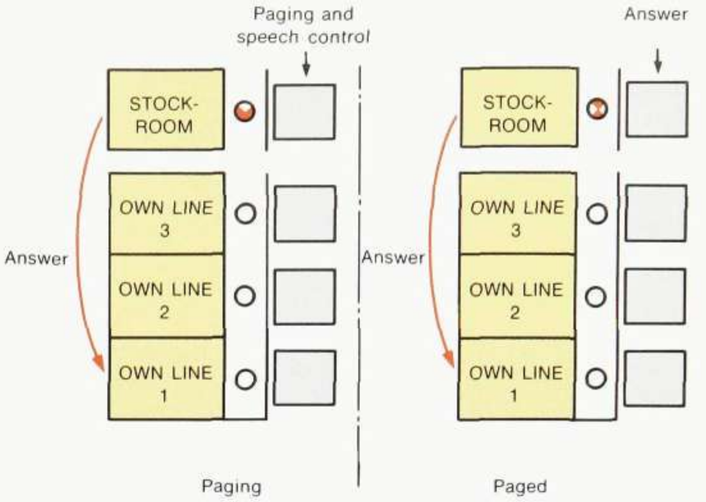


Fig. 15, left
Common bell call pickup facility

Fig. 16, right
Voice paging facility

If B is already busy with a call and prefers to answer the call later, he can advise A accordingly by pressing a button which gives A the appropriate tone message while B's LED starts flashing in a special manner as a reminder to call A later.

Exclusive external line is similar to a separate telephone subscription. A trunk line is reserved for each *exclusive external line*. This can be represented in up to 30 digital system telephones. Incoming calls are indicated directly on the telephones. For outgoing calls the trunk line is through-connected towards the digital system telephone and the extension user can dial the external number directly without route access code.

A typical beneficiary of the *exclusive external line* facility is an executive with many external calls and a secretary serving as divertee position, fig. 14.

Common bell call pickup (or line pickup group). Instead of a common bell a function field on the digital system telephone is used. Each digital system telephone can serve up to five of these groups. As long as a call queue exists this fact is indicated on the appropriate LED on all digital system telephones, maximum 30, in which the group is represented. On answer the call is switched to the answering extension's *triple access line*. This facility is used e.g. by travel agencies for booking tickets, fig. 15.

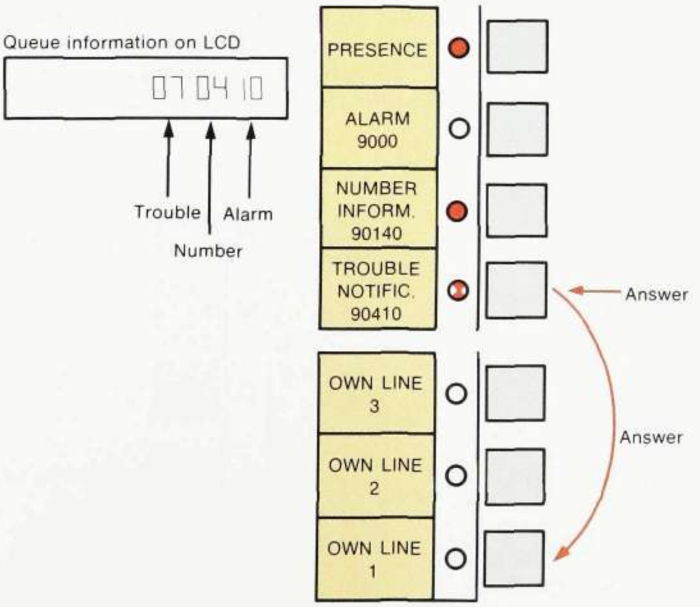
Voice paging facilitates loudspeaking messages via a defined group of digital system telephones. The same facility exists in intercom systems. Short messages can be issued e.g. "Visitor for Mr. Smith at the reception desk".

The paging call is transmitted to the loudspeaking digital system telephones that are programmed for this facility and whose voice channel is free. Paging can be combined via an optical call indication on the appropriate LED of those digital system telephones that have this facility. Up to 30 digital system telephones can be members of each group.

On depression of the button to initiate voice paging a tone burst with special character is issued for one second, whereafter the pager leaves a message. The pager then releases the button in order to disconnect the microphone. The call lasts for 20 seconds. During this period an extension in the group can answer by pressing the appropriate button. The call that is thereby established seizes the *triple access lines* of both parties, fig. 16.

Automatic call distribution is similar to *common bell call pickup*, but the call is routed to one extension only. Up to five groups with a total of 30 extensions can exist. Each group has a call number. Extensions can be members of several groups. An incoming call is distributed to the group member, extension, indicated as on duty and who has been free longest.

Fig. 17
Automatic Call Distribution (ACD) facility



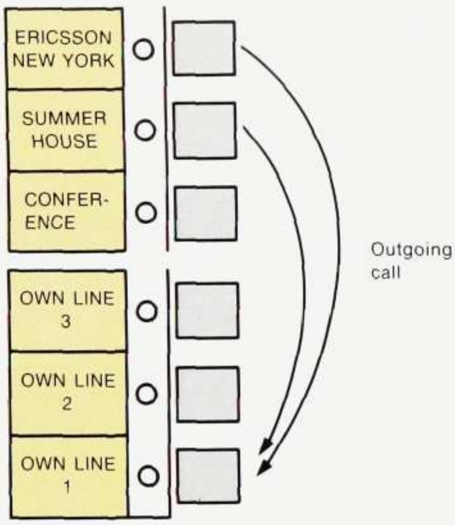


Fig. 18
Single button access facility

The queue lengths of those groups represented on the digital system telephone of a free, on duty extension are shown on the appropriate LEDs. The extension user can indicate himself on duty in any number of the groups in which he is a member. The on duty state is indicated by a steadily glowing LED and a call by e.g. a rapidly flashing LED. On answer the established call seizes the *triple access line*. Fig. 17 provides an example of LED information. This extension should switch immediately to the on duty state for the alarm number as this queue has grown.

Single button access functions as an individual abbreviated number with up to 20 digits. The extension user programs the number information with the digital system telephone in the programming mode. The stored number information can also constitute a prefix or suffix digit or entire procedures with * and # to access facilities more simply. Outgoing calls initiated with *single button access* seize the *triple access line* or the traffic facility that has already been activated and dial tone is obtained on depression of the single button access button, fig. 18.

All function fields in the digital system telephone have the *single button access* facility as basic function, before they are assigned any other traffic facility from the I/O-terminal.

Other new facilities

Group parking with call pickup function means that a call can be parked by one digital system telephone and admitted to another in which the facility is represented, like *line pickup* or another extension user's *exclusive external line*.

Immediate voice connection (or handsfree automatic answer) is a facility that the extension user can program from the digital system telephone. Each traffic facility can be assigned this facility individually, which is particularly suitable for *dedicated intercom link* and the user obtains facilities similar to those available with intercom systems.

Diary shows the year, month, day, hour and minute on the LCD when the diary button is pressed.

Message waiting is a facility, that for the called party, stores the numbers of those extensions that have placed a callback order against the called party's *triple access line* after meeting no answer. The message waiting LED lights to show that one or more stored messages exist. On return an extension user whose message waiting LED is lit can glance through the list of stored numbers to learn who has called, and automatically call these parties, one at a time, by pressing the *message waiting* button.

External number redial. An extension user can store the dialled external number by pressing the function button and transmit this number at any time by pressing the same button.

Programming mode. Extension users can place the digital system telephone in the programming mode and in this state check and program their *single button access* numbers and call alternatives for other traffic facilities.

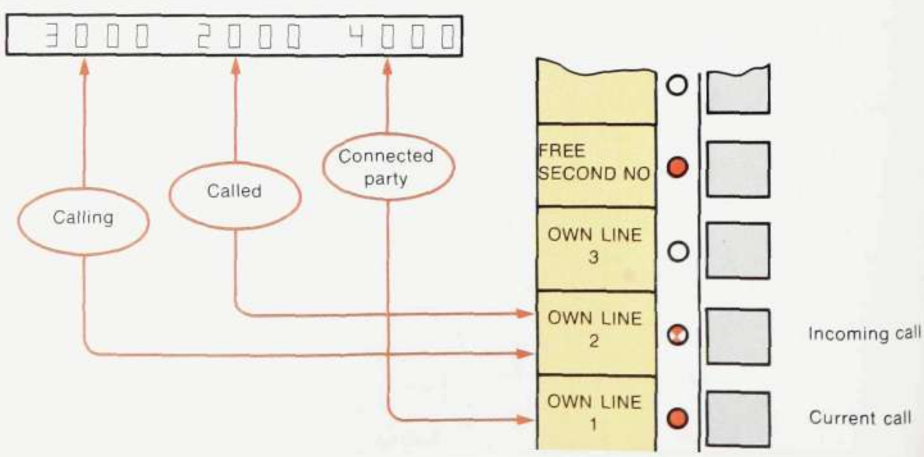


Fig. 19
Divertee position facility

Dedicated function fields

Certain facilities, that in conventional telephones require a complicated execution procedure, have been given a simpler routine in the digital system telephone by assigning the facility a dedicated function field.

Automatic callback is assigned a function field from which automatic callback is initiated and answered. The function button LED glows steadily as long as any callback remains. On an incoming callback the LED flashes for a call. After answer the conversation is held on the *triple access line*.

Diversion is assigned a function field, from which "diversion direct" and "diversion follow-me" are ordered and cancelled. The function field LED glows when the digital system telephone is diverted.

Applications

By assigning digital system telephones different combinations of the facilities described above it is possible to create different application systems, that can be optimized to the traffic requirements of individual users. Some application examples follow below.

Divertee position

Effective supervision of a large number of extensions in the diverted state is obtained by equipping the supervision position, divertee position, with one or more digital system telephones. Both caller and called party can be identified on the LCD. Simple parking and extending procedures are obtained with function buttons, and an even distribution of traffic for several telephones is achieved by extension group hunting or *common bell call pickup*, fig. 19.

The MD 110 digital system telephone can also be used as a divertee position for up to 32 extensions, where every extension is represented by a function field. The associated LEDs provide continuous information on the state of each diverted extension, e.g. incoming call, busy, parked. The ring signal for the lines of the diverted extensions can be delayed so that the divertee position receives a signal only if the call remains unanswered after a predetermined period.

Executive/secretary facility

This facility is built up by supplementing the divertee position facility with *dedicated intercom link* and *line pickup group* for those extensions for which

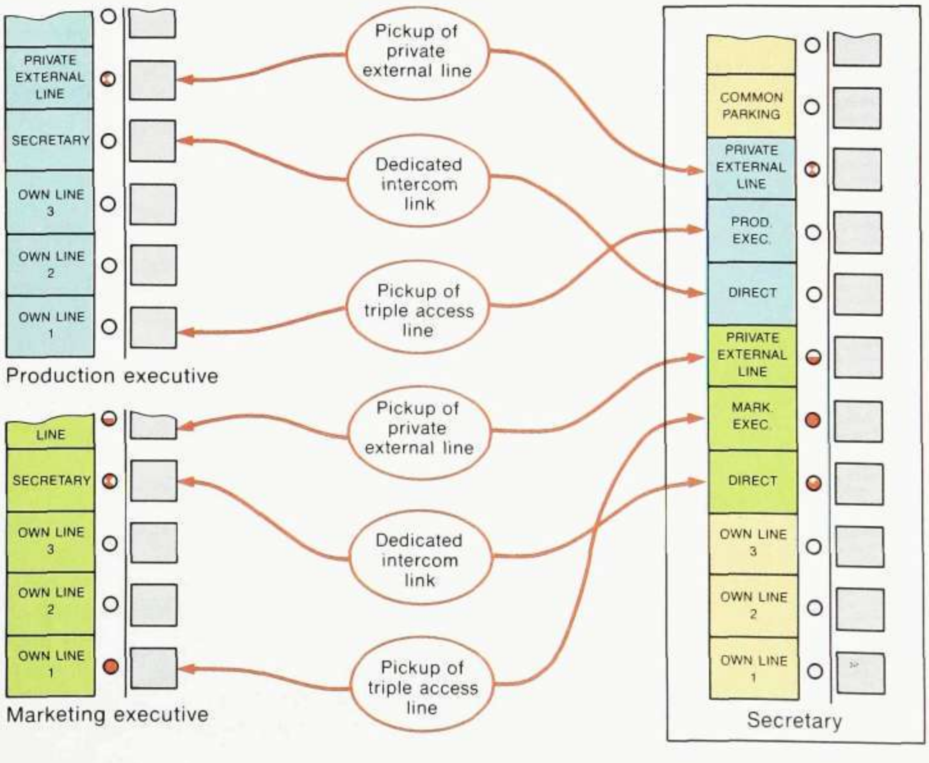


Fig. 20
Executive/secretary facility

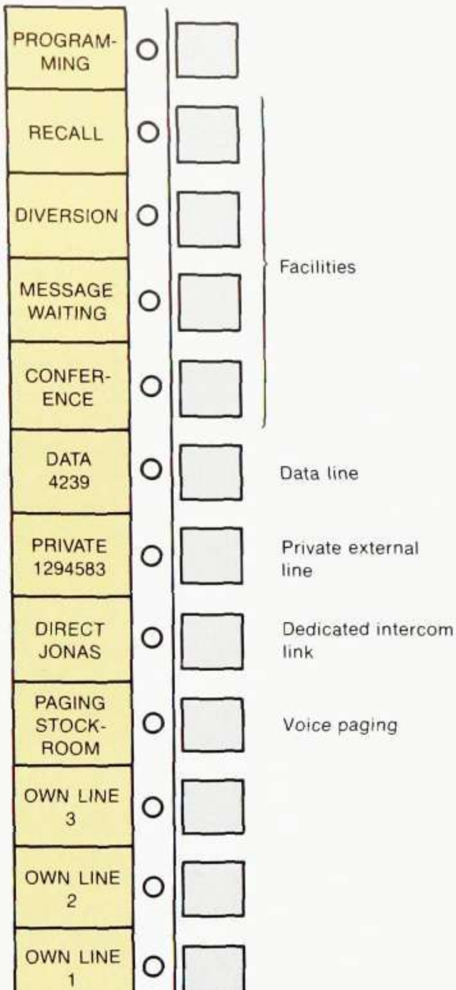


Fig. 21
Conventional extension telephone facility

special handling and supervision of traffic is desirable. This gives a secretary access to an executive's *triple access line* and *exclusive external lines*. The secretary can answer incoming calls, establish an inquiry call to the executive with the *dedicated intercom link* and then transfer the call if so desired.

Ordered, outgoing calls can be transferred easily using this range of facilities, fig. 20.

Intercom
By combining the *dedicated intercom link* and the loudspeaking *immediate voice connection* facilities an intercom facility for up to 32 extensions is obtained. Several extensions can be called simultaneously using the *voice paging* facility, fig. 12.

Line pickup group
A line pickup group is formed through the representation of *exclusive external line*, *triple access line* or *extra line* in several digital system telephones. Each digital system telephone in the group has each line represented by a function field. The extension user obtains a continuous picture of the current traffic situation in that the state of the lines is displayed on all telephones in the group.

An incoming call is thus signalled to the entire group and can be answered from any telephone. After answer the line is indicated as busy on the other telephones. The call can now be transferred to another extension inside or outside the group, or parked with the indication "available for group pickup", whereafter any extension in the group can answer the parked call.

Voice paging can be utilized and the person answering a call can inform the group which member is being paged.

Normal office extension
The digital system telephone is used as a normal office extension primarily to simplify communications. By programming the *single button access* button of the digital system telephone with procedure codes it is possible to utilize a number of extension facilities merely by pressing a single button.

The majority of the traffic facilities available to the digital system telephone can be used to advantage by a normal office extension. Particularly beneficial is the *data line* facility, that is not available to extension users with conventional telephones, fig. 21.

Operation and maintenance

One great advantage provided by the two wire connection is that the existing line network can be used when a digital sysem telephone is introduced. The conventional telephone is unjacked and the digital telephone is jacked in. On the PABX side it is necessary to switch from analog line circuit to digital line circuit in the main distribution frame, fig. 2.

Each digital system telephone can be "tailored" for its user's requirements, i.e. via commands from the I/O-terminal it is possible for the user to gear to his own telephone those traffic facilities he requires. The digital system telephone is prepared for data communications and has a jack at the back for connection of an adapter for a data terminal.

The programs for the traffic facilities are stored in the PABX; only the market-dependent cadences for tone signals and for function field LEDs are stored in the memory of the digital system telephone. The telephone can be disconnected and reconnected or moved without losing its programmed traffic facilities.

Digital system telephones are fed from the PABX, and they can therefore also function during loss of the mains voltage.

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Data Communications in MD 110

Gregory Barnicoat, Lars Boman and Olof Ulander

MD 110 is a PABX for 100 to 10,000 extensions. It fulfils traditional requirements as regards telephony facilities and it is also an effective instrument in the increased use of data communications.
The authors describe how the conventional two-wire PABX network with MD 110 is made accessible for data and text communications also. Several of the facilities designed for data users are described and examples are provided showing how it is possible to realize some common applications with these facilities. A central feature in the utilization of the data facilities is the design of user facilities and operation and maintenance functions. These are therefore dealt with at length.

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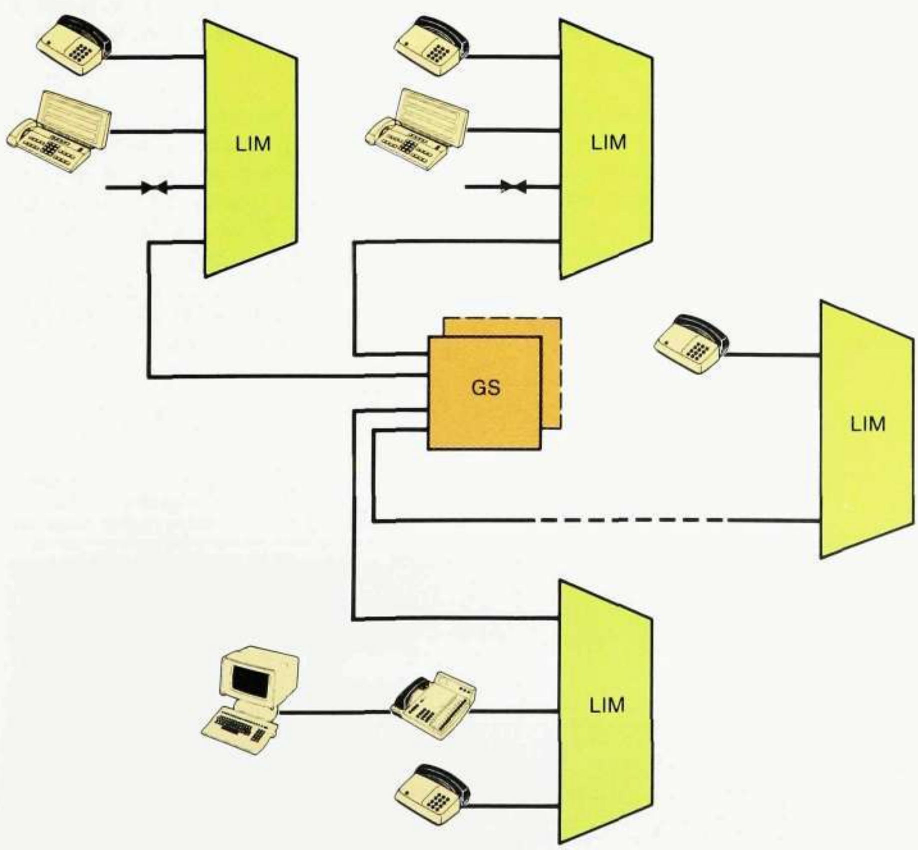
Office routines claim an increasingly larger slice of a company's outlay. Work can be rationalized and costs cut by utilizing various aids, available thanks to the rapid developments in different technical spheres.

Among the various aids are machines for duplicating, telex and word processing and also computer programs for e.g. wage routines and stock-keeping. They are all aimed at providing better solutions to the most acute problems and each problem has been optimized individually.

Fig. 1
MD 110 structure



The rationalization and data processing instances within a company plan for continued and increased use of computerized aids. More and more office staff are given access to data terminals with which to pursue their duties. As a consequence, demands increase for effective administration and maintenance of the terminal network. In the long run it is not possible to supply a unique solution for each problem as continued rationalization also demands increased integration of the work process[1].

Demands for a terminal network which is more general in application increase also as it is desired to access more users to more facilities and to utilize expensive equipment mutually with other users.

The increasingly varying requirements in respect of data management and data processing make it more and more difficult to obtain complete system solutions from a single supplier. Effective integration of systems from different suppliers presupposes general usage of standards for functions and interfaces. Intensive efforts are also in progress within national and international standardization organizations and are successively providing better and better possibilities for interconnecting the various systems. It must therefore be possible continuously to adapt the systems introduced today to later developments and changes in standards. An essential requirement is that of effective maintenance. The most reliable system can be hit by faults, but the system must be designed so that the effects of the faults are limited, and possibilities must exist for rapidly locating and eliminating those faults that occur.

The solution to data communication problems offered by a modern PABX, MD 110, from Ericsson Information Systems, is described in this article.

Technical description

MD 110 is a digital microprocessor-controlled PABX with modular building blocks[2]. It consists of line interface modules, LIMs; in large installations interconnected via a central group switch, GS, fig. 1.



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A LIM has a time switch with 512 time slots for voice and data. Each time slot has a capacity of 64 kbit/s. Each LIM can be equipped with an arbitrary combination of line units for analog telephones and digital system telephones[3], PABX operator consoles, data terminals and lines towards the public network. A LIM can either function autonomously or interwork with other LIMs via 32 channel PCM links, directly or via GS, that is controlled by the connected LIMs. GS has a modular structure and can be dimensioned for up to 248 PCM links, each with a transmission speed of 2.048 Mbit/s.

Digital system telephones, PABX operator consoles and data terminals are connected to a digital line unit board, ELU-D, that possesses eight line circuit functions and occupies 16 multiple positions in the switch, fig. 2. Normally eight time slots are used for voice channels and eight also for data channels.

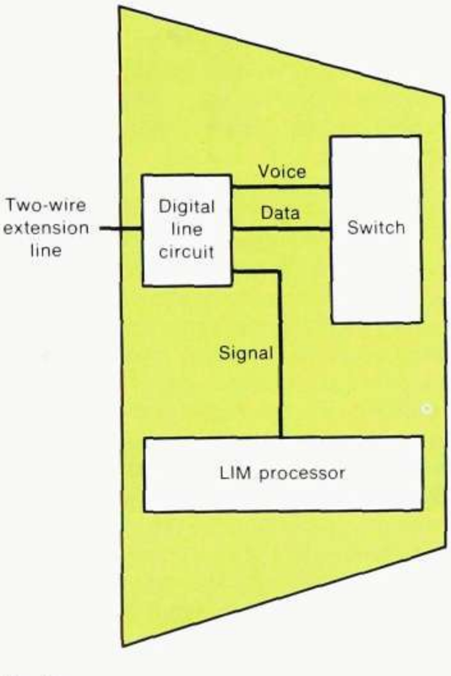


Fig. 2
Voice and data channels in a LIM

Two-wire extension lines are used. Digital units have a 96 kbit/s duplex connection with the PABX, achieved by time division multiplex burst signalling. The bursts are transmitted at a speed of 256 kbit/s. They comprise 12 bits and carry digital signals as shown in fig. 3.

The bit flow is divided into four channels, 8 kbit/s are used for synchronization of the burst signal. A channel with a capacity of 64 kbit/s is used to transmit PCM voice or highspeed data. A channel with a capacity of 16 kbit/s is used for data transmission and 8 kbit/s for control signalling between digital units and LIM.

The data channel

The format in the data channel shown in fig. 4 is used to transmit user data through MD 110.

The format is utilized both for data transmission and to balance the speed between the connected terminal and the data channel. Empty frames are sent as filling, fig. 5, to balance the speed.

The format facilitates asynchronous and synchronous transparent transmission of data. Modifier bits are used for interface signals and some test and maintenance signals.

Terminal connections

Data terminal equipment, DCE-T, that is connected directly to the rear of the telephone, fig. 6, is used to adapt the digital system telephone to different physical and logic terminal interfaces.

The bit flow from LIM is accepted and sorted in a customer-adapted LSI circuit. The 64 kbit/s channel proceeds via PCM coders and filters to the analog voice section and the 8 kbit/s channel leads to the microprocessor in the telephone. The 16 kbit/s channel leads to DCE-T and an LSI circuit for formatting data, fig. 7.

Data circuit terminating equipment, DCE-S, fig. 8, that also accesses the 64 kbit/s of the voice channel for data communications, is available for those application cases in which voice communication is not required.

The digital system telephone is fed current directly from LIM which guarantees

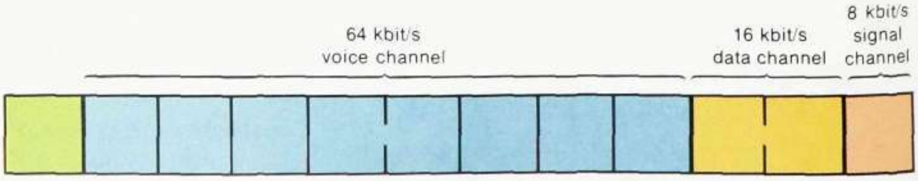


Fig. 3
Channel division for a digital signal burst

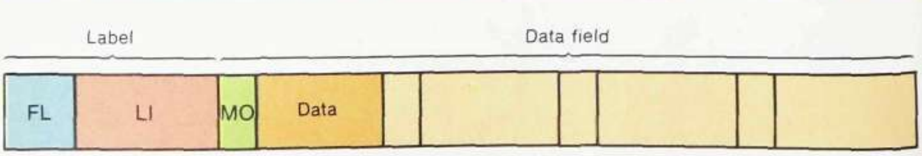


Fig. 4, below, right
Data channel format

FL Flag
LI Length indicator
MO Modifier bit

Fig. 5, below
Data channel, basic state that is transmitted as filling

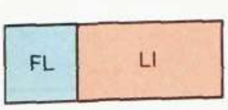
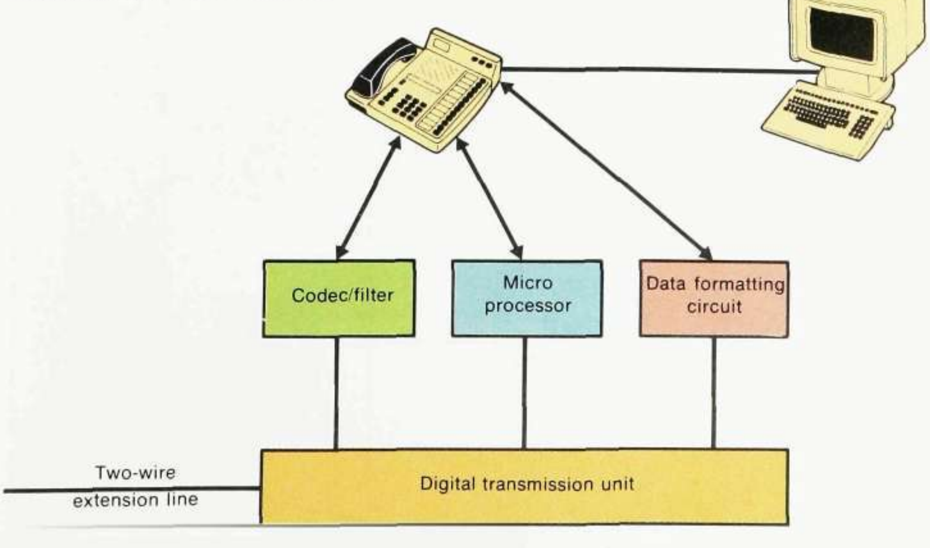


Fig. 6
Data terminal equipment DCE-T connected to DIAVOX Courier 700

Data transmission modes:	Asynchronous or synchronous, full duplex or half duplex
Data speeds, bit/s	
– asynchronous only	110, 150, 200, 300, 1,800
– synchronous or asynchronous	600, 1,200, 2,400, 4,800, 9,600
– ditto, for DTE-S only	19,200, 48,000
Character length, bits:	
– synchronous only	7 8
– reduced speed	9
Number of stop bits:	
– asynchronous only	1, 1.5, 2

Table 1
DCE-T and DCE-S interface characteristics

Fig. 7
Various adaptation circuits for voice, signals and data for the digital system telephone



the telephony function even on mains failure. DCE-T and DCE-S are connected to mains.

Connection of modems
External, permanent data connections are established via modems to special line units, *Modem Gateways*, MGs, in MD 110. A number of MG units can be assigned a group number that is utilized when establishing data connections from the terminals connected to the PABX.

If the data traffic does not motivate permanent connections the requisite quantity of modems can be connected to an independent pool. MG units are used in this case also for connection. For an external data connection a modem of the correct type is connected to the selected external connection. This means that the lines used for the connection can alternately carry voice and data and thereby be utilized more effectively. By employing modems as a common resource their number can be reduced. The modems can be located centrally whereby maintenance becomes simpler and more effective, fig. 9.

Terminal and modem interfaces
The data communication facility in MD 110 is designed for the connection of synchronous or asynchronous data terminals and modems using V.24/V.28 or V.36 interfaces (RS 232C and RS 449). Terminals that are used in parallel with common telephony facilities and consequently are connected to a DCE-T can utilize maximum 9.6 kbit/s. Higher speeds are achieved by connection to DCE-S which also permits the standard data speeds of 19.2 and 48 kbit/s.

The interface characteristics of DCE-T, DCE-S and MG are stated in table 1.

Transmission
Data traffic from one LIM to another within MD 110 is distributed on free channels in interjacent PCM links in competition with the telephony traffic. As each LIM can be connected to GS by up to four PCM links, each with 30 channels utilizing 64 kbit/s, large volumes of data traffic can be handled.

Facilities and fields of application

Facilities
A large number of application fields can be covered with a number of basic facilities. The most important facilities are:

Non-dialled connection to predetermined answer position
One requirement is that data connections shall only be established when needed. This reduces the seizure of lines and the need of entries to the computers. For this purpose most terminal interfaces towards DCE contain a call function that in MD 110 initiates automatic connection towards a predetermined address. Connection can also be initiated via a button on DCE.

Transfer of computer applications
During operation at computer centres it must be possible to move individual applications from one computer to another. MD 110 permits addressing with a unique address for each computer application. All applications in a certain computer are accessed via the same group of exits in MD 110. As each application address in MD 110 can be handled individually it is a simple task for the computer centre maintenance staff, via commands, to reroute a call from one group of exits to another when an application is moved.

It is thus not neccessary to inform or disturb users when moving computer applications.

Fig. 8
Data circuit terminating equipment DCE-S

Single button access
The majority of terminal users work towards a small number of computer applications. These users can utilize the telephony facility "single button access". The same group of computer entries can be called using several single button access addresses, corresponding to the program applications in the computer. The terminal user can program the addresses of the commonmost data applications. Each such internal or external address is assigned its own function field on the digital system telephone and on DCE-S.

Abbreviated dialling
When more single button access addresses are needed than those for which the selected digital system telephone has been prepared, the abbreviated dialling facility can be used. Each terminal user can define an abbreviated number series for internal and external addresses.

Manual dialling
The pushbutton unit can be used to dial the numbers of occasional addresses.

Automatic call re-establishment
There is an automatic supervision function for data units which lack the call initiation function in the interface and are not supervised manually. If an established connection is inadvertently disconnected this will be detected and an attempt is made automatically to establish a new connection to a predetermined answering position.

Automatic answer
The automatic answering facility in MD 110 can be used for units such as printers, memory banks and computers that cannot automatically answer calls.

Group hunting
This facility permits calls to be made to a number that is served by two or more switch exits. MD 110 offers several different methods for distributing the calls towards the switch exits.

Data line
It is possible to switch an established voice connection to a data connection by pressing a facility button. This can be appropriate when the A extension wishes to learn the data speed of the receiver, the B extension, in order to attain compatibility or during external calls to request the B subscriber in the public network to switch in a modem or, if the modem has the automatic answer function, to verify that the B side functions.

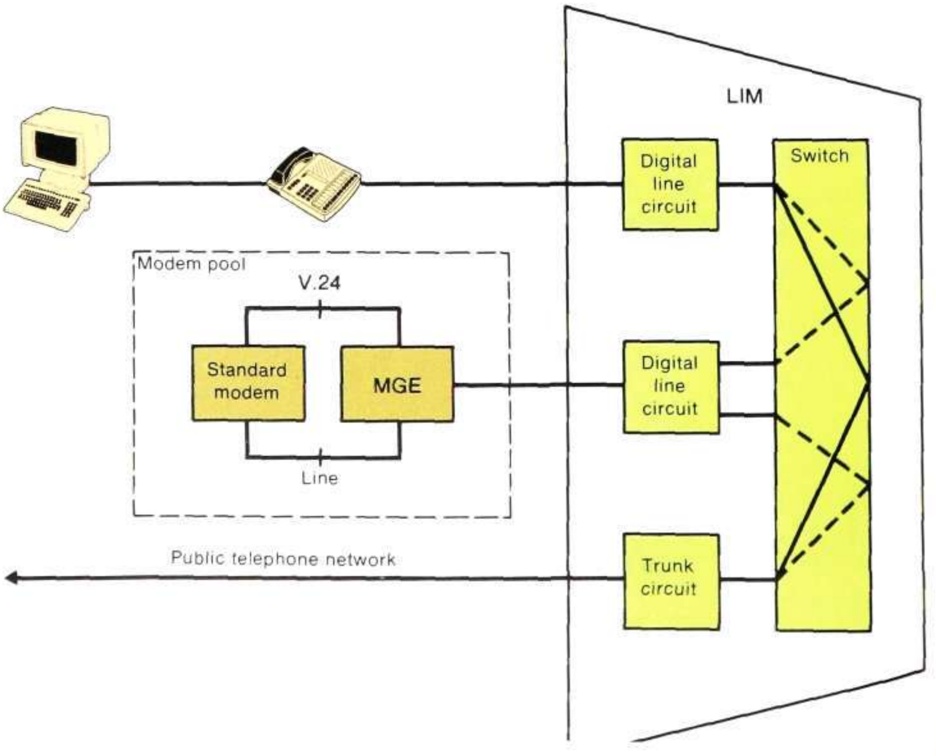
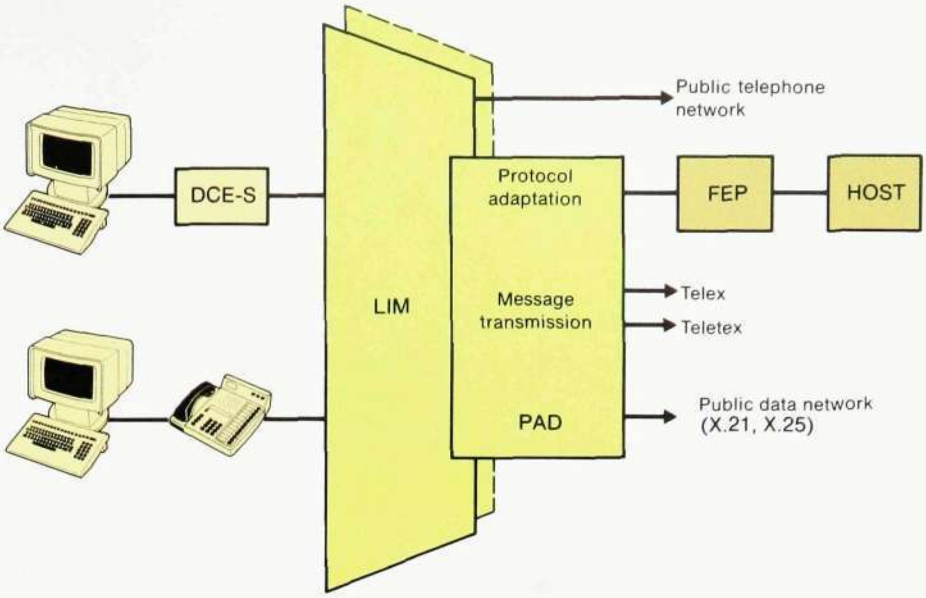


Fig. 9
Modem pool

Fig. 10
Overview showing MD 110 applications

DCE-S	Unit for direct connection of terminal to MD 110
FEP	Front end processor
HOST	Host computer
PAD	Packet assembly and disassembly
X.21, X.25	Transmission protocol recommended by CCITT



Traffic recording
Information about established connections can be supplied in reports with different sorting concepts.

Application fields
The possibility to connect data terminals to MD 110 provides the company and the individual user with numerous advantages. An example of this is the improved control and administration of the company's data connections provided by the functions of MD 110.

Even if each company's data communication frequently needs to have an individual character a number of requirements exist that are common to most companies. The realization in MD 110 of some of these is described below.

Terminal network
It is anticipated that the number of employees working with their own individual VDU terminals will increase by approximately 25% annually.

With MD 110 the opportunity arises to effectively connect inexpensive, character-oriented asynchronous or synchronous terminals. Via the PABX network for voice and data the user can gain contact with the desired internal or external computer systems, fig. 10.

The transparent data communications in MD 110 can be expanded to incorporate conversion functions. It is possible from start/stop terminals to emulate 3270 protocol for connection towards IBM's communications computers.

Electronic mail
Modernization of public networks for text processing also increases the demands for internal text communications.

Fig. 11 shows how a text communication system for telex and teletex can be connected to MD 110. The terminal users can write in and fetch messages from the text communication system via established connections. This type of traffic is characterized by short messages with relatively long intervals, i.e. the line concentration is highly effective.

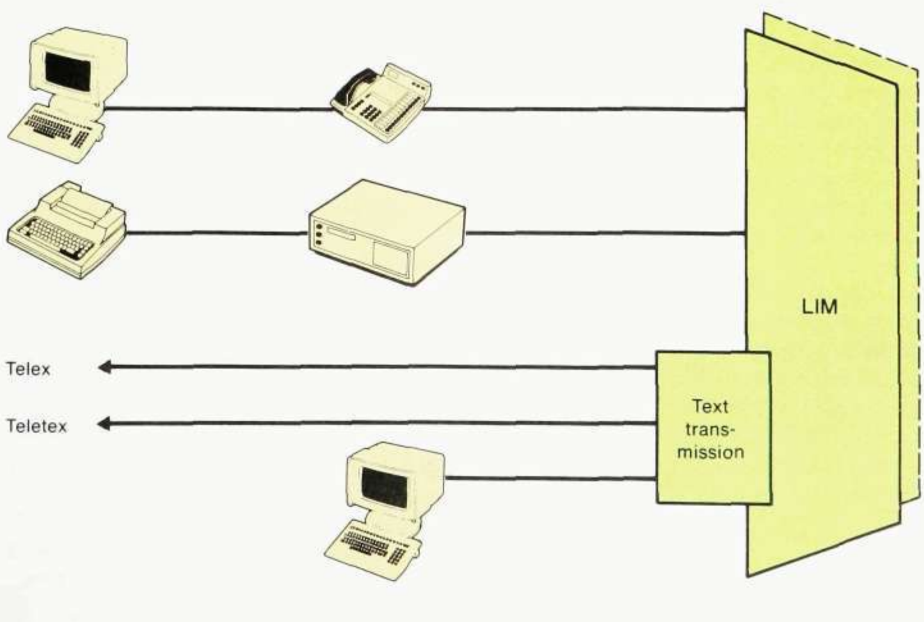


Fig. 11
Electronic mail

The heavy traffic into the departmental secretary for example can have permanent connections from the text communication system.

The text communication system requires fewer lines towards the public network and simple, inexpensive terminals can be used.

Common transmission resources
As the LIMs are connected via PCM links, by using 2 Mbit/s system lines it is possible to place them at any distance from GS.

The offices of large companies frequently have a wide geographical spread. By placing LIMs at the different locations it is possible to provide homogeneous, effective communications.

PCM connections established for voice communications can also be used for data communications. MD 110 offers controlled distribution of transmission resources for telephony or data traffic geared to the prevailing requirement.

For great distances it is often profitable to utilize data packet switching. This can be achieved by connecting a PAD or packet node function to MD 110, fig. 12.

User procedures

The procedures for data communications in MD 110 have been designed so that it is just as easy to establish a data connection as it is to make a normal telephone call, fig. 13. That it shall be easy for the user to alter the interface towards the terminal has also received great attention.

Overview
The procedures for data communications in MD 110 are based on the use of the digital system telephone together with DCE-T. One of the programmable facility buttons on the telephone can be assigned the data communication facility, *data line*. The associated light emitting diode indicates the various states of the data line.

When the data line button is pressed the pushbutton unit, facility buttons and digit display are associated to the data communication facility. All call progress for data communications is indicated visually. MD 110 permits concurrent telephony and data connections and they can be initiated without disturbing one another.

The user procedures for DCE-S and DCE-T are identical although the number of facility buttons can differ.

Outgoing calls
A call can be initiated either manually from the DCE, or from the terminal. Only calls to a predetermined address can be initiated from a terminal. From a

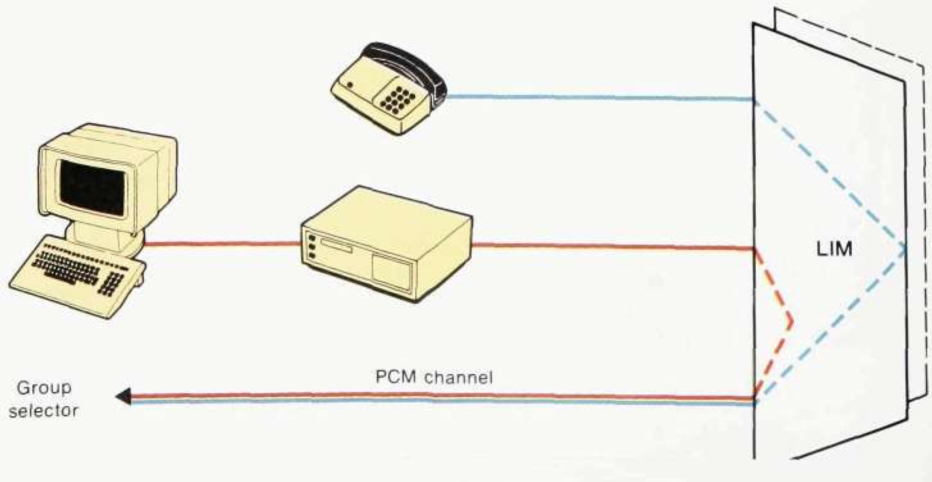


Fig. 12
Common transmission resources for voice and data for remote connection of LIM
Voice connection
Data connection

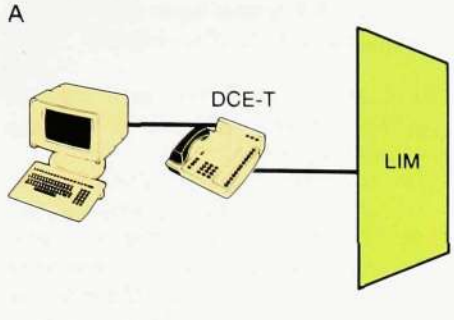


Fig. 13
Example showing how a connection is established between terminal and computer
A. The data line button is pressed to initiate a call, the LED shows dial state. The B number is dialled in the normal manner
B. LIM calls the computer
C. The computer or the computer's DCE-S answers the call and the data connection is established. The data connection can subsequently be disconnected by the calling or called party

DCE, where the facility or digit buttons of the digital system telephone are accessible, it is also possible to utilize the single button access and abbreviated dialling facilities. A call can be directed to any address, internal or external.

Incoming calls
Incoming calls are indicated in the DCE interface and visually on DCE. The call can also be indicated by ring signals in a telephone with DCE-T.

Answer
Calls can be answered manually from DCE or from terminals. DCE can also be programmed to provide automatic answer.

External traffic
For outgoing external data calls the data extension procedure is the same as for internal data calls except that the B number is complemented with the external call number. Standard procedures are used towards the B subscriber.

MD 110 adheres to standard procedures for incoming data calls. At the data extension the data call is indicated in the same manner as an internal data call.

Switching from voice to data connection
It is also possible to establish a data connection by first setting up voice connection and then switching to data connection. The switch is initiated by pressing the data line button and then the transfer button.

Signalling of states
All signals concerning states are indicated visually by light emitting diodes, LEDs, or on the digit display. The following states are indicated always:
- idle
- connected
- busy, congestion
- camped-on
- incoming calls
- data connection established.

LEDs also exist for current feed, testing, transmission of data and reception of data.

More information about the type of fault message and test result can be displayed on the digit display on the digital system telephone.

Programming the interface
The interface towards the terminal is programmable. Programming is achieved using simple commands from the telephone in the same manner as when the user programs her/his own abbreviated number. It is also possible to program interfaces from a maintenance terminal. Commands are used to initiate presentation of the relevant interface state.

Reliability

Development of MD 110 follows the tradition created for the development of Ericsson's AXE 10 system for public telephone exchanges. MD 110 and AXE 10 have mutual design rules and design aids for both hardware and software, which guarantees high reliability at device level.

At system level the modular basic structure of MD 110 guarantees high reliability through the possibilities for the distribution of critical connections, dimensioning of the permissable number of functions depending upon one fault and duplication of functions.

Some examples:
- If for any reason a LIM is isolated from GS the former can continue to process traffic between connected extensions, connections with the public network and computer connections.
- If a catastrophic fault knocks LIM out of action only those extensions and lines connected to this particular LIM are affected. Important routes can to advantage be distributed among several LIMs. In situations that dictate greater reliability than normal the common functions in LIM can be duplicated.
- The group switch function in MD 110 is duplicated. Transmission to each LIM can also be duplicated in order to increase accessibility. Connections from one LIM to another are then established simultaneously via different PCM channels and group switch units. The receiving LIM compares the signals from the two PCM channels and a fault is thus quickly identified.

Operation and maintenance

Data communication as a facility in a PABX creates the prerequisites for an effective, easy to use maintenance system. MD 110 provides improved operational reliability, co-ordinated fault localization and elimination, and simplified administration in the case of removals for example.

The operation and maintenance functions in MD 110 are common for voice and data communications with the exception of special functions for the supervision of DCEs and modems.

Common system functions are:
- automatic supervision of devices, with periodic function tests
- several alarm classes with issue of the alarm to PABX operator alternatively an alarm panel and alarm printouts
- duplicated central switching functions for automatic switchover in the event of a fault
- aids for effective fault identification facilitating replacement of faulty units
- traitional traffic recording and collection of traffic data statistics as basis for dimensioning and expansion
- advanced functions for administration of extension and PABX data
- maintenance interface utilizing standard man-machine language *(CCITT's Man-Machine Language)*.

These functions have been described in more detail in an earlier article[2].

The following operation and maintenance functions are specific for data communications:
- Data extensions are supervised continuously; when idle and when in the data transfer phase. Supervision embraces line connection, extension line and DCE function inclusive loss of power in DCE. Supervision is based on question/answer tests and supervision of the synchronization on the data channel.
- Statistical methods are used to supervise modems in the modem pool. The behaviour of a modem in respect of seizure time and call density is compared with the average behaviour of the modem group. Automatic, periodic routine tests also exist. Similar supervision exists for the modem connection (MG) as for DCE.
- Maintenance of terminals and modems is frequently the responsibility of another organization than that for the PABX. It is therefore important, by setting loops in terminal and modem interfaces, to be able to determine where a fault lies. Loop setting allows data units and modems to be tested independently of DCE and modem connection and MD 110 can be tested independently of data units and modems, fig. 14.

Loop setting can be established manu-

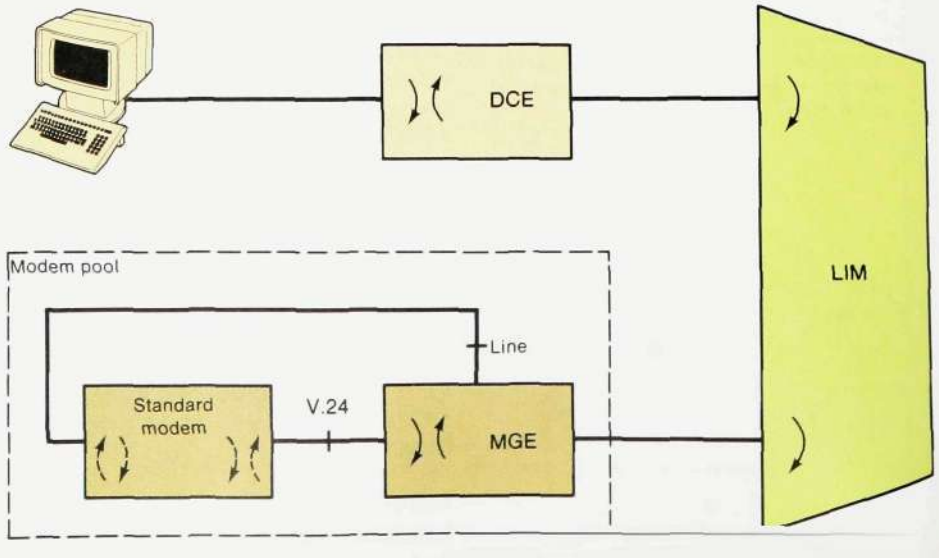


Fig. 14
Loop connection for fault localization.
CCITT recommendations V.54 and X.150 are complied with for the positioning and activation of the loop connection points

Test loops in MD 110

Test loop in the standard modem (not always included) ally, from a pushbutton on DCE or MG, or with signals via the interface. A function check is initiated when a loop has been closed. Test results are displayed on LEDs and the digit display.

Alarms from data communication functions can be dealt with specially for alarm indication and recording purposes. In the event of a fault the data connection can be blocked without the extension's telephony function being disturbed and vice versa.

Summary

The transparent data communications in MD 110 present organizations with an internal data network providing the following possibilities:

Operation and maintenance
- Installation is flexible; a terminal can be moved from one extension to another rapidly and simply.
- Supervision is effective and also embraces the terminal.
- Data communications are operationally reliable.

Better utilization of resources
- Several different applications and networks can be accessed from the same terminal.
- From locally situated computers it is possible to utilize common, large computer units for storage and print functions.

New forms of communications
- The system facilitates internal and external text traffic at economical prices for a very large number of users.

MD 110 is thus an instrument that can solve many of the problems existing in terminal networks today and it is also eminently suitable for future applications.

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MINI-LINK 15

Sivert Bergman and Kent-Arne Johnsson

MINI-LINK 15, which has been developed by the MI division, is a 15 GHz radio link for the transmission rates 2 and 8 Mbit/s, which correspond to 30 and 120 telephone channels respectively. The link can be used for transmission distances of up to about 25 km. MINI-LINK 15 is designed for simple replacement of modules, and its radio frequencies can quickly be changed on site. The equipment is designed for mounting on a mast and weighs only 25 kg including the parabolic antenna. The low weight and small wind load mean that the requirements as regards mast structure are moderate. MINI-LINK 15 is a competitive alternative to cable even for transmission distances of less than 10 km.

In this article the authors describe the electrical and mechanical design of the equipment, discuss briefly the service and maintenance aspects and give the result of a profitability study on MINI-LINK 15 in the junction network.

UDC 621.396.7

In radio link projects the cost of peripheral equipment, such as buildings, masts and power, is predominant. It is therefore an advantage if the radio link equipment can be made so light and robust that it can be mounted on masts together with the antenna. Access to new technology for increasingly high microwave frequencies makes it possible to realize such plans. This type of design eliminates the need for a special room for the electronic equipment, waveguides and pressurization equipment. The high frequencies also permit the use of small antennas, which reduces the wind load and hence also the demands made on the masts.

Fig. 1
MINI-LINK 15 mounted on a mast

Ericsson already manufactures MINI-LINK 10 and MINI-LINK 13. This article introduces MINI-LINK 15, fig. 1, which is a radio link having a modular structure and with the electronic equipment placed in a weatherproof case behind the antenna. All electronic units are of the plug-in type and are mounted in a BYB magazine. Both individual printed board assemblies and the complete magazine can easily be removed during service and maintenance work. It is also possible to install the magazine separate, for example in an equipment room.

The capacity of the equipment is set to 30 or 120 telephone channels by means of straps.

MINI-LINK 15 contains synthesizers for frequency generation. This makes it possible to select the desired radio frequency in the field, which is a unique feature in civil radio links. Thus the same type of link can be used throughout the network, which greatly simplifies planning, purchasing as well as administrating stores and spare part supplies. The frequency synthesizer gives the transmitted signal frequency crystal stability.

The radio frequency signal can be looped, which simplifies service and maintenance. Functional alarms are also available, both locally, in the mast mounted equipment and centrally, for example in an equipment room.

The installation work is made easy by the low weight, small dimensions and few connection points of the equipment. The baseband is connected via two coaxial cables, and the power and alarm outputs via a multi-pole cable with connectors.

Simple system designs and modern components have helped to limit the complexity of the equipment, which has had a favourable effect on both the MTBF and the hardware cost. One great contribution to the cost reduction was the choice of modulation method. The chosen method, frequency shift keying (FSK) with four frequencies, can withstand interference just as well as, for example, phase shift keying (PSK) between four phase states, but requires a much less complicated receiver, since the demand for coherent demodulation does not arise.

Electrical design

MINI-LINK 15 consists of the following electrical units (see also the block diagram, fig. 2):
- microwave unit
- intermediate frequency unit
- frequency unit
- baseband unit
- power unit.

The essential alarm points in each unit are connected to light emitting diodes (LEDs) on the unit fronts, which also contain analog test points for performance checks.

Microwave unit

The microwave unit, which contains all

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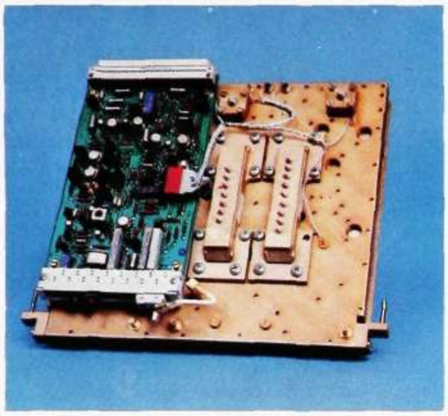
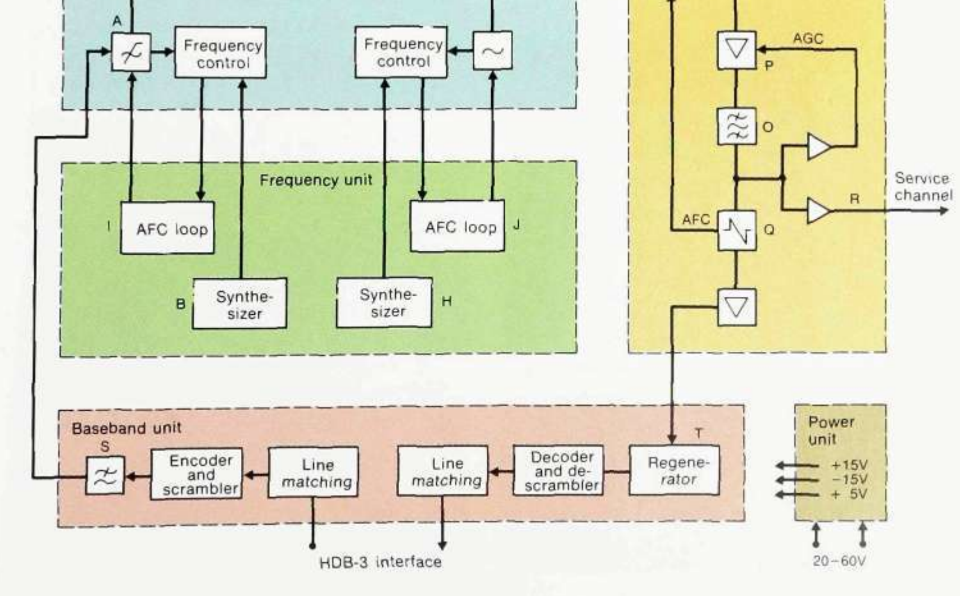


Fig. 3
The microwave unit

Fig. 2
Block diagram of MINI-LINK 15



main parts, a microwave unit with waveguide filters and a printed board assembly for current feeding and control of the microwave semiconductors, fig. 3.

The transmitter is direct modulated and consists of a cavity stabilized, voltage controlled oscillator (VCO) for 7.5 GHz, A, which is frequency locked to a certain channel frequency selected by the synthesizer, B, fig. 2. The signal is amplified and then multiplied to the 15 GHz band, and the final signal is fed via the send filter, C, and the antenna circulator to the antenna. In a similar way the signal received from the antenna goes through the receive filter, E, to the mixer, D. The send and receive filters have a bandwidth that corresponds to a frequency band in accordance with fig. 8, i.e. about 100 MHz.

The receive mixer, D, is fed with a local oscillator signal which is generated in a similar way to the signal in the transmitter. The mixer generates a first intermediate frequency of 227 MHz, which is processed in the following intermediate frequency unit.

In the microwave unit a digital service channel is added by means of amplitude modulation of the 7.5 GHz power amplifier, F, in the transmitter.

Another important function is performed by the built-in shift oscillator, G, which can be used to connect the send signal to the local receiver for fault tracing or performance tests. The shift oscillator contains an alarm function which indicates too low output power.

Frequency unit

The frequency unit contains the electronic circuits, B and H, required for generating the reference freuency for the transmitter and the receiver. The discriminator circuits I and J form part of the loops for the frequency locking of the oscillators in the microwave unit.

The frequency unit thus contains individual synthesizers for the transmitter and the receiver. The frequencies are selected by means of microswitches on the printed board assembly. The synthesizing is carried out using a common 10 MHz reference oscillator frequency.

A LED on the unit front gives an indication whether the transmitter frequency is locked to the reference oscillator or the frequency error is too great. The receiver synthesizer contains similar alarm indication circuits.

Intermediate frequence unit

In the intermediate frequency unit the received signal is filtered, level regulated and detected. Two-stage mixing has been chosen in order to obtain a high image frequency rejection while keeping the filter design simple. The 227 MHz input signal is filtered, in K, and level regulated before being mixed to the final intermediate frequency, 35 MHz. This mixing, in L, is carried out by a 262 MHz frequency and voltage controlled oscillator (AFC, VCO), M. The bandwidth of the receiver is set to 10 MHz by a 35 MHz filter, O, with high attenuation of out-band signals and high equalization of group delay. The intermediate frequency amplifiers, P, are regulated over a dynamic range of 50 dB, so that a constant level is provided for the limiter, Q, in the discriminator.

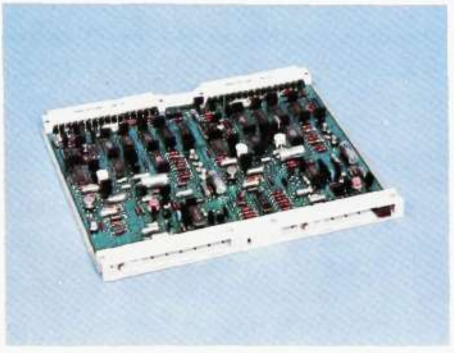


Fig. 4
The frequency unit

The FM discriminator, Q, which is of the quadrature detector type, provides both the detected baseband signal and the regulating voltage to the AFC system in the receiver. The digital, amplitude modulated service channel is extracted from the error signal of the loop for automatic gain control (AGC), R, whereas the d.c. component is used for the control function and thus provides a measure of the received signal level.

The test points for both the AFC and AGC are placed on the unit front and are thus easily accessible during installation and fault tracing. An alarm is obtained if the received signal level is too low.

Baseband unit

In the baseband unit the signal from the multiplexing equipment is regenerated, and the HDB-3 code is converted to NRZ code. A scrambler is used in order to obtain a noise-type spectrum and to ensure clock regeneration on the receive side. Finally the signal is pulse shaped in a Bessel filter, S.

The received signal is demodulated in the intermediate frequency unit, T, and then processed further in the baseband unit. The signal is filtered before threshold detection is carried out. The amplitude and d.c. component of the signal are regulated in two loops. The signal is then regenerated in a phase locked loop, after which it is decoded, passed through a descrambler and HDB-3 encoded.

Fig. 5
The magazine with the printed board assemblies installed

The baseband unit gives two independent alarm signals, one if there is no input signal in the baseband and another if the bit error ratio of the received signal is too high.

Power unit

The electronic units of the radio link require ±15 V and +5 V. These stabilized voltages are obtained through d.c./d.c. conversion of the primary voltage. The d.c./d.c. converter has been designed to meet the following requirements for several different installation variants:
- optional polarity
- complete earth insulation between the primary and secondary voltages
- input voltage range 20–60 volts
- high transient resistance.

The regulated output voltages are supervised and protected by
- short-circuit protection
- overvoltage monitor
- under-voltage monitor
- excess current alarm.

High efficiency was another important design requirement, and a level of 70 % has been achieved. It has thereby been possible to reach the set target of an overall power consumption of 20 W for MINI-LINK 15.

Mechanical construction

The chassis of the radio link equipment consists of an extruded aluminium profile, which also functions as a cooling flange. On this profile a magazine is mounted which holds five printed board assemblies and one microwave unit with the transmitter and the receiver. All units are designed for the BYB construction practice. The printed board assemblies, fig. 4, which measure 222×178 mm, are plugged into connectors at the rear of the magazine, fig. 5. The lower edge of the rear plate holds connectors for connection to the baseband, power feeding and supervision circuits. The chosen construction practice facilitates service and fault location.

The magazine is covered by a housing made of integral cell plastic, with an

Frequency band F

Frequency MHz		Channel number
15138	–	G1
15131	–	F29
15127.5	–	F28
15124	–	F27
15120.5	–	F26
15117	–	F25
15113.5	–	F24
15110	–	F23
15106.5	–	F22
15103	–	F21
15099.5	–	F20
15096	–	F19
15092.5	–	F18
15089	–	F17
15085.5	–	F16
15082	–	F15
15078.5	–	F14
15075	–	F13
15071.5	–	F12
15068	–	F11
15064.5	–	F10
15061	–	F 9
15057.5	–	F 8
15054	–	F 7
15050.5	–	F 6
15047	–	F 5
15043.5	–	F 4
15040	–	F 3
15036.5	–	F 2
15033	–	F 1
15026	–	E29

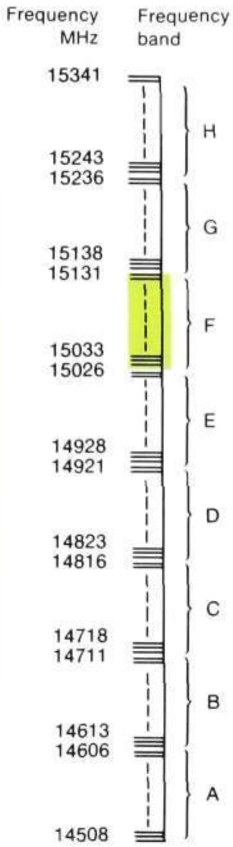


Fig. 8
Frequency plan for MINI-LINK 15

The parabolic reflector is also mounted on the aluminium profile. The antenna feeder is removable and during transport it is stored in a clip on the side of the magazine. The equipment is very robust and can be handled even on high masts.

The equipment casing is airtight, but the mounting pipe provides an opening to the surrounding air. The temperature differences that occur cause the pipe to function as a moisture separator. This gives a favourable climate for the electronic equipment, as regards moisture as well as temperature. The cooling flange is protected against sun radiation by both the parabolic disc and the insulated housing, which makes it possible to use MINI-LINK 15 even in a desert climate.

The antenna is aligned with the aid of the adjusting devices shown in fig. 6. The antenna lobe angle is only 2.3°, but the design is such that it is nevertheless easy to align.

As an alternative the antenna can be installed separated from the electronic's case when such mounting is considered more suitable, or when a larger antenna is required. In such cases the electronic part can be mounted in the ordinary case for outdoor installation or it can be mounted indoors. In the latter case only the magazine is used, mounted on a wall or in a rack.

The equipment includes a tool kit which comprises all tools needed for installation, alignment and service. An installation kit is available as an accessory and it is sufficient for most types of installation.

Antenna

The antenna is fed from the front and has a parabolic reflector. This type of antenna has been chosen in order to obtain a simple mechanical construction together with high antenna gain.

The reflector is pressed from rigid 3 mm aluminium. It has a diameter of 62 cm, which gives an antenna gain of 37 dBi with a lobe width of 2.3°.

The swan-neck feeder with an R140 waveguide has special guide tabs for simple and quick installation and choice of polarization. It is transported in a clip inside the housing and is not fitted until the radio link equipment has been installed.

The circular opening in the feeder is protected by plexiglass.

Maintenance and fault finding

The maintenance aids have been designed to enable faulty units to be identified and replaced quickly and efficiently.

Transmission outage is indicated by a combined alarm. The faulty terminal is detected by means of a loop between the transmitter and the receiver (radio frequency loop). The alarm information is evaluated and the faulty printed board assembly or the whole magazine is replaced, depending on the type of alarm.

Alarms are indicated by LEDs on the printed board assemblies. The alarm signals are also available in a multi-pole connector and can thus be connected to other station alarms if required.

A faulty printed board assembly can be replaced easily and quickly. The low weight and small dimensions of the magazine facilitates the replacement of a complete magazine.

Fig. 6, left
Devices for aligning the antenna

Fig. 7
Installation kit, suitable for most installation cases

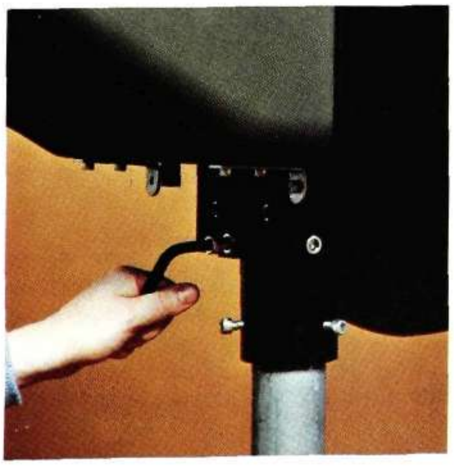
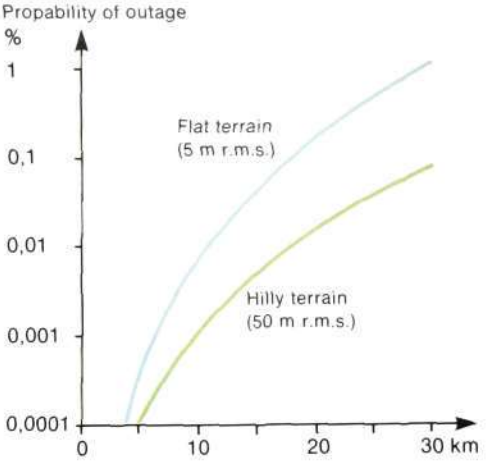
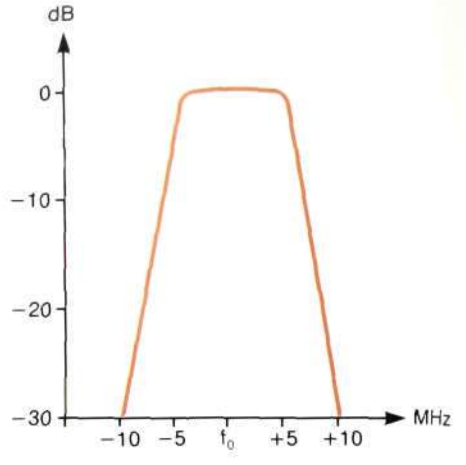


Fig. 9
The probability of transmission outage on MINI-LINK 15 with standard 62 cm antennas in the type of climate prevalent in Northern Europe. Both rain and multi-path propagation have been taken into consideration

Fig. 10
Measured power spectrum for 8 Mbit/s





The stocking of spares is simplified considerably by the possibility of selecting the desired radio frequency on site. Any pair of links can therefore function as spare for all the radio channels in the band.

Frequency plan

The ever increasing need for communications means that previously established frequency bands are often used to the limit. New digital systems and, above all, digital low capacity systems are therefore being planned for increasingly high microwave frequencies. The 15 and 18 GHz bands are about to be opened for use.

The 15 GHz band for radio links ranges from 14.50 to 15.35 GHz and thus has a bandwidth of 850 MHz. Discussions are now in progress regarding how this bandwidth should be used. The greatest possible flexibility was therefore aimed at when developing MINI-LINK 15. Fig. 8 shows a frequency plan with eight different frequency bands, but other frequency plans can be used. Both duplex distance and channel spacing are optional since separate frequency synthesizers are used for the transmitter and the receiver in the radio link. This gives complete freedom in the field to choose one of the 29 frequencies in each frequency band, which is an advantage from the point of view of both frequency planning and stocking of spare parts.

The microwave unit covers one frequency band for transmitting and another for receiving. The following combinations are offered:

transmitting	receiving
A	C
B	D
C	A
D	B
E	G
F	H
G	E
H	F

Other combinations are also possible, such as the combination A–H, H–A being discussed within CEPT.

Propagation

The availability of a link route is determined by the distance and the fading on the route, as long as the characteristics of eqipment remain the same. For correctly planned link routes, with a free first Fresnel zone around the line of sight between the antennas and without any dominant ground reflections, the amount of fading is determined mainly by two phenomena: multi-path propagation and attenuation caused by precipitation.

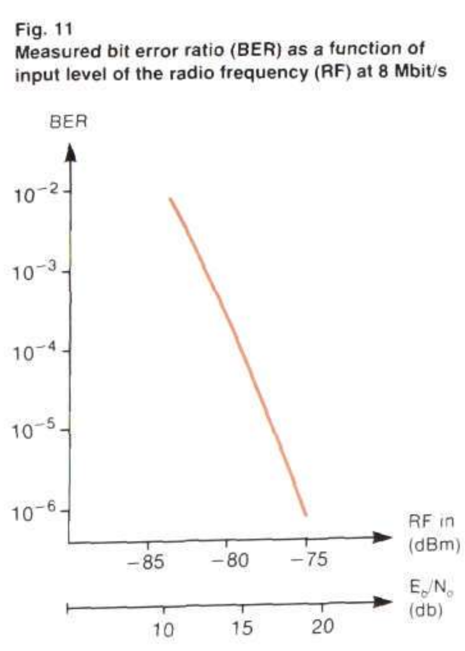
Fading caused by multi-path propagation is greatly dependent on the distance (d^3) but is only linearly proportional to the frequency. Thus the multipath fading on a 15 GHz link is only 3 dB higher than the fading on a 7–8 GHz link. The fading due to precipitation is far more dependent on frequency, and in areas with a high rainfall it can be the main cause of fading on a 15 GHz link. However, in most European countries it is nevertheless possible to bridge routes of up to 20 km using the standard 62 cm antennas, fig. 9.

Modulation method

In many countries the congestion in the various radio link bands has led to very stringent demands for narrow-spectrum modulation methods with good detection performance. The first generation of digital links, with phase shift keying (PSK) using PIN diodes direct on the final frequency is no longer a viable alternative for systems with a low capacity, since after-filtering with narrowband filters makes unreasonable demands on the Q value.

The problems of narrow-band microwave filters and the requirement that transmitters should be all solid state lead to another important requirement apart from the narrow-spectrum modulation, namely a constant signal envelope. Without a constant envelope of the

Fig. 11
Measured bit error ratio (BER) as a function of input level of the radio frequency (RF) at 8 Mbit/s

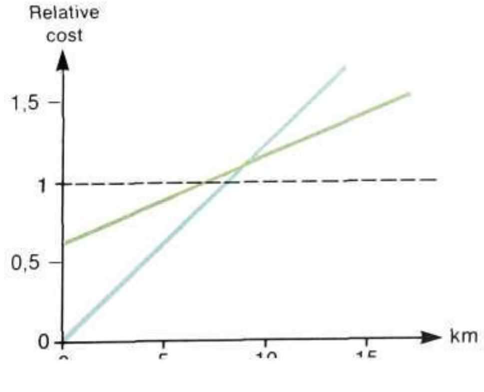


Technical data

Capacity	8.448 Mbit/s or 2.048 Mbit/s
Service channel	Digital AM up to 64 kbit/s
Output power	15 dBm
Frequency stability	±25 ppm
Noise figure	11 dB
Frequence range	14.50–15.35 GHz
Channel spacing	Controlled from the synthesizer, in steps of 3.5 MHz
Receiver dynamics	50 dB
Receiver bandwidth	10 MHz
Unwanted RF radiation	More than 70 dB below the transmitted carrier
Image frequency rejection	60 dB
Interface towards MUX	HDB-3 with 75 ohm impedance (±2.37 V)
Feeding voltage	20–60 V with optional polarity
Power consumption	20 W
Temperature range	−40° to +45°C ambient temperature + 1 kW/m² solar radiation
Weight, including parabolic antenna	25 kg
Dimensions without the parabolic antenna	370×350×650 mm
Alarms accessible externally	primary power AGC bit error ratio (BER) combined alarm
Commands accessible externally	transmitter off/on radio frequency loop
Test point accessible externally	radio frequency input signal level (AGC voltage)

Fig. 12
Cost comparison between different transmission media for 120 channels for new installations in the junction network

100-pair armoured buried cable, ∅ 0.7 mm
120 telephone channels on a new 10-pair cable (P-MOK) on an existing pole route (PCM)
MINI-LINK 15, including MUX, masts, installation etc.



deteriorate considerably during the passage through a limiting output stage, such as a semiconductor amplifier, because of the amplitude/phase conversion, AM/PM.

Frequency shift keying (FSK) with filtered modulation pulses has been chosen for MINI-LINK 15. Compared with phase shift keying (PSK) with four phase states, FSK has detection characteristics which are not quite as good, but it gives considerably better spectrum characteristics and less complex equipment. From other points of view, for example sensitivity to interference from other radio links, there are no significant differences.

Fig. 10 shows the resultant transmitted spectrum and fig. 11 the detection characteristics of MINI-LINK 15.

Costs

In many countries radio links have previously been used mainly in the long-distance network, where large transmission capacity has been required and the transmission distances have been comparatively large. The introduction of small and inexpensive links has now led to greater use of radio links in the junction network. The need of radio links in this network has been further accentuated by the introduction of digital technology.

Many of the old cables cannot be used for digital transmission. Since the labour cost is the major part of the cost of laying cable, and since this cost is approximately the same regardless of the capacity of the cable being laid, the cost per telephone channel becomes very high for low-capacity routes. The need for connections between terminal exchanges, group centres and zone centres is often in the range below 120 telephone channels, and low-capacity radio links have proved to be a competitive alternative to cables.

Fig. 12 shows a comparison of the cost of a 120 channel junction route when different types of cable and MINI-LINK 15 are used.

The link cost includes the cost of the links, two 30 m masts, PCM multiplexor, power, project planning and installation.

The comparison shows that MINI-LINK 15 is a more advantageous alternative than cable for transmission distances from about 7–8 km and upwards.

A statistical study of the Swedish junction network has shown that 30–50 % of the routes fall within the distance categories where radio links give a lower cost than cables. Today only a few percent of the junction routes consist of radio links, and there are considerable savings to be made in connection with new installations.

Summary

MINI-LINK 15 is an economical alternative to the installation of new cable as well as the introduction of PCM on old cables in the capacity range up to 120 telephone channels and for route lenghts of between 7 and 25 km. The equipment is compact, has few connection points and is placed together with the antenna, all of which makes the installation very quick and easy. Operation and fault location have been simplified by the modular structure and the possibility the equipment offers of selecting radio frequencies on site.

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AXB 30 – A Public Data Network System

Bo Ekström, Erik Hult and Gösta Leijonhufvud

The prerequisites for the public data network in the Nordic countries and for the development of Ericsson's system AXB 30 for this network have been discussed in a previous article in Ericsson Review[1].
In this article the authors describe the technical design and function of the system.

UDC 681.324

The rapid development in computer technology has brought an increasing demand for data communication. This is partly due to the great increase in data processing and partly to a successive change of the structure of computer systems towards geographical dispersal and distributed data processing. Moreover, new systems and fields of application, such as teletex, cash automats and cash points, have been introduced. System AXB 30 has therefore been developed to provide efficient data communication facilities in the form of a circuit switched network.

In order to meet the requirements of the users such a data network must offer a number of different transmission speeds, flexible switching procedure, high reliability and good transmission quality. Another factor of importance to the users is that the communication in the network preferably should be independent of their own procedures. A further prerequisite is that the system contains additional functions so that the users can, for example

- build up their own networks within the data network
- satisfy their security requirements
- make efficient use of their terminals and computers.

These requirements set a number of new problems for the Administrations in the differents countries. The rapid growth means that the data networks which are built must be easy to extend. It must be possible to add new functions quickly and easily as the need for new applications arises. The network must have internationally standardized interfaces and procedures in order to allow communication with other networks and countries.

An important economic requirement is that the number of operating staff required for the data network is low, to give a low operating cost. A data network normally covers a large geographical area, which is divided into a number of administrative regions. It must then be possible for the operation and maintenance of the equipment to be handled by the regions. This requires advanced operation and maintenance functions.

Fig. 1
The control room for AXB 30 at Hammarby, Stockholm, Sweden. Almost all operation and maintenance activities can be carried out from here

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Data network system AXB 30 meets all the demands made by the users which have been described here.

System characteristics

The basic characteristics of AXB 30 are:

- The network is synchronized and designed for circuit switching. The subscriber obtains fast and well defined transmission and also data transparency, which means that, once the connection has been established, the procedures used in the data transmision are optional.
- The data transmission in the network uses time division multiplex (TDM) and integrated switching and transmission equipment (IST), which ensures efficient utilization of the transmission equipment and the possibility of including maintenance of the transmission media in the system.
- The range of permissible transmission speeds for the subscriber equipments is from 50 bit/s asynchronously to 9600 bit/s synchronously.
- Short delay and switching times in the network.
- The data from the subscriber terminal are transmitted through the network in envelope form with 8+2 bits (8 data bits and two network bits). The two extra bits are used for signalling and for supervising the subscriber channel all the way out to the terminal.
- The design meets the applicable CCITT recommendations, which means that the network has standardized interfaces towards subscriber terminals and towards other networks (Fact Panel no. 2).
- The operation and maintenance functions are sophisticated and comprise all units in the network, including the transmission media. This means that the whole network can be managed efficiently with a limited number of personnel.
- The network can utilize existing telephone cable networks.
- The system has a modular structure which means that it can easily be modified in step with the technical development and the need for extension.

Network structure

Systems AXB 30 consists of a number of units which are connected together in accordance with fig. 2.

The data terminal, DTE, belongs to the subscriber and can be supplied by any of a number of computer and terminal manufacturers. Thus DTE does not belong to system AXB 30.

The connection unit, DCE, the data multiplexor, DMX/RMX, and the data concentrator, DCC, constitute the peripheral units of the data network, whereas the data network exchange, DSE, is the central unit that controls and supervises the operation of the network.

The transmission media used are normally telephone channels in carrier sys-

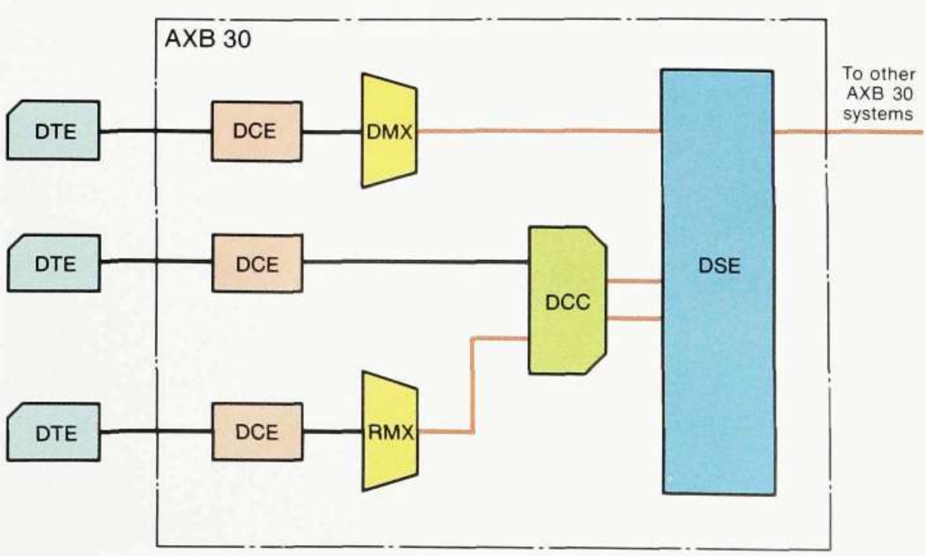


Fig. 2
The main units in system AXB 30

DTE	Data terminal or computer
DCE	Data circuit terminating equipment, connecting the subscriber's equipment to the network and placed on the subscriber's premises
DMX	Data multiplexor which is connected directly to the data network exchange
RMX	Remote data multiplexor, connected to the data network exchange via a data concentrator
DCC	Data concentrator
DSE	Data network exchange
▬▬	Subscriber line
▬▬	Multiplex circuit (64 kbit/s)

Fact Panel no. 1

Data format 8+2 and the associated multiplex structure

CCITT recommend two alternative envelopes, 6+2 and 8+2 bits. AXB 30 has been designed for the 8+2 format, and the associated multiplex structure is described here.

- An envelope consists of 8 data bits, one status bit and one synchronization bit.
- Once a connection has been completed, the data bits are entirely at the disposal of the terminal, and the network is wholly transparent to data as long as the status bit is equal to one.
- The status bit can be controlled by the terminal via special circuits in the terminal interface. The status bit is used for signalling various terminal states in the network, e.g. when terminating a call.
- The synchronization bit is used in the network to identify the data bits and the status bit, and for supervision of the subscriber circuit.

In order to be able to transmit data from the terminal at a certain speed, the network must operate at a transmission speed that is 25 % higher, in accordance with the following example:

Terminal bit/s	Network 8+2 envelope bit/s
600	750
2 400	3 000
4 800	6 000
9 600	12 000

The X.51 multiplex structure uses a transmission speed of 64 kbit/s. Of these, 60 kbit/s are used for data envelopes and 4 kbit/s for frame synchronization and control information.

The 60 kbit/s stream is divided into five groups of 12 kbit/s each. Each such group, in its turn, consists of either one subscriber channel for 9 600 bit/s, two channels for 4 800 bit/s, four for 2 400 bit/s or 16 for 600 bit/s.

The five groups in the multiplex structure can then be combined in an optional way, so that a total of 56 speed combinations can be obtained, in accordance with CCITT Recommendation X.54.

The X.51 multiplex structure is transparent, not only to subscriber data and status information, but also to the subscriber synchronization bit, which means that the synchronization at the channel level can be supervised centrally in the network and not only at the point where the subscriber line is connected.

The control information provides primarily alarm information, such as loss of synchronization in the opposite direction of transmission, but also control information for peripheral units like DMX and RMX, which have no control signal channel.

PCM systems, which are used as a transmission medium in the data network, are not synchronized to the data network. However, it is still possible to obtain synchronous operation by means of justification. This means that for each frame of 2 560 bits it is possible to add or remove a justification bit in order to compensate for the difference in transmission speed.

tems, PCM systems or four-wire circuits in telephone cables. Modems or other equipment for matching to PCM are included in the system.

The internal data structure in the network is based on the 8+2 envelope. All connections between multiplexors, concentrators and exchanges consists of one or more 64 kbit/s multiplex circuits with a structure in accordance with CCITT Recommendation X.51. Each such multiplex circuit transmits a number of traffic-carrying channels. The number of channels per multiplex circuit varies from five for 9 600 bit/s to 80 for 600 bit/s.

The various units in the network are clock controlled and synchronized in a hierarchic structure with DCE at the lowest level and DSE at the highest. A DSE, in its turn, can be controlled either from a reference clock, such as an atomic clock, or from another DSE via multiplex circuits. This flexibility makes it possible to adapt the network synchronization to different requirements.

Units

Data circuit terminating unit, DSE

DSE, which is placed on the subscriber's premises, has a standardized interface for connection to the subscriber's data terminal, DTE. DCE is therefore available in a number of versions for synchronous or asynchronous terminals with terminal interfaces in accordance with CCITT recommendations.

DCE is easily adjusted to the data speed of the terminal, fig. 3. The main task of DCE is to transfer the data bit stream to and from DTE and to adapt it to the conditions of the network. This is done, for each octet, by adding (subtracting) two bits to (from) the envelope of 8+2 bits. The two extra bits are used for envelope synchronization and signalling, see fact panel no. 1.

DCE is connected to a data multiplexor, DMX/RMX, or a data concentrator, DCC, via a physical or carrier circuit.

Data multiplexor, DMX/RMX

About 20 subscribers are normally connected to the data multiplexor, which without concentration combines the data flows from the subscriber lines to one 64 kbit/s data stream in accordance with CCITT Recommendation X.51, see fact panel no. 2.

The data multiplexor is connected to the data exchange, DSE, either direct or via a data concentrator, DCC. In the first case the multiplexor is designated DMX, in the latter case RMX. A multiplex circuit is used in both cases.

Data concentrator, DCC

The main task of the data concentrator is to concentrate and multiplex data channels from the subscriber side to a number of 64 kbit/s multiplex circuits towards the data exchange, DSE.

Fig. 4 shows a block diagram of the data concentrator. The subscriber lines are connected via modems to a line module which is common for up to 50 subscribers. The line modules communicate via a data bus with multiplex units, which each handle one 64 kbit/s multiplex circuit towards the data exchange, DSE.

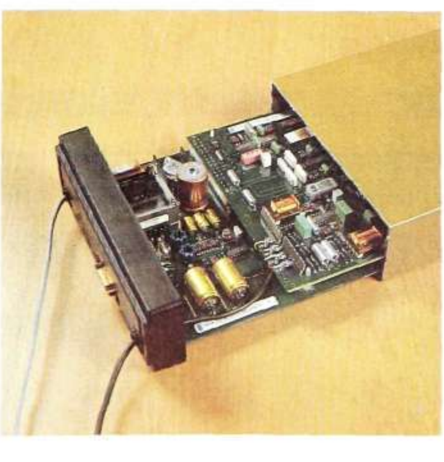
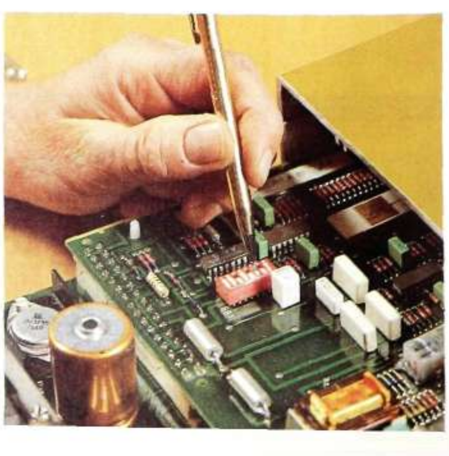


Fig. 3
Data circuit terminating unit, DCE. The transmission speed is easily adjusted when a new subscriber is connected



Fact Panel no. 2

CCITT Recommendations

A number of recommendations have been prepared within CCITT in order to standardize interface etc. for data communication. The CCITT recommendations that are most important to AXB 30 are listed here:

X.1 specifies the speed categories for subscriber terminals. AXB 30 provides for all asynchronous terminal speeds and all synchronous speeds up to and including 9 600 bit/s.

X.2 concerns subscriber facilities. AXB 30 follows the recommendations for all facilities in circuit switched networks with a few exceptions.

X.20 bis, X.21 and X.21 bis constitute recommendations for interfaces between terminals and networks, for asynchronous terminals with V interface (X.20 bis), for synchronous terminals with X interface (X.21) and for synchronous terminals with V interface (X.21 bis).

X.51 gives the frame structure on international multiplex circuits and is applied in AXB 30 between data network exchanges, and between a data network exchange and concentrators and multiplexors respectively.

X.54 gives supplementary information to X.51. The recommendation defines 56 different channel distribution alternatives for the speed categories of the subscriber channels concerned.

X.71 defines the channel associated traffic signalling between exchanges.

X.96 defines approximately 20 call progress signals. In AXB 30 these are transmitted to the DSE subscriber in the form of a two-digit code and contain information of the type: Called subscriber is engaged, please wait; Called subscriber is out of operation; Congestion in the network; etc.

X.121 gives the new international number plan for data networks.

DCC is remotely controlled by the data network exchange, DSE, over two control channels which are included in the two first multiplex circuits towards DSE. The control channels normally work in the load sharing mode, but if a fault occurs one control channel can handle all the control information.

Up to 500 subscribers can be connected to DCC, either direct or via RMX. The number of multiplex circuits to DSE is dependent on the traffic volume and varies from two to a maximum of ten.

DCC carries out the following traffic functions:

- identifying calls from connected subscribers and calling the data exchange, DSE, over the control channel
- on order from the data exchange, DSE, connecting a subscriber line to a specified channel in a multiplex circuit to DSE
- disconnecting the data circuits on command from DSE.

DCC also contains built-in functions for supervising that the equipment works satisfactorily. If a fault occurs the alarm information is transmitted to DSE via one of control channels.

The alarms are coordinated by DSE, which can, for example, order DCC to carry out tests for fault localization.

In order to obtain good operational reliability the logic units and control channels of the data concentrator are duplicated. The subscriber connections are grouped in modules of 50 subscribers, with one subscriber module per multiplex channel. The operation and maintenance module is not duplicated.

Data network exchange, DSE

The data exchange constitutes the central part of the data network. It controls and supervises the setting up and disconnection of data circuits between subscribers in the network and towards subscribers in other data networks.

DSE has a maximum connection capacity of 508 multiplex circuits of 64 kbit/s each for the connection of DCCs and DMXs/RMXs in the network and circuits to other networks. The call capacity of DSE is 70–100 calls per second.

The subscriber channels are always connected through DCC and DMX/RMX. The switching network is designed so that it is free from congestion.

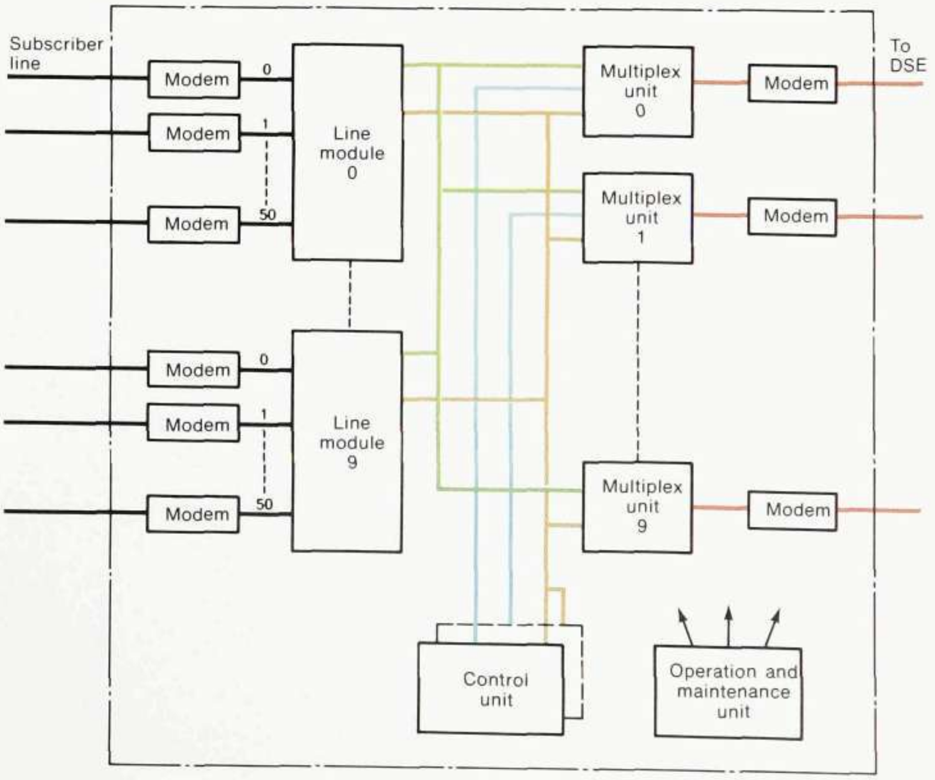


Fig. 4
Block diagram of the data concentrator, DCC

Data bus
Bus for control channel signalling
Control bus
Multiplex circuit, 64 kbit/s

DSE carries out the following traffic functions:
- controlling the setting up and disconnection procedure for subscriber channels in DCC
- supervising the subscriber states
- handling subscriber facilities
- charging
- controlling the collection of traffic data.

In the following description of the data network exchange the control system APZ 210 and and the switching system APT 611 are treated separately.

Control system

Control system APZ 210 is specially developed for the control of telecommunication systems with exacting requirements as regards real-time working. APZ 210 is used in all systems in the AX family. The lower part of fig. 5 shows a block diagram of the control system. The main features of the operation of the control system are that
- the data processing is carried out on two levels with regional processors, RP, which carry out time-demanding (real time) simple tasks, and a central processor, CP, which handles the more complex functions
- the regional processors are placed in the controlled equipment and communicate with the central processor via a bus system. Up to 512 regional processors can interwork with one central processor
- the central processor system consists of two processors that work in synchronism, one actively and the other passively. The two processors handle the same information and the results are compared at the micro

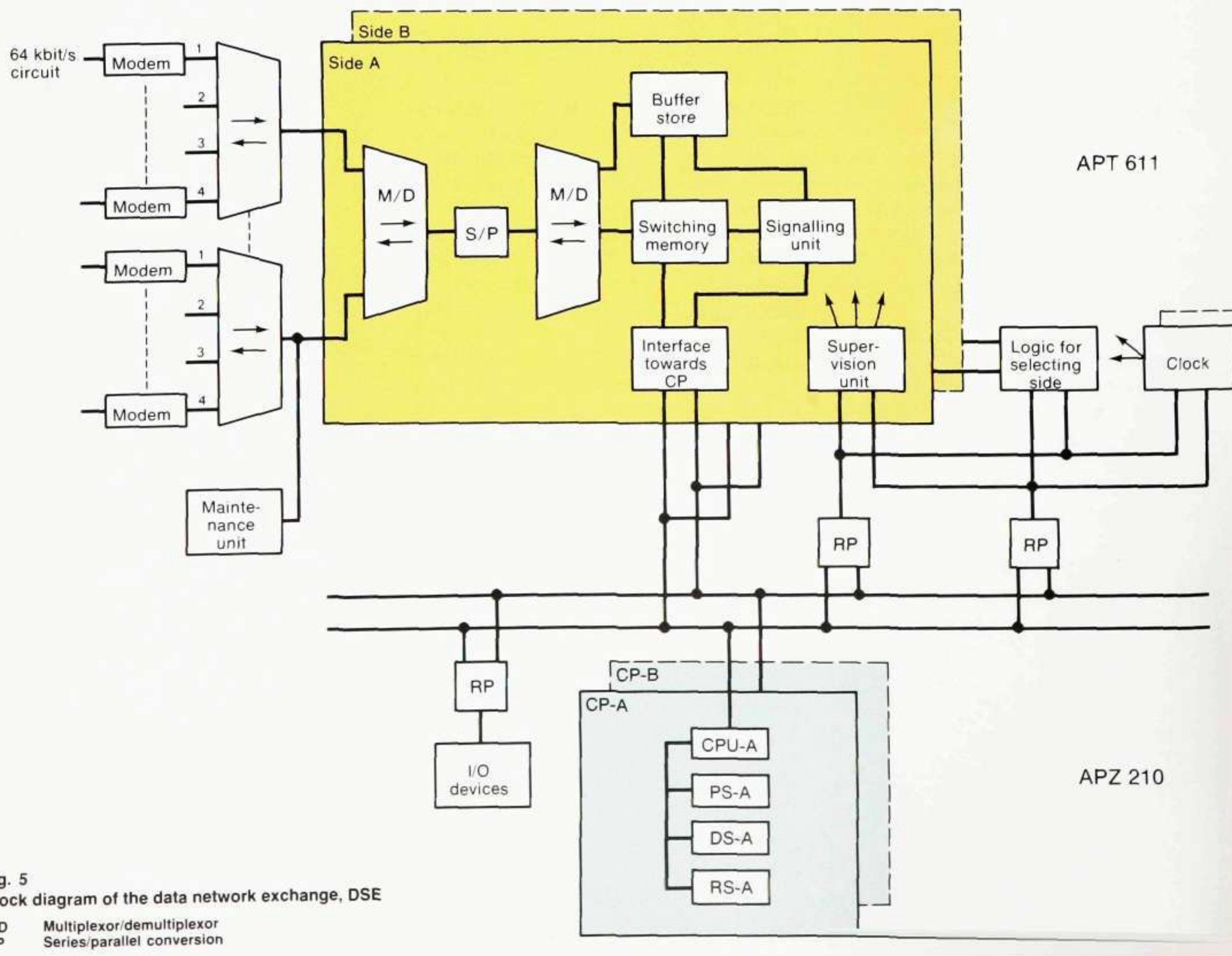


Fig. 5
Block diagram of the data network exchange, DSE

M/D Multiplexor/demultiplexor
S/P Series/parallel conversion

instruction level. If a fault occurs in the equipment a difference will be detected, and the processor system will interrupt the execution of the traffic programs and initiate a fault diagnosis, which starts with identification of the faulty processor side. The faulty processor is then immediately isolated, so that the fault can no longer interfere with the traffic handling. The execution of the traffic handling programs is resumed by the fault-free processor so quickly that the interruption does not affect the traffic

- software reliability is achieved by means of a structuring of the software combined with a similar arangment of the hardware in the central processor. Each software block in the program store, PS, can only read its own data among those stored in the data store, DS. Interworking with other software blocks is carried out by means of formal software signals. A reference store, RS, is used to indicate where in the store the programs and data for the different blocks are to be found. A plausibility check is carried out on the data in the signals in order to avoid, for example, incorrect addressing
- the reference store also simplifies reallocation of programs and data on site, i.e. in the data exchange, which facilitates changes and extensions.

Control system APZ 210 has been described previously in Ericsson Review[2].

Fig. 6
Data traffic generators are among the aids used during the installation testing of the data network system

Switching system

Switching system APT 611 consists of a switching network, to which multplex circuits from DCC and DMX/RMX are connected via modems. The switching network is controlled by application programs with the associated data, stored in the program and data stores in the central processor.

The top part of fig. 5 shows a block diagram of the switching network. The multiplex circuits are connected via modems to a multiplexing unit, which controls and supervises the frame synchronization towards DCC and DMX/RMX. Up to 508 multiplex circuits are connected through three successive multiplexing stages and series-parallel conversion to the switching memory.

The switching memory is organized with one word per channel and also contains logic circuits for analyzing incoming 8+2 envelopes so that status changes can be detected. The switching memory also stores the subscriber state for each channel (free, being connected/disconnected etc.).

When the subscriber initiates or terminates a call, the switching memory calls in the control system. If a call is to be set up the control system will respond with the address of the signalling unit with which the DCE is to interwork during the signalling.

The signalling units work autonomously and thus relieve the control system of some work during the setting-up stage. They have to store and analyze signalling information for further processing in the control system. Each signalling unit contains a microprocessor which can be loaded with different programs for different types of signalling (e.g. X.21, X.71, DCC signalling).

During the setting-up process the subscriber's data word in the switching memory is supplemented by the address to the outgoing subscriber channel. The address is used in the switching memory to address the buffer store. The buffer store is read cyclically, after which the data flow is demultiplexed in a similar way to the multiplexing described previously.

Fig. 7
Magnetic tape units for charging and statistics

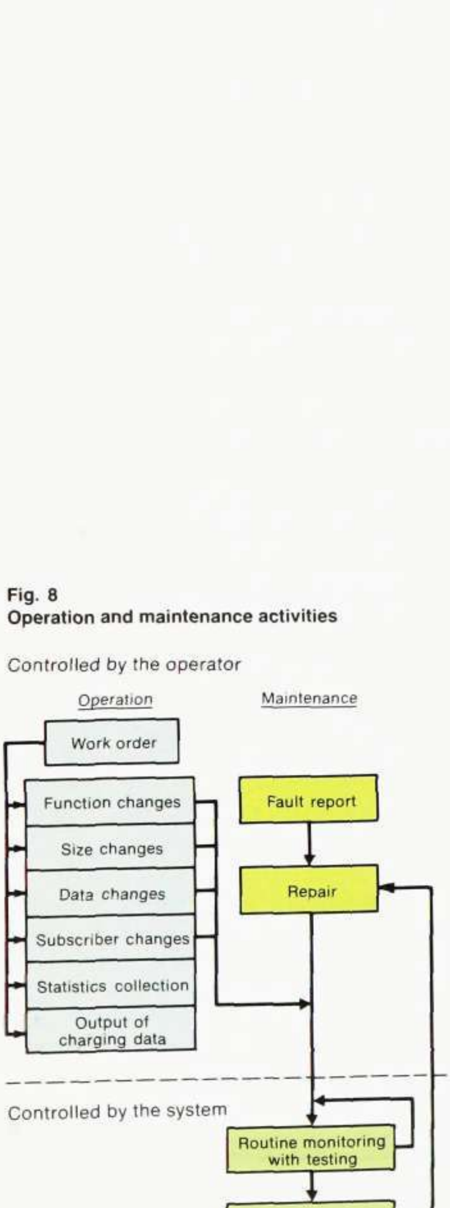


Fig. 8
Operation and maintenance activities

The switching system is synchronized by a clock which distributes timing to all units and, via the connected multiplex circuits, to the regional parts of the data network.

The switching network is monitored by means of a number of alarm points, which are scanned by the supervision unit. The alarm information is transmitted to the control system for evaluation and initiation of any alarms.

All traffic-handling equipment in the switching network is duplicated to form two sides, which work in a parallel synchronous mode with one side executive and the other on standby. For each transmitted 8+2 envelope a comparison is made between the two sides, and if a fault occurs, the faulty side is pinpointed and, if necessary, blocked. The changeover between the two sides, so that the standby side becomes the executive, takes place without any disturbance to established calls or the calls that are being set up or disconnected.

Timing and envelope synchronization

The timing in an AXB 30 system is wholly hierarchic, i.e. the exchange provides the timing for the concentrators and multiplexors, via the multiplex circuits, and they in their turn control the connected DCEs.

The synchronization is monitored on established calls. if synchronization is lost, the connection remains for half a second awaiting return of the synchronization, but is otherwise disconnected. Resynchronization can be accomplished after loss of synchronization on subscriber lines and multiplex circuits, regardless of what data and what status information were being transmitted on the subscriber channels.

AXB 30 has a synchronization system that permits timing regulation in accordance with optional network configurations. The clock in an AXB 30 exchange is duplicated and can adjust its frequency after comparison with external reference. The reference is either a separate clock, e.g. an atomic clock, or timing from a number of other exchanges in the network. Reference from other exchanges is received over the multiplex circuits used for the data transmission.

Traffic functions

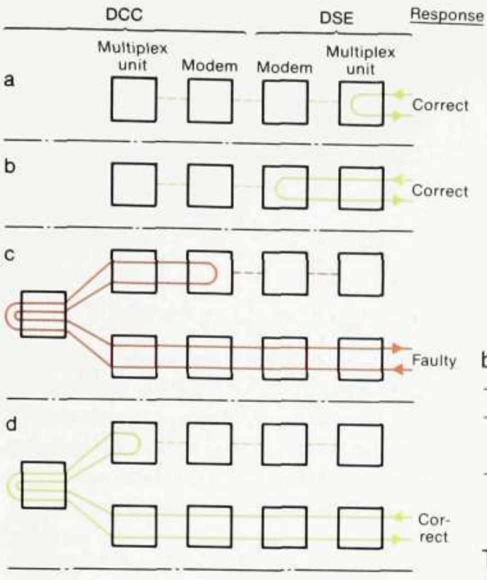
The traffic handling consists of setting up, supervising and disconnecting calls. In AXB 30 the setting up and disconnection is controlled by the central processor in the control system in DSE and executed by the hardware in DCC and DSE. Established calls are supervised by the hardware without any assistance from the control system. There is no setting up or disconnection in DMX, since all subscriber channels are permanently connected to their own timeslots on the multiplex circuits.

Operation and maintenance functions

The operation and maintenance in AXB 30 comprises activities for making changes in the network, for supervising its functions and for clearing any faults. There are autonomous activities, which are performed by the system without intervention from the operator, as well as manual actions, carried out by the operator.

The system differs from corresponding systems for telephony and telex in that AXB 30 does not consist of one exchange but a whole network. This network covers a large area, which means greater demands for advanced operation and maintenance functions, to be able to manage the network rationally. One basic principle in AXB 30 is therefore that the operator must be able to handle both fault clearing and operational functions centrally from DSE.

AXB 30 can be connected to an operation and maintenance centre from which the whole national data communication network can be supervised. It is also possible to route alarm printouts to the geographical area concerned, and to limit the authority of the operators to only the area in question. The parts of the network that fall within the administrative authority of a certain area can therefore be managed regionally via local operating terminals. Such terminals are connected to the control system in DSE via the data circuits in the network. The data network can thus



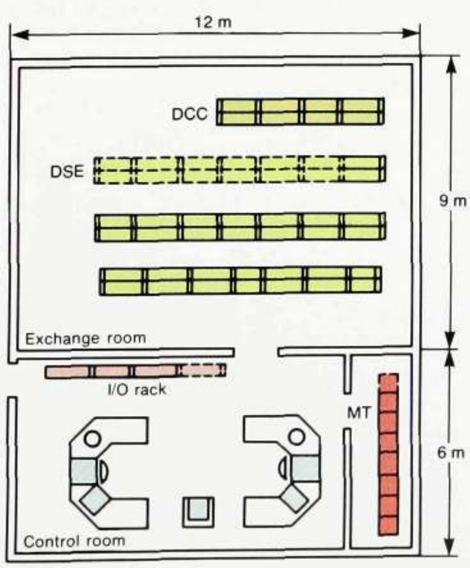
e

ALARM A2/APT
LINK LOSS OF SYNC.

MUX	FAULT	UNIT	DIRCTN
105	SYNC	MCMF	OUT

Fig. 9
Fault localization on multiplex circuits
DSE has detected loss of frame synchronization on a multiplex circuit to a concentrator. The autonomous maintenance system in DSE starts the fault localization by closing loops at the near end (stages a and b) and checking that the received data are in synchronism. If the near-end test does not result in a fault indication, a subscriber channel is connected up to a permanent loop in the DCC line module. This is done by means of two separate connection commands to the concentrator. The next step is to close the loops at the far end of the faulty circuit (stages c and d) and at the same time send a special bit pattern over the connected channel. The test results are evaluated and the faulty unit is pinpointed. Finally the system gives an alarm printout (e above) to the operator

Fig. 10
Floor plan for a data network exchange with 128 multiplex circuits for 64 kbit/s which is combined with a data concentrator for 500 subscribers. The control room contains two operators' positions with the associated I/O devices, which can be placed on a table or mounted in a rack.
The magnetic tape units, MT, are placed apart in a controlled environment



be managed in three different ways:
- from DSE
- from an operation and maintenance centre
- regionally via a local operating terminal.

This makes it possible to adapt the network management to the varying requirements of different administrations.

Fig. 8 shows the activities included in the operation and maintenance of AXB 30. The arrows indicate how the different activities follow each other. During normal traffic handling the system monitors the operation continuously ("routine monitoring with testing" in fig. 8). If a fault is detected, a sequence of maintenance activities is initiated which eventually leads back to the normal state with routine monitoring. The activities that concern changes in the network lead to the normal state after the changes have been tested.

Fig. 8 shows that all operational activities in AXB 30 are controlled by the operator. The maintenance activities, on the other hand, are mainly controlled by the system. The manual maintenance activities are always initiated by an alarm printout, which helps the operator to pinpoint the faulty unit immediately.

This clear division of the activities makes the system easy to handle. The great demands made on the operation and maintenance functions have led to approximately 70 % of the software in system AXB 30 consisting of programs for these functions.

Maintenance procedure

The sequence of maintenance activities in fig. 8 is illustrated by the following example.

In the normal state, i.e. routine monitoring, loss of synchronization occurs in the outgoing direction on a multiplex circuit between DSE and DCC. The fault is detected immediately by the monitoring function in DCC and is reported to DSE.

The maintenance programs in DSE check whether the synchronization fault on the multiplex circuit persists. The fault is then automatically located by the software in DSE by means of step-wise activating of a number of loop tests towards the faulty circuit. Fig. 9a-d illustrates these loop tests. In this example the fault was traced to the far-end modem (the DCC side) and isolated by DSE blocking the faulty multiplex circuit.

The system then provides an alarm printout for the operator, fig. 9e. The printout states unambiguosly the faulty multiplex circuit, "105", the type of fault, "SYNC", the unit, "MCMF", and the direction, "OUT".

When the fault has been cleared the system tests the multiplex circuit and then returns to the normal state, routine monitoring.

Charging

AXB 30 permits two different charging methods: toll ticketing and pulse metering.

Each subscriber can choose which one of these methods he prefers. Toll ticketing means that all data concerning a call are fed out when the call is disconnected. These data are afterwards processed in an external computer, and very detailed subscriber invoices can therefore be issued.

Pulse metering means that the subscriber invoice only gives the number of meter pulses. However, the pulse metering method is extremely flexible, so that it is possible to charge for different services by means of different numbers of pulses.

Structure

AXB 30 is built up in the same way as the other AX systems, with all the advantages these systems have proved to have, such as
- the system structure, with modular units and the function block as the basic unit for both software and hardware, which facilitiates function changes and extension
- the high-level language PLEX, for programming the computers, which, together with the various programming aids, ensures clear and lucid

Fact Panel no. 3

System data for AXB 30

System limits

DSE:	65 000 subscribers 508 multiplex circuits for 64 kbit/s 70–100 calls/second 24 mbit/s through-going flow
DCC:	500 subscribers 10 multiplex circuits 10 calls/second
DMX:	Typically 20 subscribers (max. 60) 1 multiplex circuit

Characteristics

Data speeds:	asynchronously 50–2 400 bit/s synchronously 600, 2 400, 4 800 and 9 600 bit/s
Envelope:	8+2
Setting-up time:	100–300 ms

Environmental requirements

Space requirements:	DSE (128 multiplex connections) 80 m^2 DCC (500 subscribers) 10 m^2
Power requirements:	DSE 17 000 W (128 multiplex connections) DCC 2 200 W (500 subscribers)

programs as well as high software quality

- the construction practice, which provides the best possible matching between the hardware function and the construction module (printed board assembly or magazine).

Space requirements

The equipment for DMX/RMX, DCC and DSE is mounted in standardized rack sections. Such a section is 2320 mm high for DMX/RMX and DCC, and 2970 mm for DSE. The section is 975 mm wide and 640 mm deep with the equipment placed back to back.

The number of single rack sections required are:

– DMX/RMX for 32 subscribers	1
– DCC for 500 subscribers	8
– DSE for 128 multiplex circuits	33

Fig. 10 shows the floor plan for an AXB 30 exchange with central equipment, DSE, for approximately 5000 subscribers, a data concentrator, DCC, for 500 subscribers and a control room.

Matching to other types of networks and systems

Standardized interfaces and protocols are required in order to obtain a universally usable communication system. AXB 30 meets the CCITT recommendations for data communication over switched circuits, fact panel no. 2. CCITT has standardized different types of networks. The two main groups for data communication are circuit switched and packet switched networks. The circuit switched networks category contains two variants, with the envelopes 8+2 and 6+2 bits respectively. AXB 30 meets the CCITT recommendations for circuit switched networks and the envelope 8+2, fact panel no. 1.

System AXB 30 contains all functions that are required for an all-embracing national and international data network. Functions are available for local exchanges, transit exchanges and international exchanges.

AXB 30 can also be connected to systems which use the control information in the X.51 frame in a different way.

Matching to other networks with different structures will be a common requirement in future. Some examples of the flexibility of AXB 30 as regards adaption to new facilities and networks are given below.

Traffic to a network with the 6+2 envelope and a multiplex structure in accordance with X.50 will only require a simple conversion in AXB 30 between the different frame structures. Data or signalling conversion between the two networks will not be necessary, since both networks are transparent to data and use signalling system X.71. The adaption is carried out by means of modified multiplexing units, which are installed in DSE.

In certain cases adaption will be required when introducing a new facility. One such facility is teletex, which has already been introduced in AXB 30.

Briefly, the teletex facility means that a modern text and word processing system is combined with communication facilities. The facility and the communication protocal are standardized in order to make general communication possible, regardless of the manufacture of the terminal.

In accordance with current CCITT recommendations teletex in AXB 30 uses the synchronous subscriber speed 2400 bit/s.

The only function which then had to be added to AXB 30 was a compatibility checking function for the teletex subscribers, which meant using a special category indication.

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140 Mbit/s Line System

Mats Eneborg, Örjan Mattsson and Giorgio Squartini

This article introduces Ericsson's digital line system for 140 Mbit/s, corresponding to 1920 telephone channels, on normal and small-core coaxial cable. The system has been developed in collaboration with the Italian company FATME, a member of the Ericsson Group. The line system meets all relevant CCITT and CEPT recommendations. It is particularly suitable for spare tube pairs in existing coaxial cables. The chosen repeater spacing is therefore the same as for the analog 12 MHz line system in order to simplify conversion from analog to digital transmission as well as to permit mixed operation of analog and digital systems on the same cable. The equipment is characterized by good transmission performance and easy installation, characteristics which have been thoroughly verified in field trials.

UDC 621.395.34:
621.315.2

The increase in the number of digital telephone exchanges means a growing need for digital line systems. In urban networks a transmision speedof2Mbit/s is usually sufficient at first, but the need for systems with a considerably higher capacity, for example 140 Mbit/s, will soon arise, particularly in cities. The digital islands which such cities form must be connected together via long-distance networks, which are today equipped with analog 4, 12 or 60 MHz systems. Since the coaxial cable constitutes the major cost in long-distance networks, it is an advantage to be able to use the installed cable with the intermediate repeater stations also for the digital systems. For this reason the digital line system ZAY 140-1, which transmits 140 Mbit/s (139.264 Mbit/s), corresponding to 1 920 speech channels, has been designed with the same repeater spacing as the analog 12 MHz line system. The nominal repeater spacing is thus 2 km on small-core (1.2/4.4 mm) coaxial cable and 4.65 km on normal (2.6/9.5 mm) coaxial cable. The distance between power feeding stations is 120 km and 280 km respectively. Using the same mechanical construction for the repeaters as that of Ericsson's analog 4, 12 and 60 MHz line systems facilitates conversion and mixed operation of analog and digital systems. The high cable attenuation caused by this repeater spacing means that the digital signal has to be recoded in the terminal in order to reduce the symbol rate and hence the signal attenuation. Otherwise the signal would be lost in the thermal noise in the repeaters.

The main requirements for the line system are:
- maximum cable attenuation 85 dB at 52.224 MHz
- bit error rate less than 10^{-10} per km
- low output jitter, less than 0.3 UI (Unit Interval) for a line system with 60 intermediate repeaters.

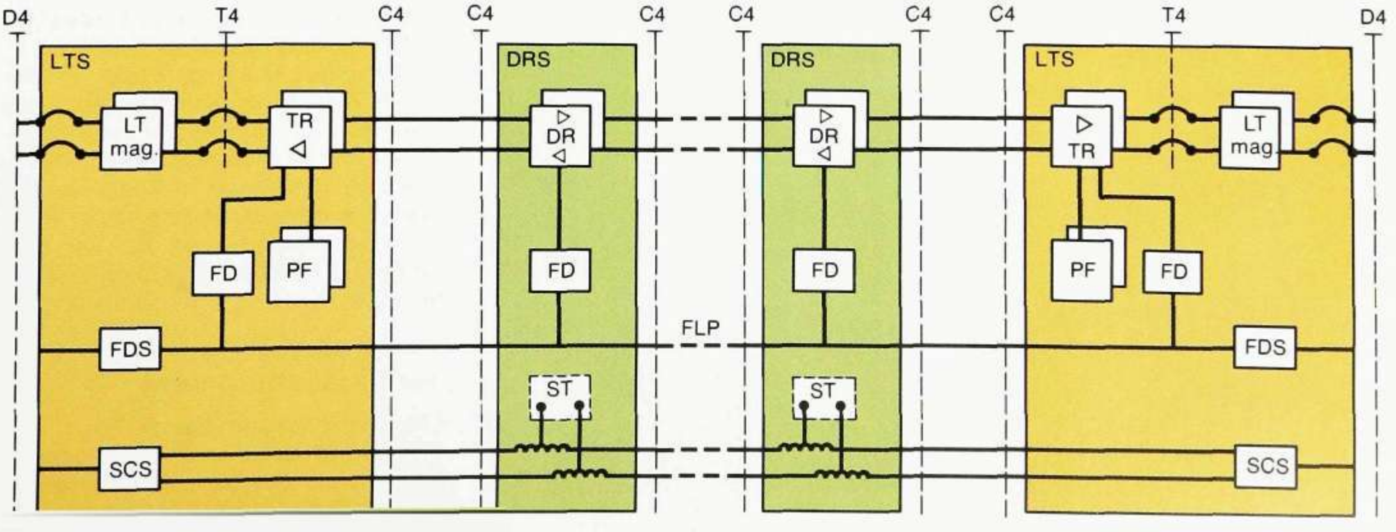
These requirements impose great demands on the design. Technically advanced but also robust designs have therefore been chosen, to ensure reliable operation throughout the life of the system.

System description

The interface that applies for the 140 Mbit/s line system is the one specified by CCITT in G.703, here designated D4, fig. 1. A 34/140 Mbit/s multiplexor, e.g. Ericsson's ZAK 34/140-1[1], or a digital radio relay link can be connected at this interface.

Fig. 1
The main units in the 140 Mbit/s line system

LTS	Line terminal station
LT mag.	Line terminating magazine
TR	Terminal repeater
FD	Fault detector unit
PF	Remote power feeding unit
FDS	Fault detector shelf
SCS	Service circuit shelf
FLP	Fault location pair
D4	140 Mbit/s interface, CCITT G.703
C4	Coaxial cable interface
T4	Ternary interface
DRS	Intermediate repeater station
DR	Intermediate repeater
ST	Service telephone



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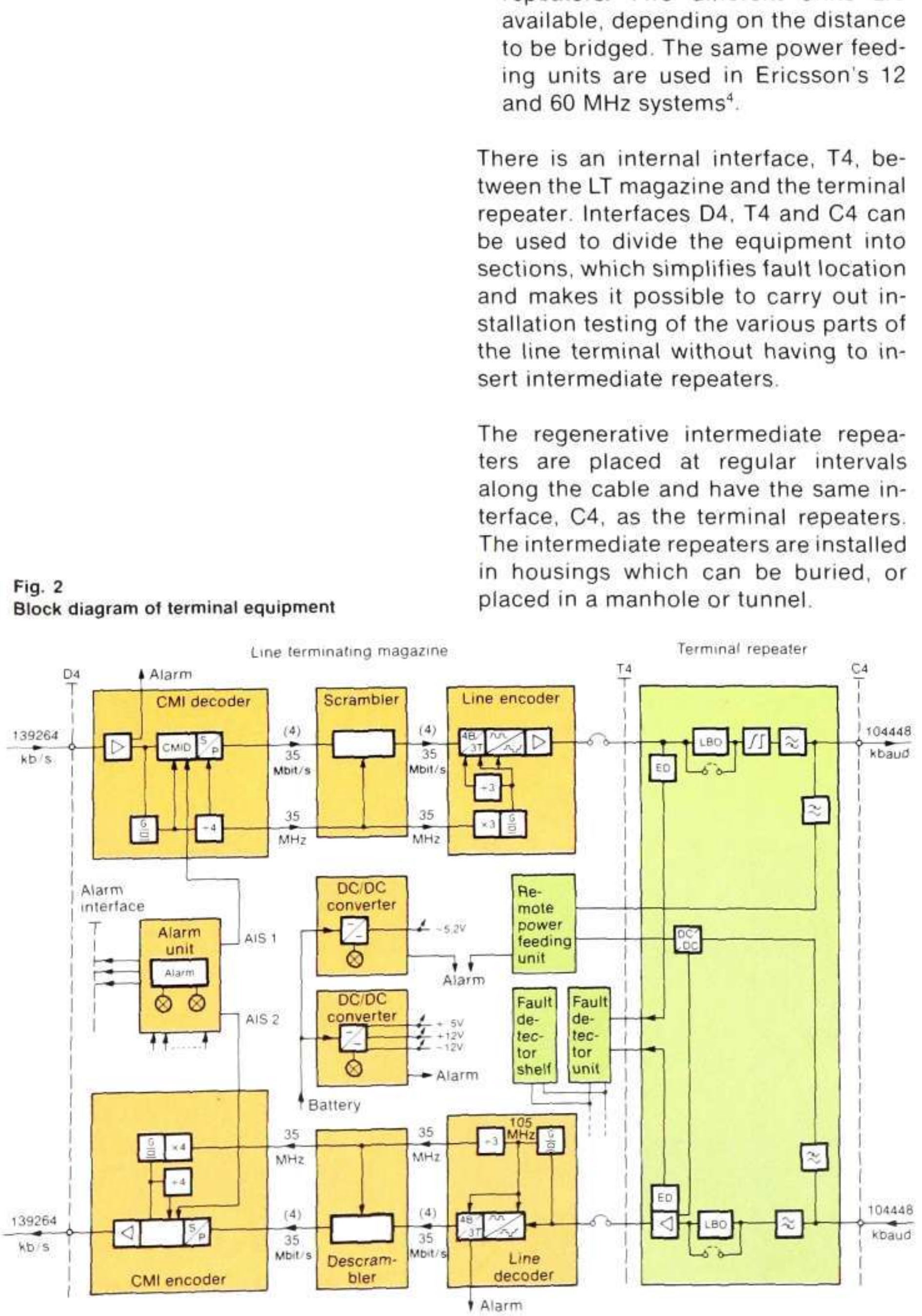
The terminal equipment, which is mounted in a M5/BYB rack[2], consists of three main units per line system:

- the line terminating magazine, LT mag., which recodes the signal, reduces the jitter, supervises the line system and initiates alarms if any faults occur
- the terminal repeater, which adapts the signal to the C4 cable interface
- the remote power feeding unit, which powers the terminal and intermediate repeaters. Two different units are available, depending on the distance to be bridged. The same power feeding units are used in Ericsson's 12 and 60 MHz systems[4].

There is an internal interface, T4, between the LT magazine and the terminal repeater. Interfaces D4, T4 and C4 can be used to divide the equipment into sections, which simplifies fault location and makes it possible to carry out installation testing of the various parts of the line terminal without having to insert intermediate repeaters.

The regenerative intermediate repeaters are placed at regular intervals along the cable and have the same interface, C4, as the terminal repeaters. The intermediate repeaters are installed in housings which can be buried, or placed in a manhole or tunnel.

Fig. 2
Block diagram of terminal equipment



A power feeding intermediate repeater station is obtained by installing two terminal repeaters back to back in a station together with the associated remote power feeding unit.

Faulty repeaters are traced by means of a fault location system which can be common for several 140 Mbit/s line systems. The fault location system consists of a fault detector shelf installed in the station from which the fault location is to be carried out, and fault detector units installed in the repeater housings. The fault detector shelf is the same as that used for Ericsson's 2 Mbit/s and 8 Mbit/s line systems[5], which simplifies the work of the station staff. In addition the terminal rack contains equipment for a 4-wire service telephone circuit, and connection points for this circuit are included in the repeater housings.

A variant of the terminal and fault location equipment in the N2 construction practice[6] has been developed by FATME in close collaboration with Ericsson.

Many parts of ZAY 140-1 are the same as those used in Ericsson's analog line systems, for example housings, the mechanical parts of the repeaters, the remote feeding units and the service telephone circuit equipment. This is extremely advantageous with regard to compatibility, training and the storing of spares. It also means that advantage has been takes of field experience gained from the use of these equipments.

The line system is easy to install and has well defined internal interfaces. The BYB magazines have front connectors and come fully equipped. The number of strappings required during installation is kept to a minimum, since the equipment is delivered already strapped for the most common applications. The system can also supervise its own operation without needing any external signals connected in, which is very useful during installation testing or when the system is used as a standby.

Terminal equipment

Line terminating magazine

The signal processing necessary to adapt the 140 Mbit/s interface signal

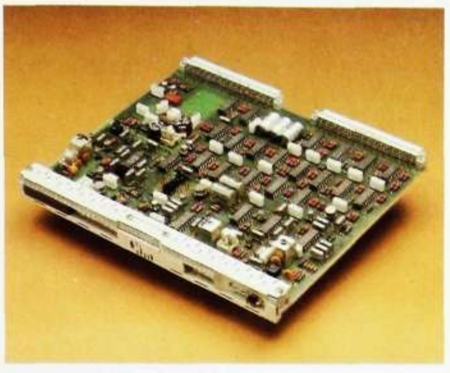


Fig. 4
The CMI decoder, which consists of a four-layer printed wiring board on which are mounted ECL circuits for the decoding, a controlled crystal oscillator for the timing recovery and 5 GHz transistors for the input stage

on coaxial cable is carried out in the line terminating magazine, fig. 2. Conversion between CMI code and the internal binary code is done in the CMI encoder and decoder. The send side scrambler makes the signal pattern more random, mainly, in order to simplify the timing recovery in the repeaters. The original signal pattern is restored in the receive side descrambler. These two units also contain special supervision circuits which prevent certain repetitive patterns from ocurring. The line encoder carries out the recoding of the signal that is necessary in order to reduce the symbol rate, and thus also the influence of the cable attenuation which the intermediate repeaters must bridge. The chosen code is of the 4B3T type, i.e. four incoming binary symbols are reduced to three outgoing ternary symbols. This gives a symbol rate on the cable of $^3/_4 \times 140 \sim$ 105 Mbaud. Since the number of binary states, $2^4 = 16$, is less than the number of ternary states, $3^3 = 27$, the code has a built-in redundancy. This spare capacity is used for fault location. It can also be used for data transmission. The magazine is therefore designed so that a data adapter for a 300 baud asynchronous data channel can be connected.

It would have been possible to use another code giving greater reduction of the symbol rate than 4B3T, but such codes would have rendered the signal more sensitive to variations in the transmission quality of the cable.

The line terminating magazine also reduces the jitter. Within the magazine the mode of operation is in four parallel 35 Mbit/s flows instead of one serial 140 Mbit/s flow. This means that the line and CMI encoders tolerate four times more jitter. Consequently the bandwidth of the timing recovery circuits can be made narrow in order to reduce the jitter on the signal before it is sent out on the cable or to the D4 interface. All timing recovery circuits in the magazine have been designed as phase-locked circuits with crystal oscillators in order to obtain good jitter characteristics.

The primary alarms are those recommended by CCITT. They are compiled by the alarm unit in the magazine. The compilation and transmission of alarms are carried out in the same way as in other transmission equipment in the BYB construction practice[2]. In the case of a loss of signal or high error rate an alarm indication signal (AIS) is inserted in the send or receive direction. In the 140 Mbit/s system an AIS is also required in the send direction, since recoding is carried out. The AIS in the send direction, which is made more random by the scrambler, can be used instead of an external pattern generator, for example during commissioning or when the system serves as a standby. The alarm "absence of signal in the send direction", which is obtained when the equipment is not connected to the D4 interface, can be inhibited by means of a switch on the front of the magazine.

Fig. 3
The line terminating magazine where recoding, jitter reduction and supervision are carried out

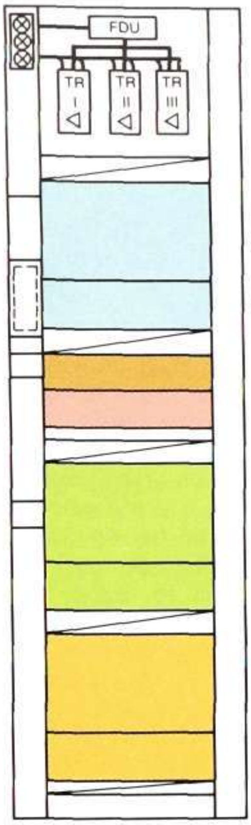


Fig. 6
A line terminal rack equipped with three systems

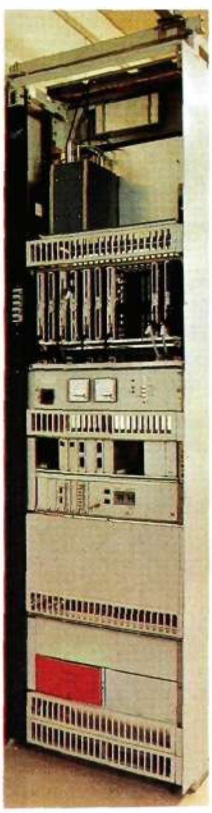


Fig. 7
A line terminal rack equipped with one system and a rectifier

The equipment is mounted in a BYB magazine, fig. 3, and is powered from the station battery via a d.c./d.c. converter in the magazine.

Since the bit rate within the magazine has been limited to 35 Mbit/s it is possible to use ordinary wire-wrapping for the connections. Four-layer printed wiring boards with, for example, impedance matched conductors enable different, fast circuit functions to be combined on one and the same board, fig. 4. All external connections are made on the front of the magazine, both the coaxial connections towards the signal interfaces and the alarm and test point connections.

Terminal repeater

The terminal repeater, which has the same mechanical construction as the intermediate repeater, adapts the signal received from the line terminating magazine to the cable and vice versa, fig. 2. The receive part consists of a regenerator of the same type as in the intermediate repeater. Like the intermediate repeater the terminal repeater can compensate an attenuation of 85 dB. Extra line extension networks can be strapped into the terminal repeaters so that attenuation values down to 0 dB can be handled.

Power feeding of the repeaters

The repeaters are powered either from a remote power feeding unit which provides 125 mA/1040 V and which is also used in Ericsson's 12 MHz system, or from one which provides 230 mA/1200 V and which corresponds to the type used in Ericsson's 60 MHz systems[4].

The maximum number of dependent intermediate repeaters is 21 and 31 respectively. Both units have a high impedance to earth, both for safety reasons and to limit induced hum. The units are automatically disconnected if a cable break occurs and if the output current becomes excessive.

No extra equipment is required in the repeaters where the power feeding circuit is looped, only a simple strapping.

Rack

The terminal equipment for three systems can be housed in one M5/BYB rack, fig. 6. If mains voltage is the only power source available, the rack is used for two systems and one rectifier, fig. 7.

The internal T4 interface allows high-voltage equipment, such as remote power feeding units and terminal repeaters, to be placed in one rack and the rest of the equipment in another if desired.

A service unit for the connection of station alarms and a service telephone is placed at the bottom of the rack.

Intermediate repeater

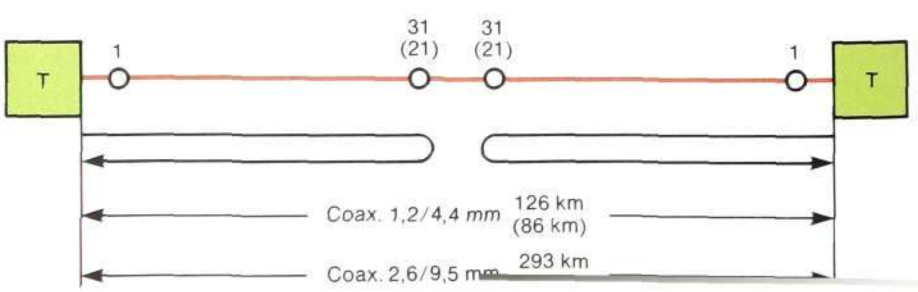
Design and operation

The intermediate repeater, fig. 8, contains functions for

- equalization
- timing recovery
- detection and pulse generation
- power feeding.

The repeater consists of two regenerators, one for each direction of transmission. The repeater spacing, which corresponds to a cable attenuation of 85 dB at 52 MHz, means that great demands are made on the transmision characteristics of the repeaters. In order to be able to guarantee the characteristics, and thus the function of the system, computer-aided designs were used to a great extent in the development work. Ericsson have developed

Fig. 5
The terminal and intermediate repeaters are power fed by a direct current fed through the centre conductor in the coaxial cable. The current is looped in the last dependent repeater by means of a simple strapping. The feeding current is 230 mA, alternatively 125 mA. The values for the latter alternative are given in brackets



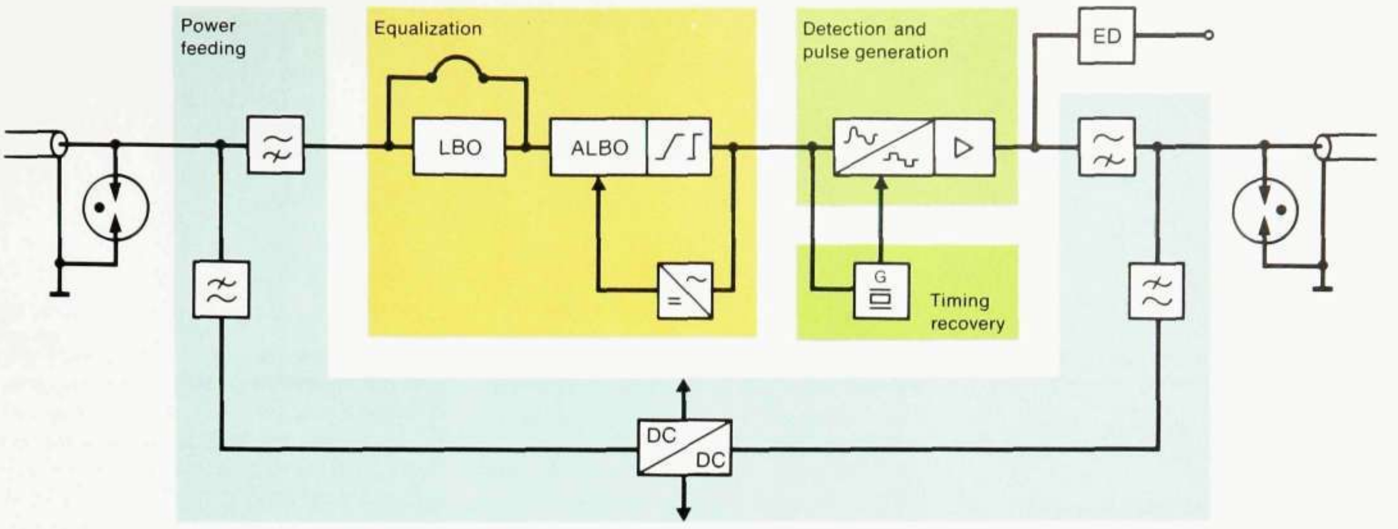


Fig. 8
Block diagram of one transmission direction in the intermediate repeater. The d.c./d.c. converter is common for both directions

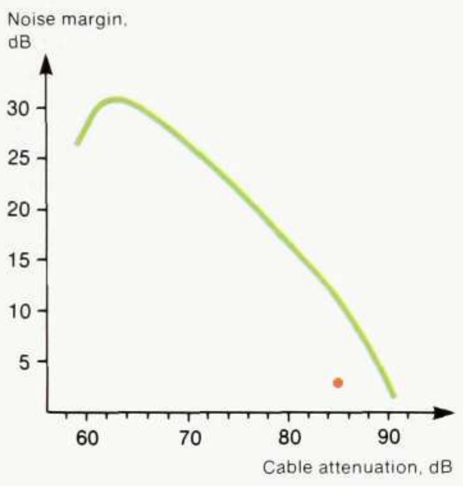
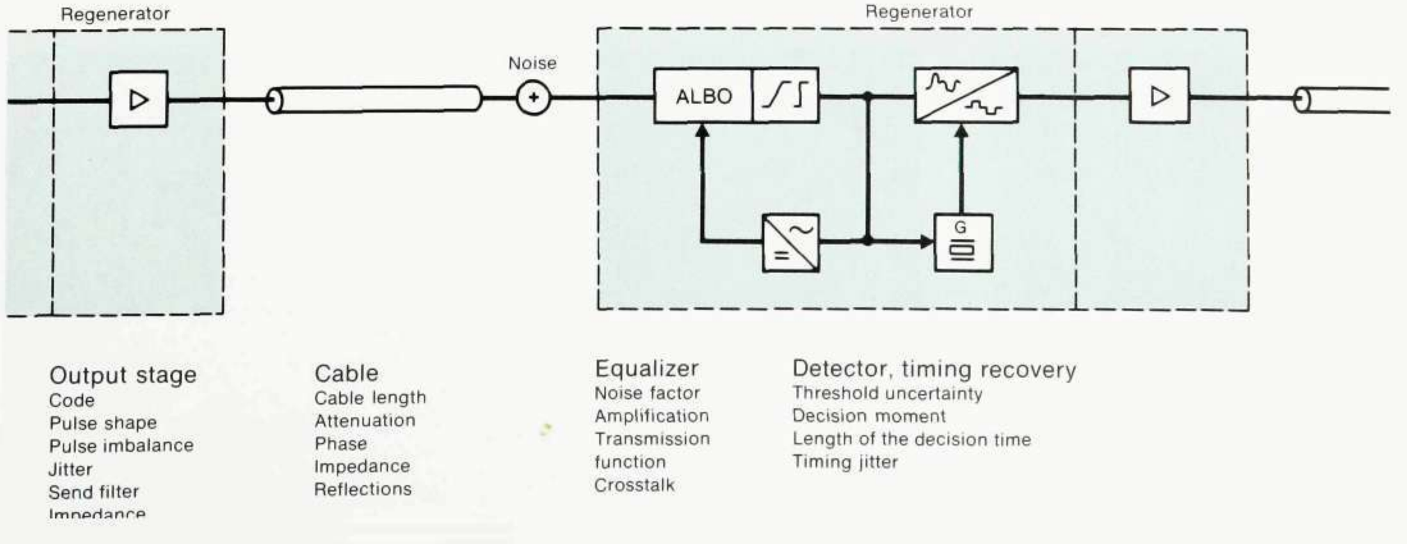


Fig. 10
Measured noise margin at an error rate of 10^{-8}, as a function of the cable attenuation

• Indicates the result of the worst possible case found during the simulation with maximum cable length (85dB), both as regards operating conditions and parameter dispersion

Fig. 9
Model of a repeater section used for computer calculation of performance, for example the noise margin of an intermediate repeater. The program calculates, on the basis of specified parameters, how much additional noise the repeater can handle before the specified error rate is exceeded



computer programs which simulate and optimize the system parameters that affect the performance of a repeater section, fig. 9. These programs can be combined with circuit implementation programs that simulate and optimize both the linear and non-linear circuits included in the design. These computer programs have been used to

- optimize the transmission function of the equalizer to obtain the best possible system performance with due consideration to the variation of cable characteristics, the tolerances of the included circuits etc.
- optimize and verify the circuit designs and ensure that the specified requirements are met in spite of the distribution of component data, trimming limits, temperature and ageing variations etc.
- simulate the transmission characteristics of the repeater under all imaginable operating conditions, in order to investigate the sensitivity of the system to variations of different parameters and also the function of the system during the most adverse operating conditions.

By these computer aids it has been possible to design an intermediate repeater that meets the requirements with good margin even in the worst possible operating conditions. This has been verified by measurements, fig. 10.

In order to obtain low thermal noise the repeater has been designed with a type of balanced feedback, which improves the noise characteristics. The equalizer, which compensates 85 dB cable attenuation, has an automatic adjusting range of 20 dB. For cable sections with an attenuation of less than 65 dB a fixed line extension network can be strapped in steps down to 25 dB. The active part of the equalizer consists of four identical amplifier hybrids in thick film technology. The amplifiers have local frequency-dependent feedback circuits. The final pulse shaping is carried out with the aid of passive networks. The chosen design gives a very stable construction. Fig. 11 shows the equalizer output signal in the form of an eye diagram.

The timing recovery circuit contains a phase-locked circuit with a crystal oscillator, which gives the circuit the necessary temperature and ageing characteristics. The use of a crystal permits a high Q-value, which together with other parameters give the whole line system very good jitter properties.

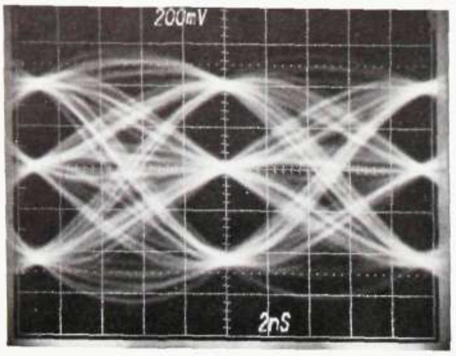


Fig. 11
Oscilloscope picture of the equalizer output signal in the form of an eye diagram. The cable attenuation is 85 dB at 52.224 MHz

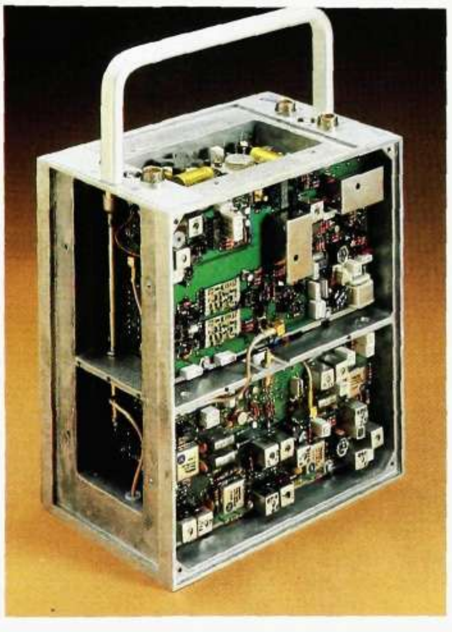


Fig. 12a
Intermediate repeater with all covers removed

Fig. 12b
The cover of the repeater holds the four coaxial line cable connectors and the two connectors for the fault location equipment. The top cover of the repeater also serves as a label where the chosen settings can be recorded

Fig. 13
Intermediate repeater housings for three and six systems respectively. (The volume of the six-system housing is approximately twice the volume of the three-system housing.)

The received ternary symbol is detected with the aid of the equalized signal and the extracted timing, and the original signal is regenerated in the subsequent output stage. The detector circuits must have a small threshold uncertainty interval and a short detection time, in the order of a few hundred picoseconds. This has been achieved through the use of thick-film technology and a circuit design which is symmetrical throughout.

In the 140 Mbit/s intermediate repeater a d.c./d.c. converter is used instead of a zener diode to provide the voltages required by the repeater. A d.c./d.c. converter which is common for both transmission directions has several advantages, such as

- hum suppression
- easy to restrap for different remote power feeding alternatives
- freedom of choice as regards secondary currents and voltages
- common signal earth and chassis earth, which gives more stable high frequency decoupling.

Both the input and output are equipped with lightning protectors in the form of rare gas overvoltage protectors, fig. 8. These are supplemented by a semiconductor protection. The repeaters have been tested with lightning voltages in excess of the CCITT requirements without any deterioration of the performance.

Reliability

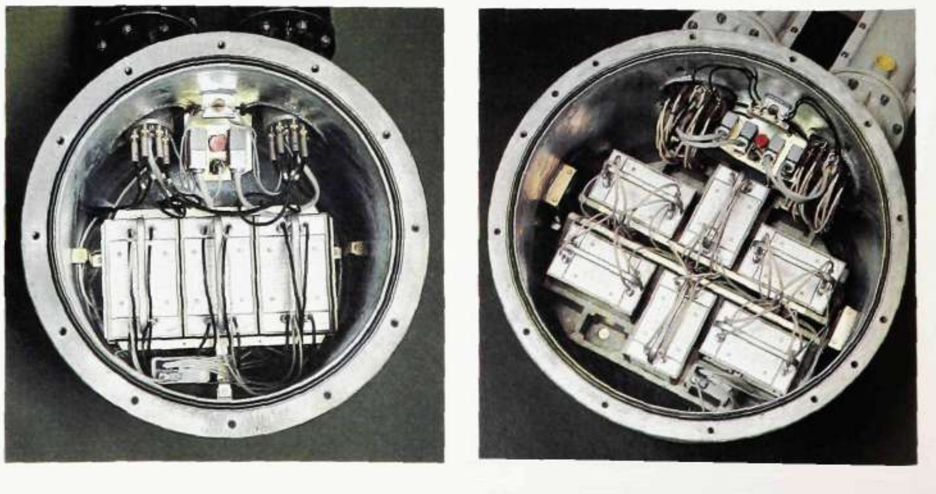
This type of line equipment with its large transmission capacity requires high reliability. The components have therefore been chosen exclusively from among those that meet the requirements for buried equipment. Certain of these components undergo special delivery tests, and their reliability characteristics have previously been proven in similar equipment. Certain components, such as hybrid circuits, also undergo burn-in.

Mechanical construction

Like the terminal repeaters, the intermediate repeaters are installed in the same type of case, fig. 12, as is used for analog repeaters. The die-cast aluminium box, which holds the equipment for both directions of transmission, measures 310×210×130 mm including the handle. The box is designed for easy handling. It meets the requirements for good screening between different function blocks as well as short electrical connections.

Housings for intermediate repeater stations

Cylindrical housings are available which can house the repeaters for three or six systems. The three-system housing, fig. 13, is the same as that used in Ericsson's analog systems[3,4]. The housings can be buried, since they are made of galvanized steel and coated with epoxy tar. No stub cable is required, and the main cable is jointed direct to the cable terminals of the housing. The container lids are sealed with ring seals. Pressurization can be arranged from the cable. Alternatively the cable pressurization can be through-connected to the following cable section. In the three-system housing the repeaters are mounted in a metal frame and in the six-



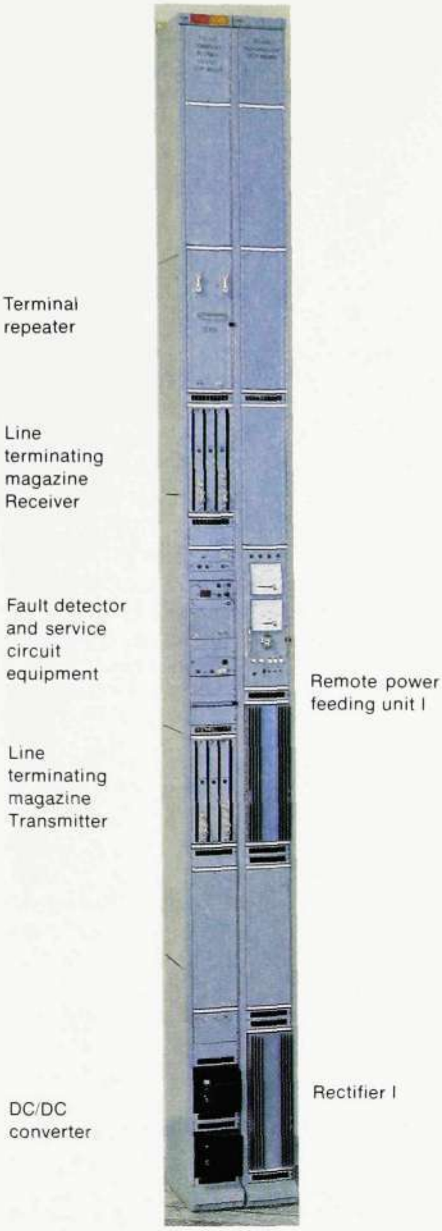
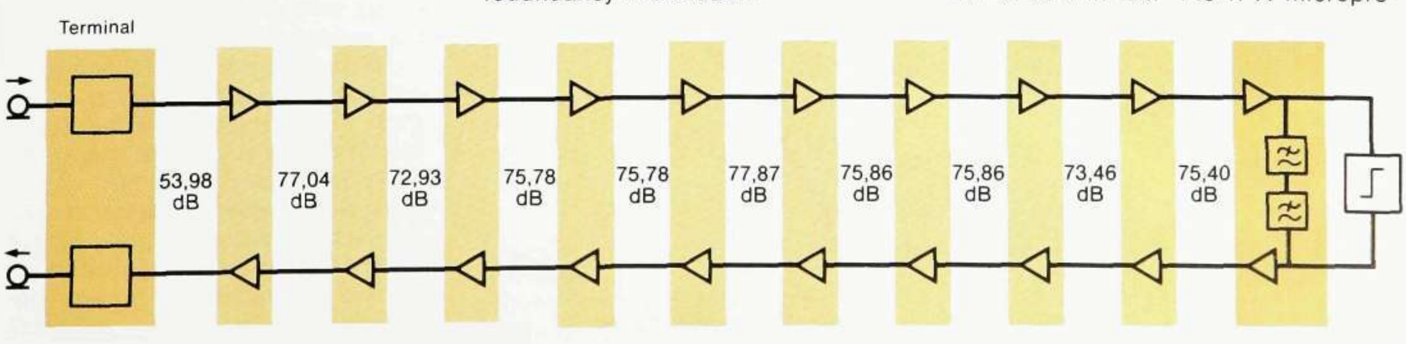


Fig. 14
Line terminal rack for a system in the N2 construction practice. Dimensions: height 2 600 mm, width 120 mm and depth 225 mm. The right-hand rack can be equipped with two remote power feeding units

Fig. 15
The trial route near Stockholm



system housing in a polystyrene insert. The repeaters are connected to the cable terminals via flexible cables.

The repeaters should be insulated from earth if there is large hum induction in the cable. In such cases all metal parts in the housing are, for safety reasons, automatically earthed when the lid is opened.

The housing also holds the fault detector unit and a connector for connecting a service telephone to the service circuit.

Fault location

Faulty intermediate repeaters are located with the aid of the same system as is used in the 2 Mbit/s and 8 Mbit/s line systems[5]. The terminal equipments are identical, which makes it possible to connect the transmission maintenance system ZAN 01.

The fault detector system allows detection of bit errors in the line signal during traffic by using the redundancy in the line code. Each intermediate repeater contains a fault detector circuit, which is connected to a common fault detector unit. The unit is connected to the fault detector shelf via a separate pair.

Information regarding the bit error rate on a certain intermediate repeater output can be transmitted to the fault detector shelf for analysis by means of a simple addressing procedure. The fault is located by addressing consecutive intermediate repeaters and comparing the bit error rates. The system can be tested by injecting code errors in the line terminating magazine. The error injection will not interfere with the traffic, since the errors are corrected in the terminal equipment by means of the redundancy in the code.

Locating a cable break

In the case of a cable break all power to the dependent repeaters is immediately cut off and the signal disappears, which makes it impossible to locate the break by means of the fault detector system.

In order to find the break the current from the remote power feeding unit is measured. The unit will be providing a constant output voltage of 600 V[4]. Each repeater contains a high-impedance resistor between the two directions of transmision, which gives a leakage current of a certain value. During the installation a calibration is carried out, which provides a reference list of the leakage readings for the individual repeaters.

Service telephone circuit

When ZAY 140-1 is used in the long-distance network a 4-wire service telephone circuit is often necessary because of the long distances. At the terminals the 4-wire service circuit equipment is connected to the ordinary 2-wire service telephone in the M5/BYB rack. The permitted attenuation on the loaded service circuit pair may be up to about 40 dB.

FATME terminal equipment

FATME has developed a version of line system ZAY 140-1 which has been designated ZAYF 140, and which meets the special Italian requirements. The main difference is that the narrow rack in the N2 construction practice has been used for the line terminal, which takes up two racks, fig. 14.

One of the racks holds the terminal repeaters (top) and the line terminating, fault detector and power feeding equipment. The fault location system is similar to that in ZAY 140-1. A micropro-

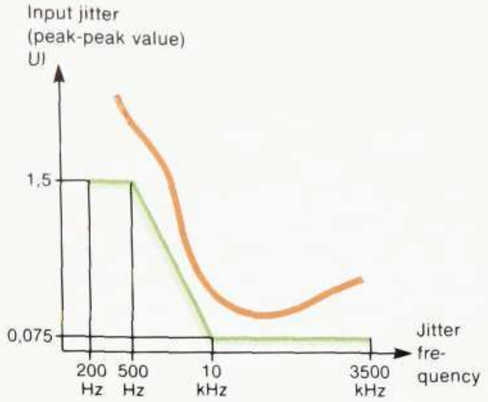


Fig. 16
Acceptable input jitter as a function of the frequency. The CCITT recommendation is included for comparison purposes

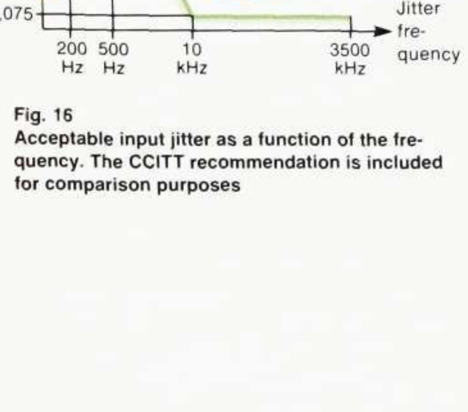


Fig. 17
The jitter transmission function for 21 regenerators illustrates the jitter reducing effect of the terminal
—— without terminal equipment
—— with terminal equipment

Fig. 18
The output jitter, measured (x) and calculated (•) values for up to 60 regenerators. The maximum permissible output jitter in accordance with the CCITT recommendations is 1.2 UI, the internal system requirement for 60 regenerators is 0.3 UI

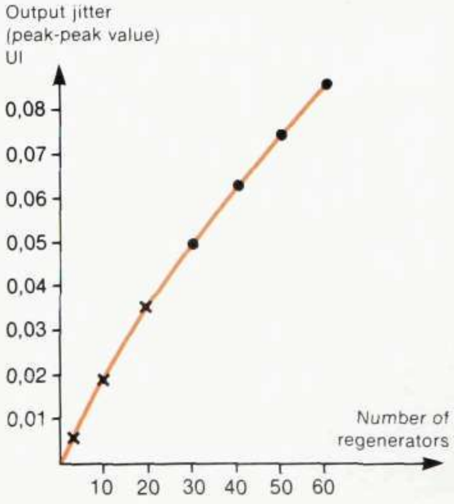
cessor is included in the terminal, and thus the equipment can automatically check the error rate of the terminal and intermediate repeaters in turn.

The other rack containstheremote power feeding equipment, 210 mA/1000 V, which can feed 30 dependent repeaters. The output voltage is reversed when a cable break is to be located.

Field trials

The system has been evaluated during field trials in Italy and in Sweden. The results meet the specified requirements with good margins. The equipment was installed and put into operation in a few days, which illustrates how easy the system is to handle.

The Italian administration ASST provided a pair in a normal diameter coaxial cable. The trial route contained 10 intermediate repeaters, and the signal was looped back in the last repeater. The terminal and fault location equipment developed by FATME was used. The system measurements, which started in March 1981, gave the initial confirmation of the good performance of the system. For example, the loss on one section was increased to 89.7 dB and the equipment on the route still worked perfectly. The results of the measurements have been presented to ASST and also to ISPT, the national administration for post and telecommunications. They have also carried out their own measurements on the route.



The Swedish field trial was carried out during the autumn of 1981 on a small-core coaxial cable near Stockholm, fig. 15. The measurements were made in collaboration with the Swedish Telecommunications Administration, and the equipment was taken from the ordinary production line. The purpose of the trial was to confirm the good results obtained in Italy and to study the system performance under adverse operating conditions. The most important results were:

- no bit errors were detected during a 64-hour test period, which corresponds to an error rate of better than 3×10^{-14}. During the whole test period the system was subjected to maximum input jitter in accordance with CCITT, hum current with a peak value of 100 mA and a timing deviation of 100 ppm
- no bit errors were detected when the attenuation on one section was increased to 89 dB, which is 4 dB above the specified value
- the ability to withstand jitter is clearly better than that recommended by CCITT, fig. 16
- no overall jitter amplification could be detected in the system, fig. 17. The jitter amplification per regenerator was only about 0.01 dB
- the output jitter, recalculated for 60 repeaters, was less than 0.1 UI, fig. 18. These good jitter characteristics were obtained mainly because crystal oscillators are used in the timing recovery circuits in the repeaters. This means that the system is an excellent long-distance system and no jitter reducers are necessary
- the system worked faultlessly with a timing deviation of ±210 ppm. This is another example of the large built-in margins, since it is calculated that in practice the deviation will never exceed a total of ±70 ppm
- the remote power feeding current could be reduced from 125 mA to 80 mA before bit errors started to appear. In the Italian trial the corresponding figures were 210 mA, which could be reduced to 160 mA
- the system worked faultlessly in spite of a hum current with a peak value of 300 mA which was superposed on the 125 mA remote feeding current. This

Technical data

	D4 interface	C4 interface
Transmission parameters		
Bit rate	139.264 Mbit/s	
Symbol rate		104.448 Mbaud
Code	CMI	MS43
Impedance, unbalanced	75 ohms	75 ohms
Pulse amplitude	1±0.1 V	±6 V

Transmission medium	Coaxial cable
Maximum cable attenuation	85 dB at 52.224 MHz
Nominal repeater spacing	2 km (1.2/4.4 mm coax.) 4.65 km (2.5/9.5 mm coax.)
Automatic regulating range	20 dB at 52.224 MHz
Bit error rate per repeater	$<10^{-10}$
Noise margin	5 dB with 85dB cable attenuation and a maximum bit error rate of 10^{-10}
Power supply	
Nominal battery voltage	36, 48 and 60 V
Mains voltage	110, 127 and 220 V ±10%
Mains frequency	45–60 Hz
Remote power feeding	
Current	230 mA or 125mA
Maximum voltage	1200 V or 1040 V
Maximum number of terminal and intermediate repeaters per remote power feeding unit	32 (230 mA) or 22 (125 mA)
Fault location	
Location of faulty intermediate repeaters	Over a separate pair
Maximum number of connected fault detector units	32
Cable fault location	Per power feeding section
Service circuit	2-wire or 4-wire
Temperature	
– line terminal	0 to +45°C
– intermediate repeater	−20 to +55°C
Mechanical data	
Rack	H×W×D
– for 1 or 2 systems	2134×600×260 mm
– for 2 or 3 systems	2743×600×260 mm
Intermediate repeater housing	Ø×H
– for 3 systems	628×760 mm
– for 6 systems	830×970 mm
Two-way intermediate repeater including handle	310×210×130 mm

provided by the d.c./d.c. converters in the repeaters.

The cable on the trial route also contained an analog 12 MHz system from Ericsson, so the opportunity was taken to carry out noise measurements on this system. No influence from the 140 Mbit/s system could be detected.

Summary

The digital line system ZAY 140-1 is of necessity technically complicated, particularly since it had to be compatible with existing analog 12 MHz systems. At the same time the system had to be robust, reliable and easy to use. For these reasons
- good margins have been built into the design
- careful analyses have been made, for example by means of computer simulation of the different function blocks
- experience and equipment units from analog and lower order digital line systems have been utilized
- the system has been designed with well defined internal interfaces between subsystems that are easy to handle.

The sucessful field trials in Italy and in Sweden have demonstrated that the equipment meets the requirements with good margins.

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CCITT Signalling System No. 7 in AXE 10

Jan Du Rietz and Hans Giertz

CCITT signalling system No. 7 is a system for common channel signalling. It has been developed to meet the growing need for the transfer of information in telecommunication networks and to obtain better utilization of the new, stored program controlled exchanges. Even with a moderate number of lines the amount of equipment required for common channel signalling is less than that needed for channel-associated signalling, so the introduction of signalling system No. 7 is profitable even in the short term. In the long run the system permits the introduction of new facilities in the telecommunications network. The authors describe signalling system No. 7, how it is introduced in AXE 10, and the possibilities it offers in combination with AXE 10.

UDC 621.391:
621.395.7

Signalling over junction lines between telephone exchanges is used to establish connections between calling and called subscribers. Signalling is used to set up each part of the connection and to transmit information forward through the network regarding the destination exchange and the number of the called subscriber. Previously signalling was tied to the same line as the speech, channel-associated signalling. The junction lines had individual signalling relay sets, whose performance was limited by technical and economic restrictions. This type of signalling was natural to the older telephone systems. A network providing stored program control (SPC) capabilities in the switching nodes offers advantages in reduced call set-up time, better economy, greater reliability, improved maintenance and new facilities. However, the full capabilities of the features inherent in the application of SPC techniques can only be achieved if sufficient information can be transferred between switching centres. The CCITT No. 7 Common Channel Signalling System (CCS) provides this information transfer capability.

CCS transmits address signals (the dialled digits) and supervisory signals (on-hook/off-hook) on data links (signalling links) that are separate from the voice network. A signalling link is common for a large number of channels (e.g. one or more speech routes).

Since the signalling path is separated from the speech path, the signalling system can be used for various purposes in addition to call handling. CCS is profitable even in the short term because common equipment is used for several speech channels.

CCITT Signalling System No. 7

CCITT has specified two different CCS systems, No. 6 and No. 7. Signalling system No. 6, which was designed primarily for international telephony, has previously been described in Ericsson Review[1]. Based on the experience obtained from signalling system No. 6, CCITT started to specify signalling system No. 7 in the 1970s. In addition to signalling between exchanges in international and national networks, signalling system No. 7 also permits signalling to operation and maintenance centers, PBXs, and remote subscriber stages. In the future it will enable very sophisticated subscriber facilities such as those proposed for the integrated services digital network, ISDN, to be properly implemented. Signalling system No. 7 is optimized for digital networks, but it

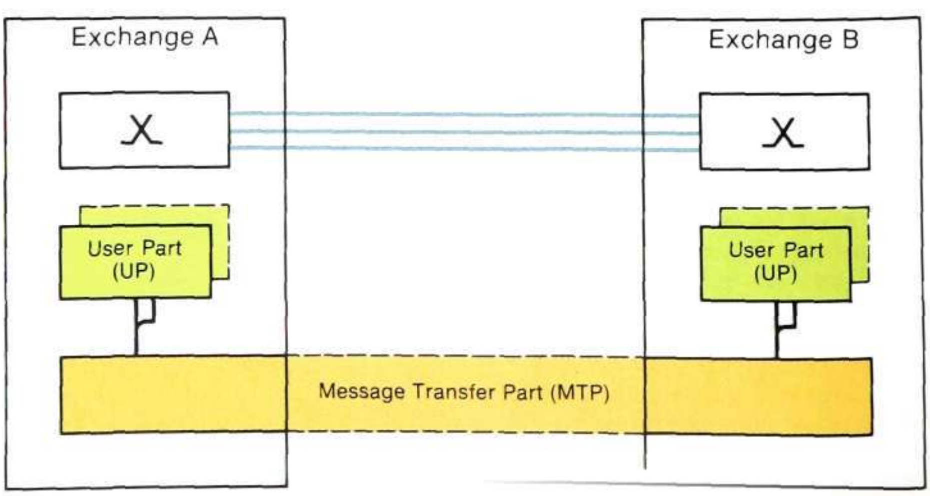

Fig. 1
The basic structure of signalling system No. 7. Signalling system No. 7 consists of a Message Transfer Part (MTP) and User Parts (UPs). MTP transmits Message Signal Units, MSUs, reliably (i.e. without errors) between UPs. UP processes the signal information contained in MSU

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can also be used for signalling over analog transmission circuits.

This wide range of applications places varying demands on the signalling system, and a functional structure is therefore a prerequisite. Functionally signalling system No. 7 consists of separate parts working independently of one another, namely User Parts, UPs, and a common Message Transfer Part, MTP. The task of the MTP is to transfer information reliably between UPs in two exchanges, fig. 1. The design of UPs varies with the application. Thus there is among others a Telephone User Part, TUP, a Data User Part, DUP and a Maintenance User Part, MUP.

The information is transferred over the signalling links in the form of Message Signal Units, MSU. An MSU contains the information to be transmitted, a message label for routing the MSU through the network, infomation about the receiving UP and fields for detection and correction of errors, fig. 2. The length of the MSU depends on the amount of information to be transmitted.

The UP contains data which has been compiled in a suitable format. The information to be transmitted in the MSU is assembled together with the message label and information about the receiving UP. The UP also processes the information in a received MSU.

The message transfer part content is divided into three levels, fig. 3. Level 1 encompasses the physical signalling data link, which in a digital network consists of a 64 kbit/s timeslot in a PCM system. Level 2 encompasses the signalling terminal together with functions for adaptation between the processor software signals and the bit stream of the signalling data link. Fields for error detection and correction are added by the signalling terminal in order to ensure error-free transmission of MSU. These fields are analysed in the receiving signalling terminal, and repetition is requested if an error is detected. This method ensures that

- MSU is transmitted without errors
- different MSUs are transmitted in the correct sequence and only once.

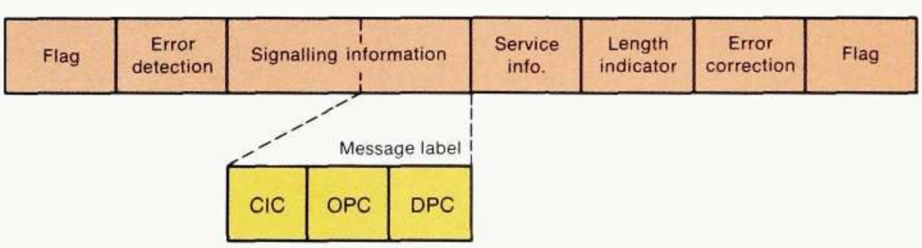


Fig. 2
Message Signal Unit (MSU)
An MSU is the data packet which is transmitted over the signalling link. An MSU consists of the following fields:

Flag	**The beginning and the end of an MSU are identified by a unique 8-bit word**
Error detection	**A 16-bit checksum is used for detecting errors**
Signalling information	**This field contains the information to be transmitted and a message label. The length of the information field is variable. The message label is used to route the MSU through the network and consists of an originating point code, OPC, a destination point code, DPC, and a circuit identification code, CIC**
Service information	**Specifies to which UP and which hierarchic level in the network the message belongs**
Lenght indicator	**Specifies the length of the MSU**
Error correction	**Ensures that MSUs are is received in the correct sequence, and orders repetition of any incorrectly received MSUs**

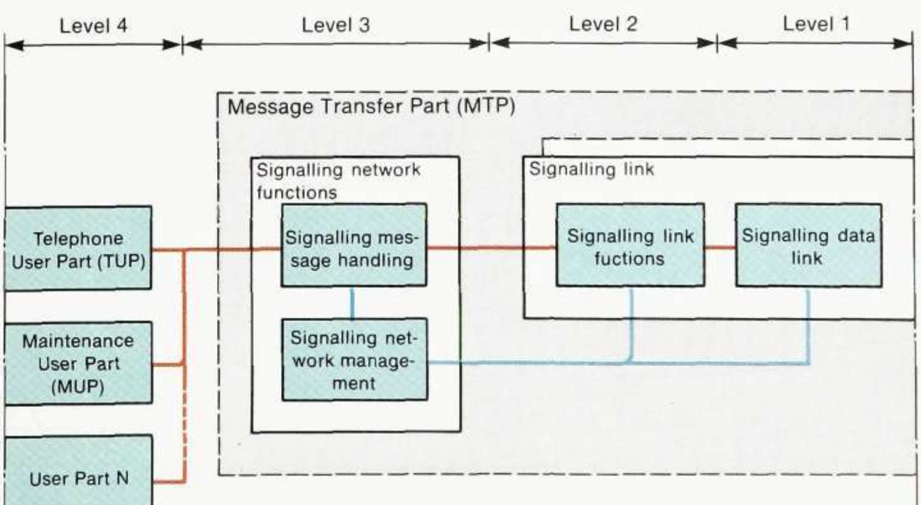
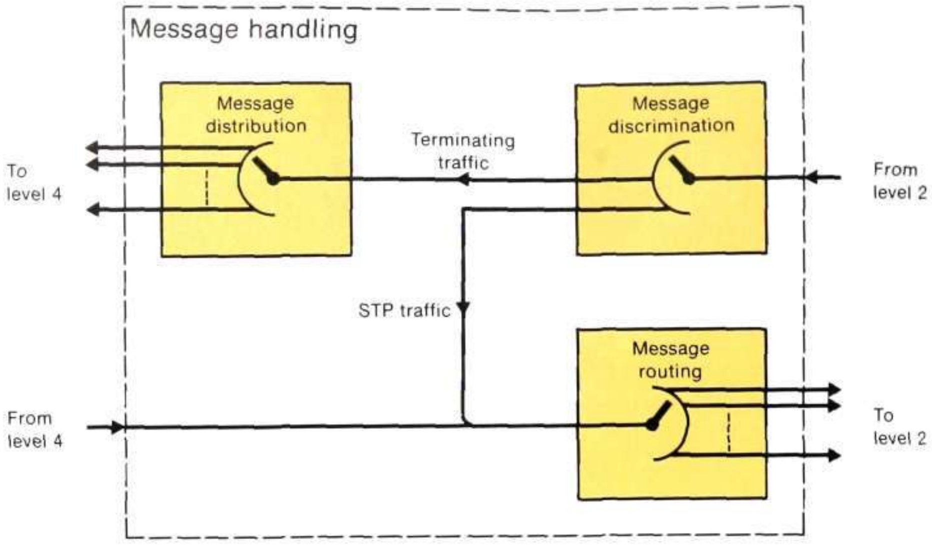


Fig. 3
Signalling system No. 7 has been designed with a functional structure that makes it suitable for telephone signalling, transmission of messages concerning operation and maintenance etc. An exchange can contain several UPs, one for each application. The structure also permits the successive incorporation of new fuctions

Signalling message
Internal control signals

Fig. 4
Message handling

Message routing:	The message label field DPC is analysed and the MSU is routed to the correct signalling link
Message discrimination:	The message label field DPC is analysed, and on the basis of this information the MSU is either terminated in the exchange or passed on through the signalling network (STP traffic)
Message distribution:	A received MSU which is to terminate in the exchange is distributed to the correct UP. This requires analysis of the service information



The error probability is less than 1 in 10^{10}. The UPs can thus regard MTP as a transparent transmission medium. Levels 1 and 2 together form a signalling link.

Level 3 comprises the signalling message handling and signalling management functions.

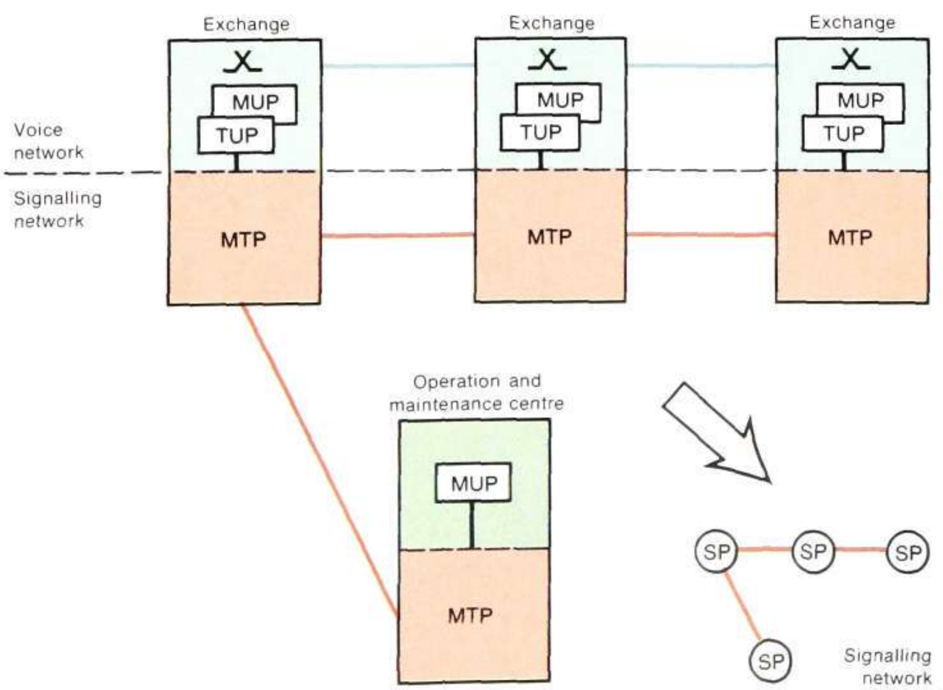
The function "signalling message handling" is the "switch" of the signalling system. In the send direction, i.e. when an MSU is received from a UP, the MSU is routed to the correct signalling link. In the receive direction, i.e. when an MSU is received from a signalling link, the correct UP is selected (message distribution). Message discrimination means that received MSUs that are to be passed on through the signalling network must be separated from the MSUs that are to terminate in the switching centre. Fig. 4 illustrates the signalling message handling in more detail.

The signalling network can be made very reliable through the use of alternative signalling links. The function "signalling network management" handles information regarding these alternative signalling links. If a fault occurs, it orders a reconfiguration of the signalling network. The signalling message handling function is then provided with new routing directives. All signalling points in the signalling network that are affected by the reconfiguration are also informed.

The future telecommunications network with signalling system No. 7 should be regarded as two logically separate parts: A voice network and a signalling network. The signalling network consists of signalling points (the exchanges etc. in the telecommunications network) which are connected together by signalling links, fig. 5. When signalling system No. 7 is introduced to supplement or to replace the conventional signalling systems, a signalling network is obtained which can also be used for a number of other purposes. For example, an operation and maintenance centre can be connected to the nearest exchange through a signalling link, and can then communicate with the other exchanges in the telecommunications network via the existing signalling network.

Fig. 5
A telecommunications network equipped with signalling system No. 7 should be considered as two separate networks, a voice network and a signalling network. The signalling network can be used for many other applications in addition to telephone signalling.
The signalling network consists of signalling links which interconnect the signalling points in the network. Each installation in the network constitutes a signalling point (SP)

Speech route
Signalling link
TUP Telephone User Part
MUP Maintenance User Part
MTP Message Transfer Part

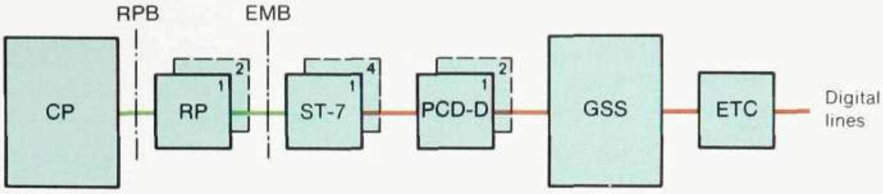


Implementation in AXE 10

CCITT signalling system No. 7 is presently being introduced in AXE 10. Initially signalling system No. 7 will provide for signalling between exchanges in digital national telecommunication networks. It is implemented by means of a new subsystem, Common Channel Signalling Subsystem, CCS, which corresponds to MTP, and an addition to the Trunk and Signalling Subsystem[2], TSS, for the function of TUP. Fig. 6 shows how signalling system No. 7 is to be introduced in AXE 10, and fig. 7 shows the hardware structure. Signalling system No. 7 in AXE 10 is de-

Fig. 7
Signalling system No. 7 in AXE 10, hardware structure.
The signalling data link consists of a 64 kbit/s timeslot in the PCM system (speech route). Commands are used to connect the signalling terminal via a semipermanent connection through PCD-D and the group swith to an optional timeslot in the speech route. RP and PCD-D are duplicated in order to ensure high reliability

CP Central processor
RPB Regional processor bus
RP Regional processor
EMB Bus between RP and the devices
ST-7 Signalling terminal for signalling system No. 7
PCD-D Multiplexor with 64 kbit/s digital inputs
GSS Group switch
ETC Exchange terminal circuit
Signalling message flow
Signalling data link



signed in accordance with CCITT recommendations[3].

TSS controls the connection and disconnection of telephone circuits. In TSS functions specific for signalling system No. 7 are the translation of the signalling information into the format standardized by CCITT, and the translation from the internal indication of the speech circuit in AXE 10 into the message label standardized by CCITT. AXE 10 can contain speech routes with several different conventional signalling systems and speech routes with signalling system No. 7. The functional content of TUP therefore corresponds to the R2/MFC signalling diagram recommended byt CCITT, which in turn covers most of the existing signalling schemes. A standard version of TSS can therefore be used for many markets. Market adaptation of TSS is made for markets with special signalling requirements.

Subsystem CCS consists of a number of function blocks divided into four different groups, namely signalling link functions, message handling functions, singalling network management functions, and operation and maintenance functions. CCS contains most of the functions specified for MTP by CCITT. These functions cover all the applications for signalling system No. 7 that are envisaged in AXE 10 within the foreseeable future. CCS has been designed to permit the successive incorporation of new functions and new UPs in the AXE 10 system.

The design of a signalling network must economically meet the requirements on reliability and capacity set by the user of the network. The signalling network should also be designed for ease of administration. The following description of signalling system No. 7 in AXE 10 is based on these requirements.

Economy

In a telecommunications network the need for communication between the exchanges is great. When signalling system No. 7 is introduced the amount of signalling equipment will be reduced because the signalling for a large num-

Fig. 6
Signalling system No. 7 in AXE 10.
Signalling system No. 7 in AXE 10 consists of subsystem CCS, which corresponds to MTP, and an addition to TSS, which corresponds to TUP

SSS Subscriber Switching Subsystem
GSS Group Switching Subsystem
TSS Trunk and Signalling Subsystem
CCS Common Channel Signalling Subsystem for signalling system No. 7
ETC Exchange Terminal Circuit
Signalling data link
Signalling message flow
Internal control signals in CCS
Speech circuit

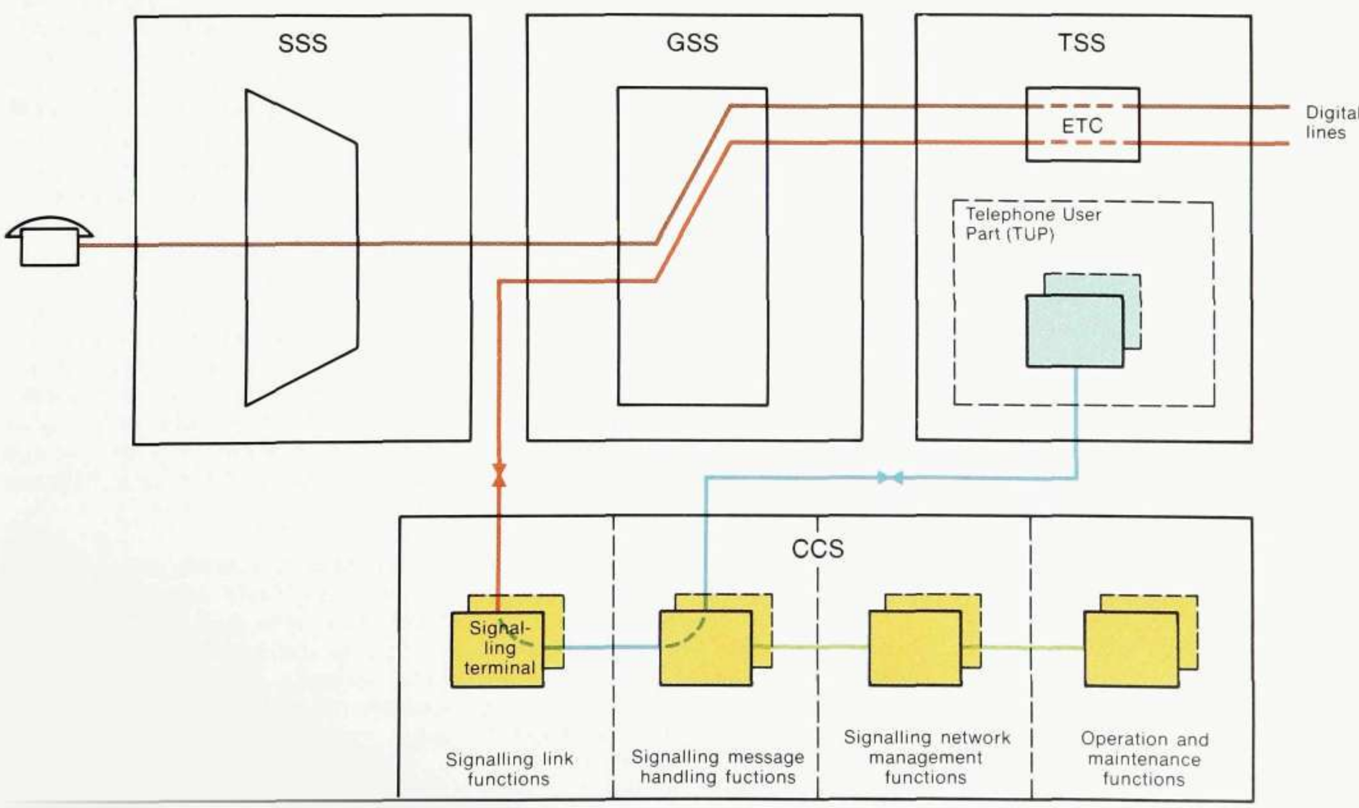
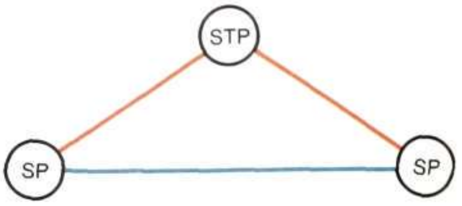


Fig. 8
Alternative signalling methods.
Associated signalling (to the left) and quasi-associated signalling (to the right)

Signalling link
Speech route
STP Signalling transfer point
SP Signalling point



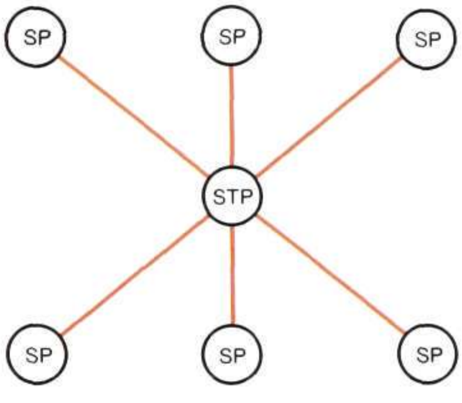
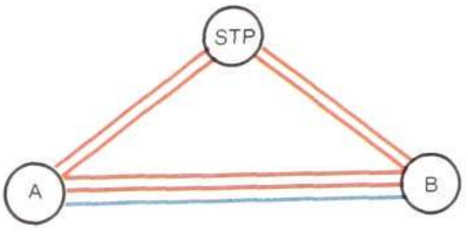


Fig. 9.
STP used as a message switch.
Quasi-associated signalling permits efficient utilization of the signalling links. The number of signalling links increases linearly (n−1) with the number of signalling points (n) in the signalling network. With associated signalling the number of signalling links increases linearly with the number of speech routes, i.e. in the worst case quadratically (n(n−1)/2) with the number of signalling points in the signalling network

Signalling link
STP Signalling transfer point
SP Signalling point

Fig. 10
An example of how the reliability in the signalling network can be increased by using two signalling paths for the speech route, one associated between A and B and one quasi-associated over STP. Each signalling path can have double signalling links

Signalling line
Speech route



ber of speech channels is transmitted over one common signalling link.

Signalling system No. 7 can be used in two different ways in AXE 10: Associated signalling and quasi-associated signalling, fig. 8. This enables the user to build up a signalling network where the signalling links are utilized very efficiently. Associated signalling means that the signalling link runs in parallel with the speech route, i.e. the signalling link transmits the signalling for one speech route. Quasi-associated signalling means that the signalling route is separated from the speech route. The signalling route thus consists of two or more signalling links in series, connected together by one or more signalling transfer points, STP. STP receives incoming MSUs and sends them out on new signalling links without changing the information content, fig. 4. With quasi-associated signalling, the signalling points in the signalling network are thus connected to a few STPs, and the signalling links are shared by a large number of routes, fig. 9.

The signalling network will normally be based on quasi-associated signalling. However, it may be advantageous to use associated signalling for very large speech routes and thus reduce the load on the STPs.

Reliability

Common channel signalling implies *concentrating the signalling to a few* signalling links. In most cases the signalling network must then be equipped with spare signalling links to prevent individual component faults from completely interrupting the telephone traffic.

Fig. 10 shows how the reliability of the signalling network can be improved. The signalling between the signalling points in AXE 10 can be routed over different signalling paths, and each such path can contain two signalling links.

Faults in signalling networks are usually handled by automatic rerouting of the signalling traffic. Changeover means that the traffic on a faulty signalling link or path is transferred to the other signalling link when the signalling path has double signalling links, or to the other signalling path. When the fault has been cleared, the traffic is automatically transferred back to the original signalling link ("changeback"). Changeover and changeback are carried out in a controlled manner so that no MSUs are lost and their sequence is not disturbed.

Thus the reliability of the signalling network is a matter of dimensioning and is not limited by the system concept.

Capacity

The signalling links consist of 64 kbit/s timeslots in PCM systems. This corresponds to a very high signalling capacity, sufficient to handle approximately 10000 telephone circuits. The signalling traffic in AXE 10 can be divided equally between signalling links, "load sharing". Load sharing increases the signalling capacity, but is primarily used to distribute the traffic equally throughout the signalling network, and hence reduce the risk of occasional overloading.

Administration of the signalling network

A signalling network should be divided into sections for easier administration. The aim is to administer the sections independently as far as possible. Signalling system No. 7 is so specified that the signalling network can be divided into hierarchic levels. The telecommunication networks are by tradition *divided into an international network* and national networks. With signalling system No. 7 the national networks can be used for signalling between exchanges and for signalling concerning, for example, centralized operation and maintenance and centralized administration and control of subscriber facilities. AXE 10 also permits another level which can comprise local or regional networks, connected to the national network.

Fig. 11 shows a possible division of the network into levels, where each local network is connected to a parent exchange in the national network. The local networks can include signalling between the parent exchange and PBXs etc.

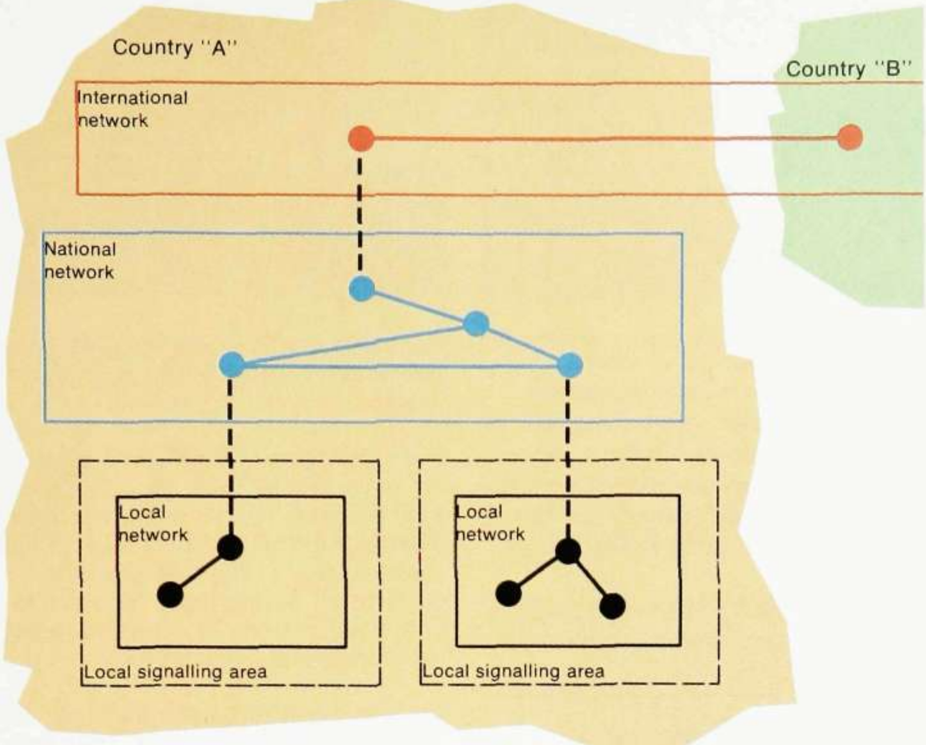


Fig. 11.
A signalling network divided into hierarchic levels.
The administration of the telecommunications network is simplified by dividing it into sections. Signalling system No. 7 in AXE 10 gives the administrations freedom to divide the telecommunications network into hierarchic sections, which can be administered independently of one another

- ● Signalling point in the international network
- ● Signalling point in the national network
- ● Signalling point in the local network

respect to administration. Signalling points, signalling links and signalling routes are assigned one level and are identified (numbered) internally within this level.

Operation and maintenance

When new traffic requirements arise exchange data can be modified by means of commands. New signalling routes, new reserve links etc. can thus be intruduced without disturbing the traffic in progress.

In many fault situations the system corrects itself without manual intervention (changeover). Examples of such situations are sporadic transmission failures on signalling links as well as service interruptions in interworking exchanges. Fault printouts are obtained for serious faults in the signalling network.

Commands are also provided so that maintenance staff may, at any time, question the system about the condition of the signalling network and the exchanges, and also current exchange data. In addition all operator information can be routed to maintenance centres an be presented on suitable displays or I/O devices.

Future prospects

The version of signalling system No. 7 described here will be taken into service in 1983. The first application will supplement and replace the conventional signalling systems. This provides a rationalization gain even in the short term. Moreover, the foundation will be laid for a signalling system that will give a radical improvement compared with the situation in the existing network.

The new possibilities offered by signalling system No. 7 will stimulate the development of telecommunication networks, and as a result the administrations will be able to introduce new subscriber facilities and functions[4].

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Optical Transmission Link in ATC Systems

Gunnar Forsberg, Torbjörn Larsson and Karl-Erik Sundvall

Ericsson's automatic train control system, JZG 700, has been described previously in Ericsson Review[1]. In this system information is transmitted from information points, beacons, on the track to antennas on the locomotives. Normally the beacons receive their information from railway lamp signals via multi-wire electrical cables. An optical transmission link has been designed for the system for use in the cases where the distance between the signal and the beacon is so large that the ordinary electrical cable cannot be used because of interference from the traction current.
The authors describe the need for and demands on such a transmission link, the design of the new link, the design of the optical cable and various possible ways of installing the cable.

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621.315:535.3

The Nordic Railway Administrations are now installing an automatic safety system which will control the speed of the trains (ATC). The system chosen is manufactured by Ericsson and designated JZG 700. It consists of

- a transmission system with inductive transmission of data from the track to the locomotive at fixed points, beacons, placed along the track
- a microcomputer-controlled locomotive unit with a driver's panel for supervision of the speed of the train and with automatic braking if the train speed is too high.

Descriptions of the system have been published in earlier issues of Ericsson Review[1,2].

The supervision of train speed is based on various data, such as information from the lamp signals along the track. This information is transmitted to the beacons, fig. 1. The beacons also include fixed, programmed information. Both types of information are transmitted to the passing locomotives.

The information from the lamp signals to the beacons can be transmitted over an electrical or an optical transmission link.

The electrical link contains an encoder which scans the lamp currents in the light signal. The encoded data are transmitted in a multi-wire cable. The link has a simple design, requires little current and contains few circuits, which gives it high reliability. It is suitable for most installation cases.

The disadvantage of the electrical cable is that the transmission distance is limited to approximately 300 m because of interference from the traction network,

Fig. 1
Optical cable for transmitting information from the signal to the beacon

GUNNAR FORSBERG
Fibre optics and line transmission
Telefonaktiebolaget LM Ericsson
TORBJÖRN LARSSON
Sieverts Kabelverk AB
KARL-ERIK SUNDVALL
Department for signalling systems
Telefonaktiebolaget LM Ericsson

short circuits in the network and earth faults.

An optical transmission link used for transmitting the same information does not limit the distance in the same way. Fig. 2 shows a comparison between an electrical and an optical fibre link.

Optical transmission link: system and applications

The optical system comprises:
- optical transmitter
- optical receiver included in the beacon
- optical single-fibre cable, including installation material
- splicing equipment for the fibre cable
- test instruments for installation and maintenance
- optical fibre connector.

The system characteristics can be summarized as follows:
- transmission distance < 3 km
- the transmitter is powered from the lamp signal or by a separate source
- the transmitter is matched to the existing encoder output towards the beacon
- the transmitter is adapted to the construction practice in existing buildings for signalling equipment.

The environmental requirements are severe, see table 1.

One unique feature of the system is that only one optical fibre is used and that the synchronization of the signal from the optical transmitter with the continuously transmitted scanning signal from the locomotive takes place in the beacon. Synchronization could have been obtained by transmitting the synchronization pulses from the locomotive via the beacon to the transmitter, but that would have required another fibre in the cable and twice as many transmitters, receivers and fibre splices.

The cable can be installed in different ways. One way which has been found to be successful is to place the cable in the waist of the rail. However, running the cable along the track and in the rail waist is not an innovation. There are ATC systems which use electrical cable installed in this way for continuous transmission of safety data between the track and the locomotive. Figs. 3 and 4 show the structure of the new optical link.

Design

OPTICAL TRANSMITTER

The transmitter receives data in parallel form, as three eight-bit words (X, Y, Z). The words can be controlled by relays, ATC encoders or a fixed strapping. The parallel/series conversion is carried out and a synchronization word is added, which gives a 32-bit telegram. The conversion equipment is designed in accordance with the same principles as the equipment in a beacon controlled via an electrical cable.

Table 1
Environmental requirements for the optical transmission link

	Transmitter	Beacon	Cable
Temperature, °C	−40-+70	−40-+70	−30*-+80
Shocks, m/s²	50	300	420
Relative humidity, %	10-95	10-100	10-100

* Temperature requirement for field trial systems

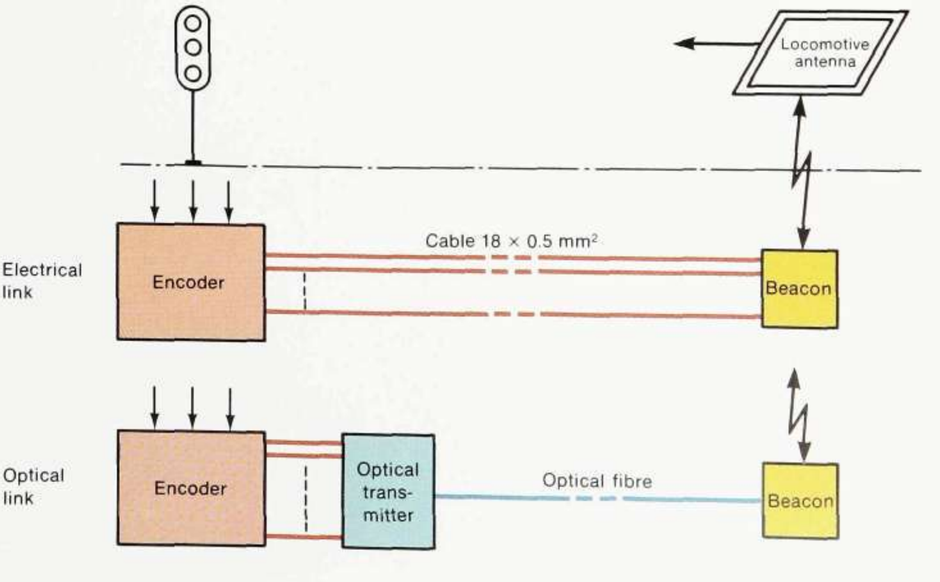


Fig. 2
A comparison between an electrical and an optical fibre link

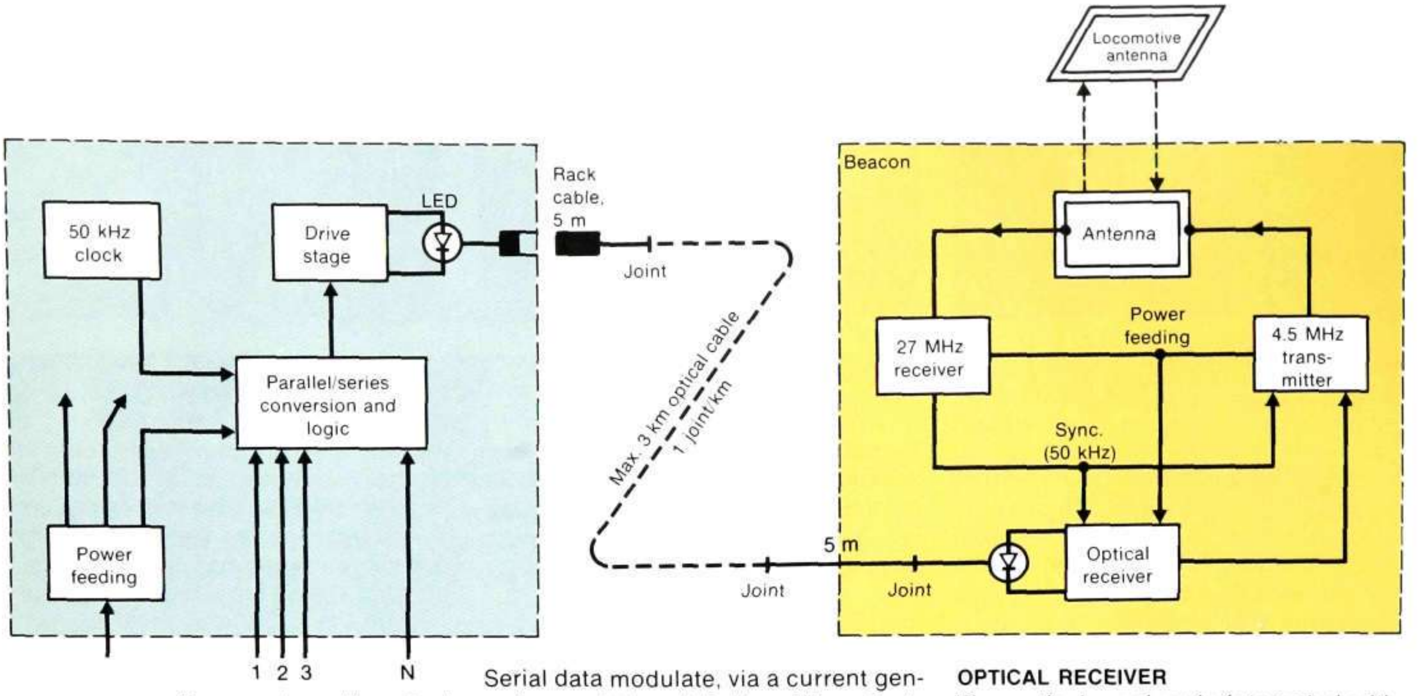


Fig. 3
A block diagram of optical fibre link equipment

Serial data modulate, via a current generator, an infrared light-emitting diode with short pulses. The LED is optimized for low power consumption, good coupling to the fibre and long life. The logic circuits are controlled by a 50 kHz clock, which gives a bit rate of 50 kbit/s for the transmission to the beacon. The transmitter can be power fed by the encoder, which has a limited supply of power (approximately 1 W).

Technical data for the optical transmitter, fig. 5, are:

– Wavelength	850 nm
– Pulse width	3 μs
– Optical pulse power	>25 μW
– Possible power reduction	10–15dB

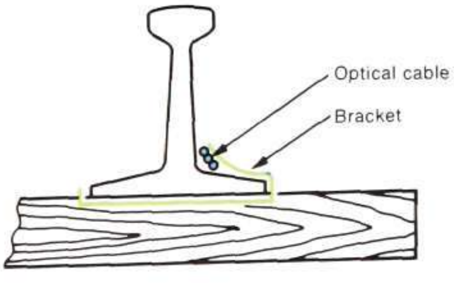
The light-emitting diode is alight approximately 7% of the time (varies slightly with the information content), which means that 25 μW pulse power corresponds to approximately 1.8 μW mean power. The low mean power ensures that there is very little probability that the optical power of the LEDs will decrease with time. The reliability is therefore high. The output power can be reduced further if the fibre route is short.

OPTICAL RECEIVER

The optical receiver is integrated with the rest of the beacon electronics. The received serial data are detected in a sensitive optical input stage and are converted in an asynchronous/synchronous converter, which is controlled by a clock signal. The clock signal, like the power feeding for the receiver and other electronic circuits in the beacon, is obtained from a 27 MHz carrier, which is transmitted by the equipment in the locomotive when it passes over the beacon. The output from the optical reciver modulates a 4.5 MHz carrier which transmits data in serial form from the beacon to the locomotive.

The optical receiver, fig. 6, meets the following requirements:

- low power consumption, 10 mW (1 mA, 10 V)
- correct data detection 0.5 ms after the beacon has received the feeding voltage. A high-speed train can pass the beacon in less than 6 ms
- a high degree of sensitivity, the sensitivity limit must be better than 5 nW pulse power, which corresponds to approximately 0.4 nW mean power
- a large dynamic range for detecting optical signals with a pulse power of



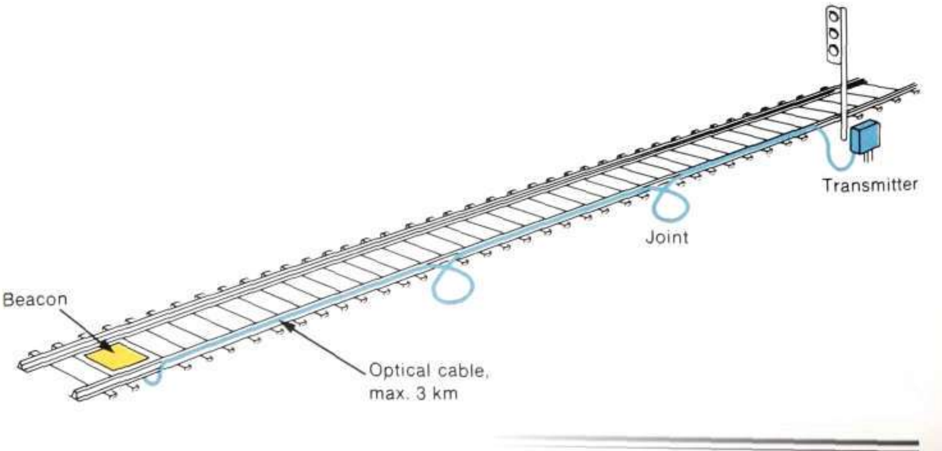


Fig. 4
Installation of the optical fibre link equipment between the signal and the beacon

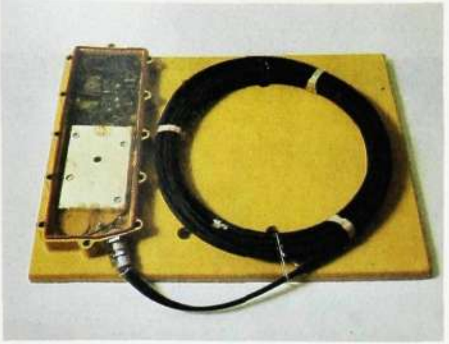


Fig. 6
Beacon with the optical receiver

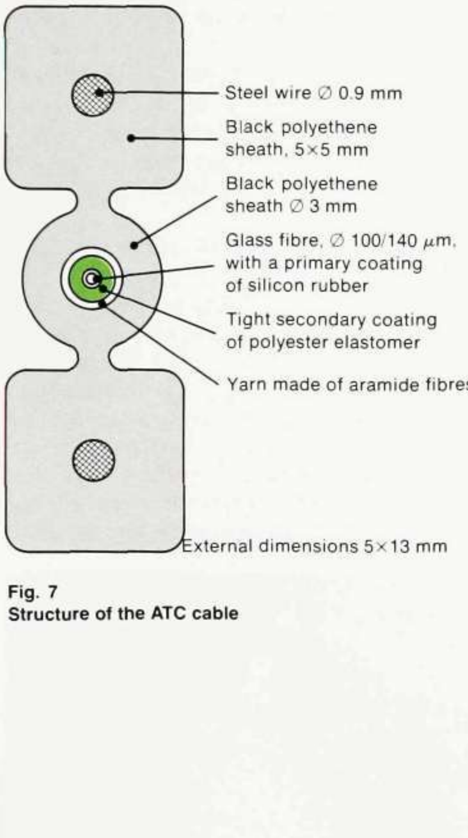


Fig. 7
Structure of the ATC cable

between 5 nW and 50 μW (corresponds to a 10000 times change of signal amplitude). The receiver must quickly adapt to instantaneous changes in amplitude in order to meet the second requirement above
- synchronization of the input signal to the beacon with the clock frequency of the locomotive equipment
- ability to withstand large variations in climate.

OPTICAL FIBRE CONNECTOR

The optical connector matches the fibre cable to the light emitting diode in the transmitter. Two types of connectors have been tried in the system. Moisture-proof mounting of the LEDs and the fixing devices was one of the design aims. The results of temperature cycling tests in high humidity show that the function is not impaired even in severe cold.

The optical fibre connector constitutes a test point for the transmitter power and the digital data from the transmitter, and also a testpoint for the fibre cable and beacon characteristics.

CABLE DESIGN

Fibre

The transmission medium is a double crucible (DC) glass fibre for industrial applications. The Swedish cable manufacturer Sieverts Kabelverk develope and manufacture fibre and fibre cables[3].

The DC fibre has good transmission characteristics for optical signals. The attenuation is 7–9 dB/km at a wavelength of 850 nm.

The fibre consists of a multi-component glass core with a high refractive index, covered with a glass cladding having a lower refractive index. The cladding also contains small quantities of various additives which greatly increase the ability of the fibre to withstand the chemical effects of the atmosphere. The additives also increase the mechanical strength.

A primary coating of silicon rubber protects the fibre against microcracks and dust particles. The coating is applied when the fibre is drawn. The strength is tested during the manufacture by the whole length of fibre being elongated by 0.5–1%. A soft but tough secondary coating, a large diameter and a large numerical aperture (NA) are the main factors that prevent any increase in attenuation as a result of the cabling and the laying of the cable. The increase in attenuation caused by the connection of the fibre to the light source and any splicing is also kept low.

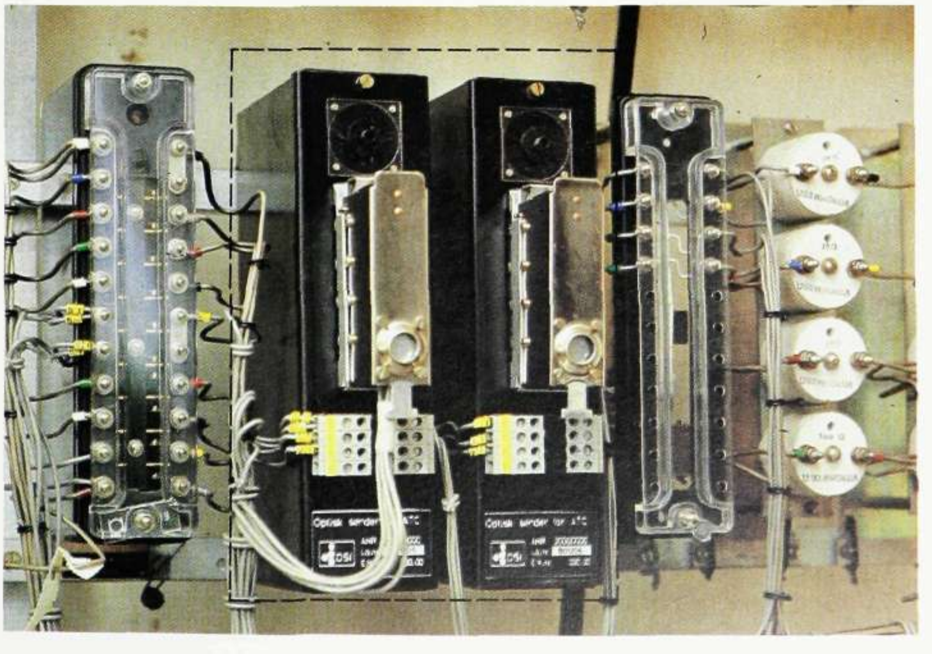


Fig. 5
Two optical transmitters

Fig. 8
A bracket holds the cable in place

Cable
A flat cable structure, fig. 7, was chosen because of the special laying method. The fibre with is primary coating is given a secondary, extruded coating consisting of an extremely durable plastic (polyester elastomer). Aramide yarn is then stranded round the fibre, followed by a layer of polyester tape. Finally the cable body is covered with a sheath of black polyethene. In this process the fibre with its protective layers is fed between two parallel steel wires through a nozzle, where the black polyethene is extruded to the desired profile. The wires take the strain during the cable laying and also any strain caused by temperature variations. The sheath gives protections against mechanical damage and the environment. The centre part of the cable, which contains the glass fibre, is smaller than the two outer parts, which therefore protect the centre against wear.

The ATC system also required a lighter, more flexible cable. It was to be used for wiring in the beacon and in racks and shelves on the transmitter side. This cable has the same structure as the cable body in the flat cable and a sheath of durable polyurethane.

Cable laying

The normal way of laying a cable is by means of burying. Cables for railways can either be laid in the roadbed or in cable conduits by the side of the track. In the first case the cable is usually ploughed in, and only short distances have to be dug by hand.

Other methods can be used for laying optical cables. There are two alternatives, both of which utilize the insensitivity of the fibre as regards traction current:
- running the cable along the rail, with bracket fastening, fig. 8
- running the cable on a pole line with its own suspension wire or on the return wire for the traction current.

The advantage of the first method is that the digging cost is avoided. This cost is high, particularly if much hand digging has to be done. However, other costs are incurred instead, such as the alteration cost when rails are changed and, possibly, increased maintenance costs for damaged fibre cables. The cable must be laid in such a way that mechanized track maintenance will not be prevented by the cable, nor affect it.

Fig. 9
Splice fusing unit with a fixture for fibre with secondary coating or a small round cable

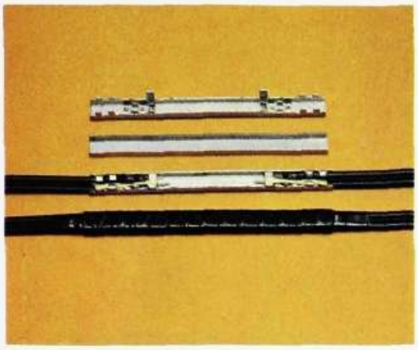
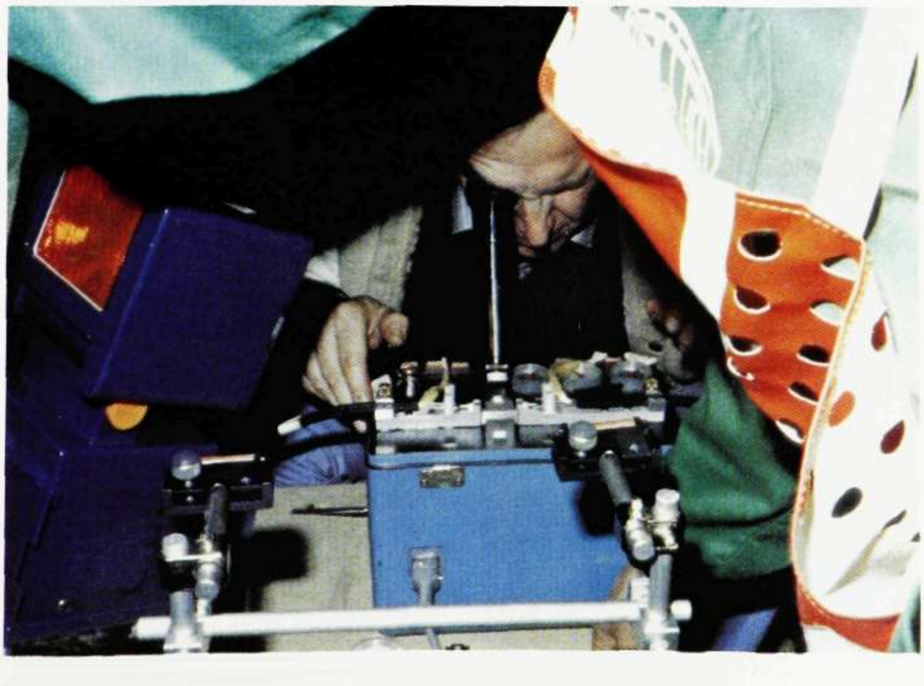


Fig. 10
Joint sleeve

Fig. 11
Jointing cable in the field



In the case of the second alternative using suspended cable, consideration must be paid to the climatic conditions (wind, cold, snow etc.) that make maintenance more difficult. For example, what action could be taken in the case of a break in the fibre if it is not possible to disconnect the traction current because of the railway traffic? In the autumn of 1981 a trial with suspended cable was started on a cable section in Stockholm, Sweden. The cable was suspended from the return wire for the traction current. The trial results have yet to be evaluated.

In Denmark a technical/economic investigation was held which resulted in a decision to carry out a field trial with the cable laid in the waist of the rail and fastened with spring steel brackets, figs. 4 and 8. The bracket is installed first, then the cable is run out along the rail and is finally inserted under the bracket. The installation can be carried out without any interruption in the railway traffic if guards are posted.

At exposed points, for example rail joints, the cable is protected against wear by means of a piece of polyethene tubing. If the metallic conductors in the cable have to be insulated galvanically, the cable can be jointed by cutting the two steel wires and encasing the joint in epoxy resin to maintain the original strength.

Another way of fastening the cable to the rail is by means of so-called stud brazing. A special "brazing gun" is used to braze a cable clamp to a suitable spot in the waist of the rail. Both the brazing and the cable laying can be carried out more or less automatically from a rail vehicle. This method is approximately 3–4 times faster than the previous method and gives better friction against cable movements. Field trials have shown that the cable may other-wise move as a result of vibrations from trains on tracks with one general direction of traffic (double track).

The installation can be simplified by manufacturing custom-made cables with the desired length and factory-mounted connectors. Such cables should also be used when immediate repairs are necessary.

Jointing

A special jointing method has been developed that is suitable for the cable profile, the laying method and the severe environmental requirements.

The first stage in the jointing is to prepare the cable ends, after which they are placed in a fixture in a splice fusing unit, fig. 9. The glass fibres are fused together by means of an electric arc.

The melting temperature of DC fibre is relatively low since it consists of multi-component glass. In order to obtain a good splice the energy must be concentrated at the fibre ends and the fusing time must be fairly short (0.15 s). Prefusion is not required. The arc is ignited with a small gap between the fibre ends (5–10 μm). The gap is closed by the thermal expansion of the fibre. The splice loss between identical fibres is less than 0.20 dB. This fusing method has been described in detail previously[3].

The spliced fibre is placed in a stainless steel sleeve, fig. 10, with a small excess of fibre so that the splice is not stretched. The steel wires and the aramide yarn are then fixed to the sleeve by means of a clamping tool. The sleeve is

Table 2
Trial routes and evaluation

Trial routes	Denmark				Sweden
	A	B	C	D	Stockholm
Structure					
Length (m)	10	3000	520	1683	1000
Number of fibre joints	1	4	2	3	1
Number of fibre connectors	1	1	1	2	
Complete ATC system	+	+	+	–	–
Evaluation					
Trial runs with ATC trains	+	+	+	–	–
Recording of short-term and long-term stability	–	+	–	+	+
Track maintenance	+	+	+	–	–
Change of rails (planned)	–	–	–	+	–

filled with vaseline and closed with a stainless steel lid and a shrunk-on tube. The completed cable joint can withstand temperature variations and also water, ice and mechanical shocks. The joint can be placed direct on the rail or buried by the side of the track.

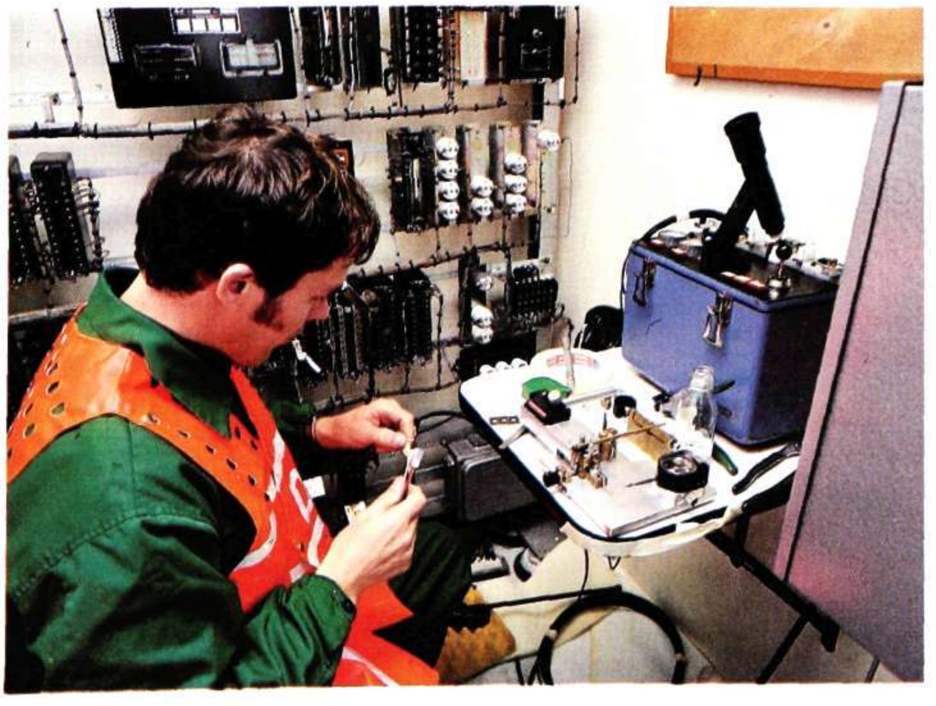
It takes approximately 30 minutes to complete a cable joint. Great care must be taken to arrange the work site so that the shortest possible work and preparation times are obtained.

Measuring methods in the field, fault localization

Cable checks can be carried out in connection with the jointing. Both cable and jointing losses can be determined with a simple transmission measurement. The test transmitter must be stable and have a well-defined light input to the fibre. Both the test transmitter and the test receiver are battery operated and are equipped with cable and fibre sockets for easy connection.

The present system design does not require bandwidth measurements on installed cables.

Fig. 12
Jointing fibre cable in a building for signalling equipment



The installation time can be reduced by making measurements only on jointed cable sections.

Fault localization, for example if there is a break in the fibre, is carried out with the aid of a time domain reflectometer (TDR). Breaks can then be found with an uncertainty of only ±5 m.

When the installation testing of the cable has been completed, the optical output power of the transmitter is measured. The system margin can be checked by placing a variable optical attenuator between the transmitter and the cable. A test instrument records the function of the beacon. The optical output power is reduced until functional errors occur. The system margin is calculated on the basis of the difference in output power from the optical transmitter and the optical attenuator.

Completed field trials

Stockholm, Sweden,
The Swedish State Railways provided a trial route at Duvbo, near Stockholm, for testing the installation method and investigating whether passing trains would disturb the transmission in the glass fibre. This particular route was chosen because it had a high traffic volume and was due for maintenance. Large rail movements could therefore be expected when trains passed.

In January 1980 a 1 km long cable was run in a loop along the rail. The cable ends were connected to recording equipment, which continuosly monitored the cable and which was sensitive to rapid changes in attenuation caused by, for example, a passing train. Such interference will cause modulation, which if significant, may deteriorate the transmission performance. The recording continued until July 1980, when the trial had to cease because the rails on the route were to be changed. No attenuation changes were obtained with a threshold level of 1% and an upper frequency limit of 10 kHz. In other words the result was favourable for further trials with the cable.

Denmark
In October 1980 a large-scale field trial

was started by the Danish State Railways, Dansk Signal Industri, Ericsson and Sieverts Kabelverk AB.

Three optical link systems with a total of 5 km of cable were supplied for the trial route. The aim of the trial was to study the effect on the systems of the climate, train traffic and maintenance.

During the installation attenuation measurements were carried out before and after the cables were jointed. A total of 10 joints were made. Five were made in buildings for signalling equipment situated by the side of the track, fig. 12. Using a tent for protection the flat cable was jointed in five places. The joints and the extra cable at the joints were buried. System testing was carried out after the cable, beacons and encoders had been installed, table 2.

Another system test was carried out in June 1981. The output power of the light emitting diodes had not decreased, and the sensitivity limit of the beacon receivers had not changed.

When repairing a damaged cable on route C, a cable joint was mounted in the waist of the rail. This test mounting proved satisfactory.

Long-term recording of changes in attenuation was carried out on the high-traffic route D during November 1980 and June 1981. The changes were insignificant and no faults were recorded.

In August 1981 route B was treated with a machine that tamps down the crushed rock on the roadbed, with simultaneous recording of the attenuation. No changes were observed with an alarm threshold of 0.5 dB and a time constant of 0.5 ms. The attenuation on the route was 31.1 dB at the time of installation, 31.5 dB before the roadbed treatment and 31.2 dB after. A cable joint was removed and tested in a laboratory. No deterioration could be detected.

A train equipped with test equipment for the ATC system travels routes A, B and C regularly. No faults have been reported.

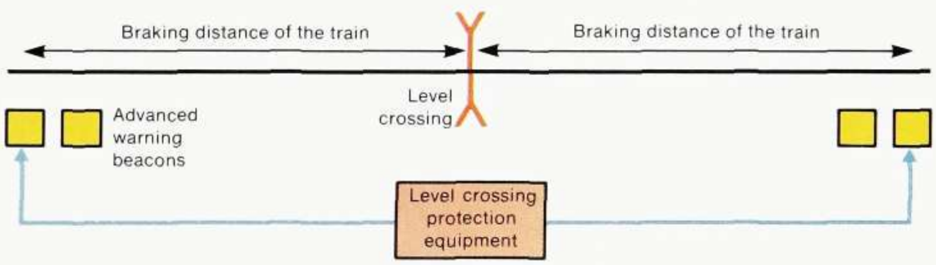


Fig. 13
System application for level crossings
—— Optical cable

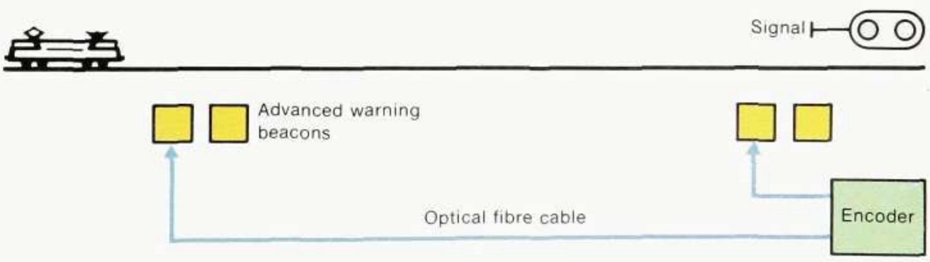


Fig. 14
System application for additional advanced warning

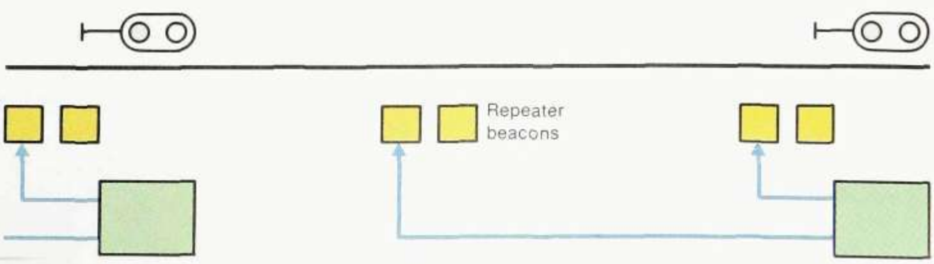


Fig. 15
System application for beacons placed between signals

Technical data for the fibre cable

Number of fibres	1
Type of fibre	Step index
Core/cladding glass diameter	100/140 μm
Numerical aperture	0.30
Attenuation (850 nm)	<10 dB/km
Pulse dispersion	20 ns/km
Minimum bending radius	100 mm

Fig. 16
Optical fibre beacon installed at a beacon site which contains two beacons

System aspects

The link between the light signal and the beacon must be at least of the same quality as the other equipment in the ATC system so as not to reduce the performance of the system. The system has therefore been designed with a high level of both safety and reliability. For example, the link is designed so that if there is a break in the cable or an electrical fault occurs, the beacon will transmit a bit pattern to the locomotive which shows that the beacon is faulty. In this way the fault cannot affect safety, and the fault can be recorded and located.

A data bit must be inserted or removed periodically in order to obtain synchronism between the locomotive unit and the optical link. This does not affect the safety or the reliability, since the control information in the locomotive is evaluated from a number of repeated telegrams.

After the parallel/series conversion the link stores the information, but only one bit at a time. This means that out-of-date information in the form of a whole word will never be stored in the link, which is one of the safety prerequisites.

The optical link is compatible with the electrical link but has a wider field of application. For example, it can be used to

- transmit advanced warning information concerning level crossings (usually at a distance from the crossing that is equal to the braking distance of the train) fig. 13
- transmit advanced warning information to the train when the distance between the signal and the advanced warning signal is large, fig. 14
- transmit information to a beacon situated between two signals in order to increase the capacity of the track. An initiated braking can then be cancelled quicker, fig. 15.

It is also possible to connect several repeater beacons to the same fibre cable and encoder by using optical branching equipment. This provides sophisticated supervision facilities, for example at the approaches to stations.

The fibre cable also permits the use of the alternative laying methods described above.

The optical link has a role to play in future ATC systems because of its insensitivity to interference. The field trials have also proved that the optical link is a reliable and stable part of the ATC system even in extreme environmental and operating conditions.

Further work is now in progress on ergonomic aspects and installation aids, as well as means of improving the installation and maintenance methods, among other things in order to find the best way of fixing the cable to the rail.

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The Railway Traffic in Melbourne

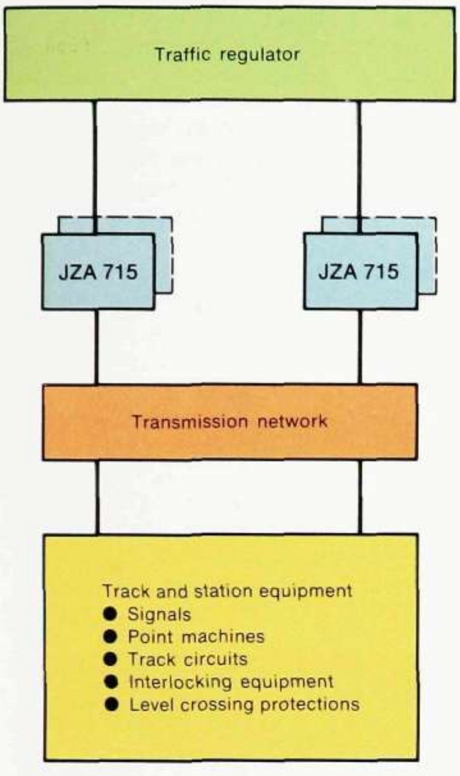


Fig. 2
Block diagram of the control and supervision system

Melbourne is the capital of Victoria in South-Eastern Australia. The city has a population of almost 2.7 million and consists of a downtown area at the head of Port Phillip Bay, surrounded by residential areas with mostly single family dwellings. Being a seaport and an industrial centre the city provides many employment opportunities, and the movement of traffic to and from the downtown area is heavy.

The Victorian Railway has an extensive suburban rail network connecting the suburbs with downtown Melbourne. The network extends over 330 route kilometres with a traffic density of about 2000 electric trains per day. The congestion of passengers, especially in the vicinity of Flinders Street station, creates difficult traffic problems mornings and afternoons.

To relieve the streets of the traffic MURLA (Melbourne Underground Rail Loop Authority) was formed at the beginning of the 1970s. This organization has planned and is in the process of executing a project in which most of the suburban lines are routed through the central business district in a loop of four tunnels, fig. 1. The passenger flow will then be distributed through five stations within the loop. The project has now reached the stage where two tunnels are in service.

In conjunction with this tunnel project, the train control and signalling system of the suburban lines is being modernized. One phase of this modernization program was the delivery from Ericsson of two control and supervision systems JZA 715 in January 1982. The system includes train describer, control and display, figs. 2 and 3. About 60 stations with approximately 4000 different supervised or controlled objects, such as track circuits, signals and point ma-

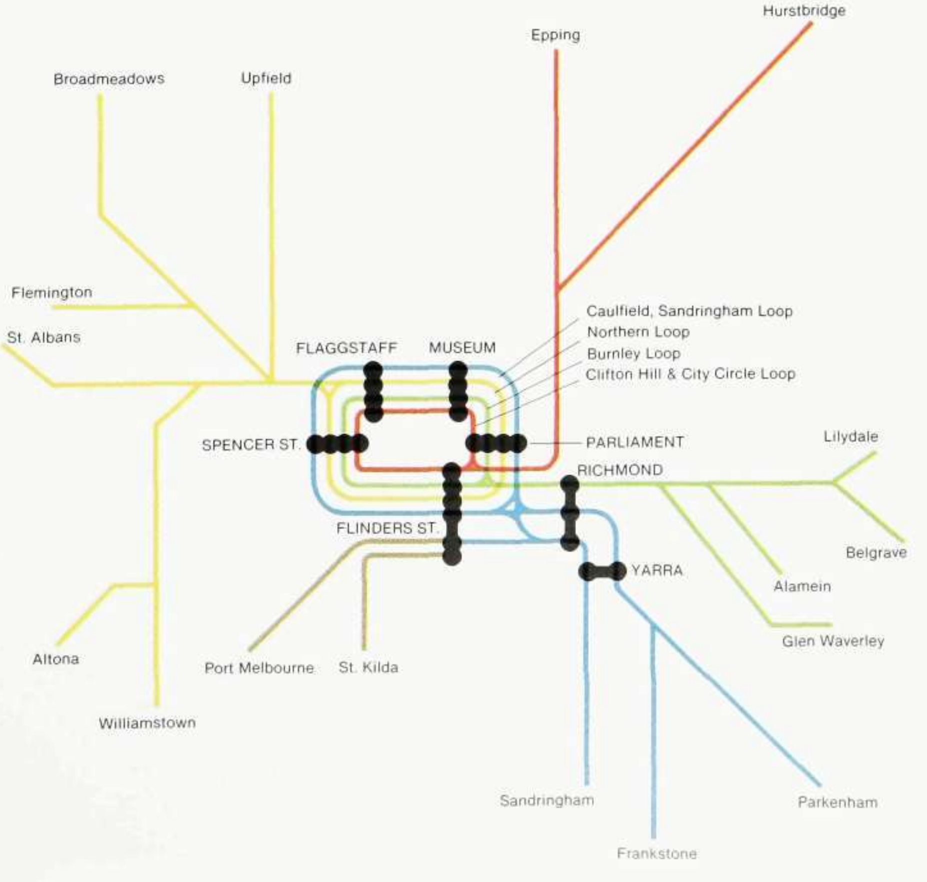
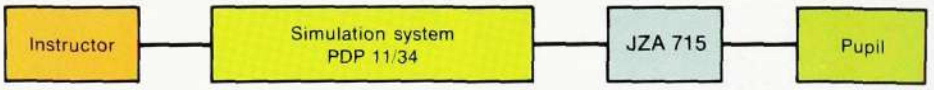


Fig. 1
Map of the railway network in Melbourne

Fig. 4
Block diagram of the simulation equipment



chines, are presently included. The final testing of the control and supervision system is presently being carried out by the Victorian Railways.

In parallel with the testing, the traffic regulators are being trained in the use of the system. This will continue until all staff has been trained. For the training another JZA 715 system from Ericsson is used. This system forms part of a simulation model controlled by a PDP 11/34 computer, fig. 4. It is possible for the instructors training the regulators to vary different parameters, such as time table, train speeds, delays and train movements. The train delays for a training session are accumulated and the total can be used as a measure of the skill of the trainee. By varying the train speeds the simulation can be carried out at speeds other than the normal. In this manner it is possible to study the effects of, for example, different loading. The simulation equipment is constructed so that it can also be used in tests and for checking time table layouts.

By changing the input data the simulation system can be used with any other optional railway installation and for studying different traffic situations and track layouts. Ericsson has an option permitting them to use and market the simulation system for such purposes.

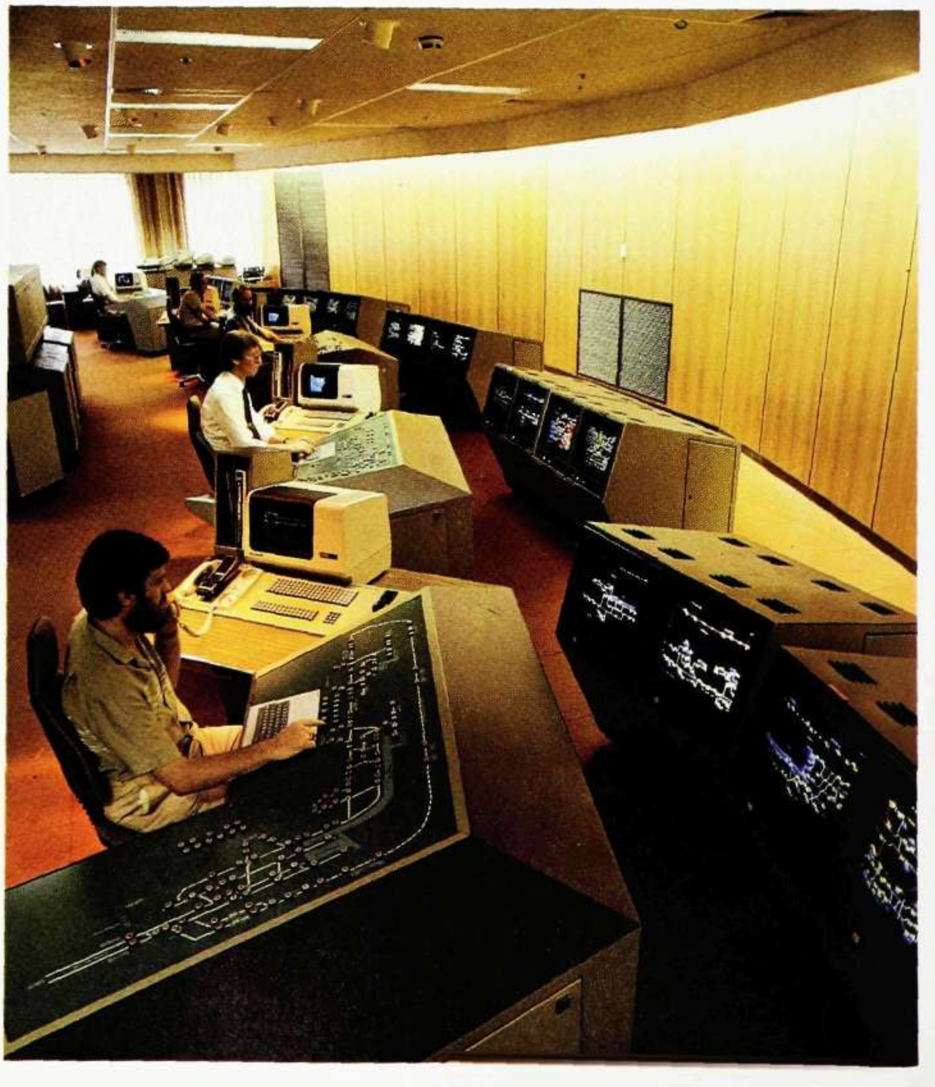


Fig. 3
Traffic regulators, control consoles and display equipment

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ERICSSON REVIEW

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EMERGENCY TELEPHONE SYSTEM SOS 512
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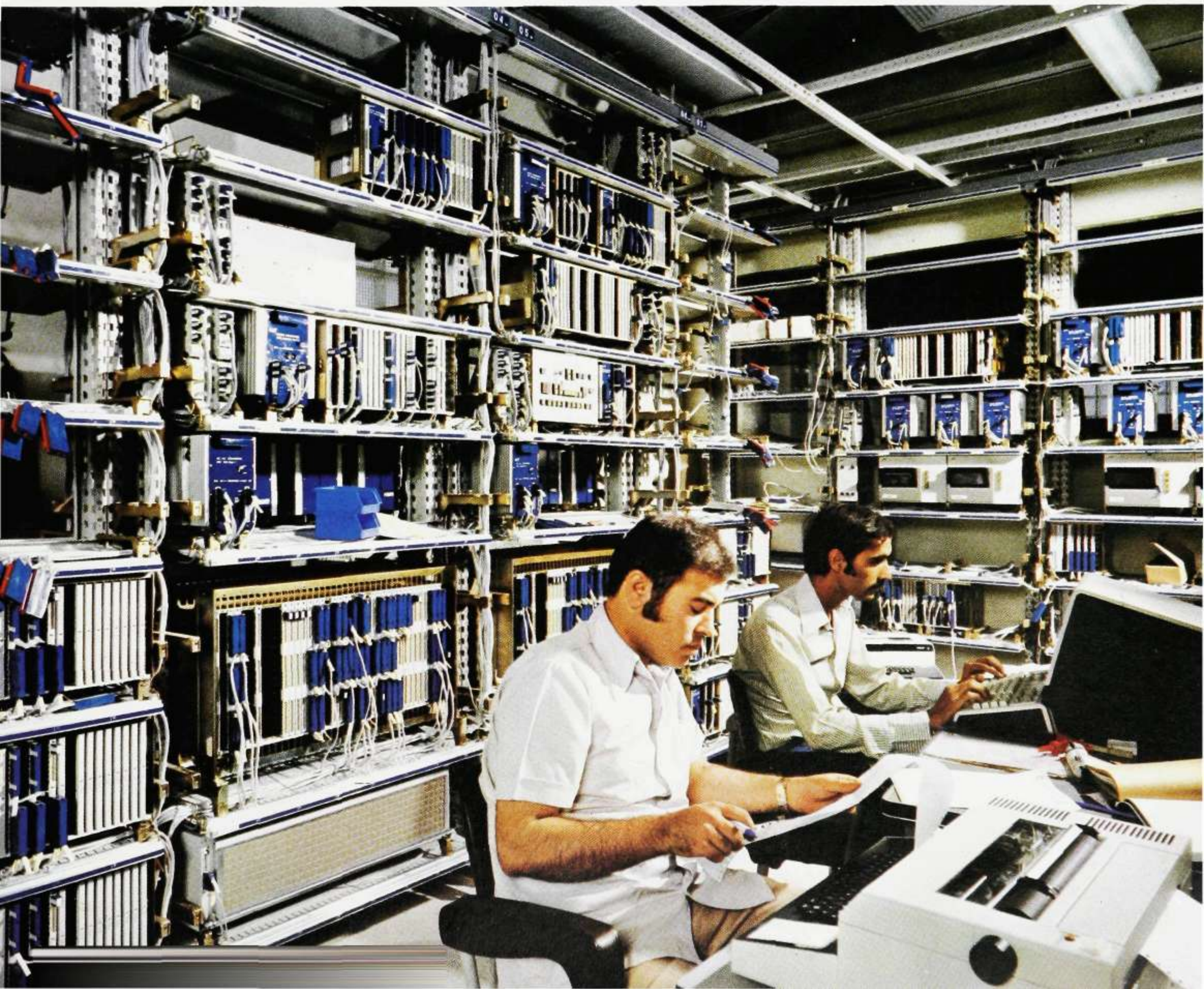

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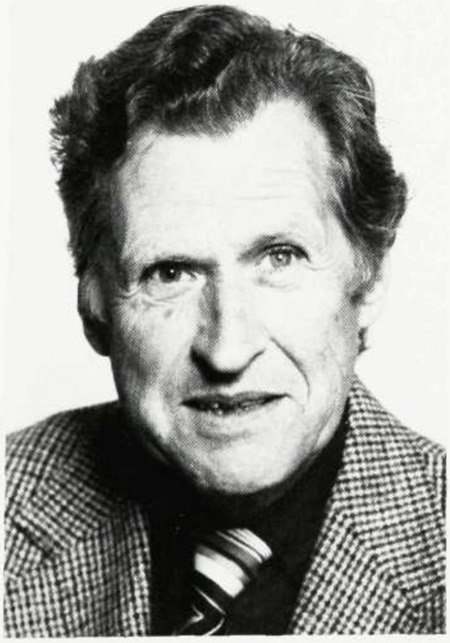
COVER
Installation testing of an AXE 10 exchange in Saudi Arabia

Contents



Folke Berg retires from Ericsson Review

Folke Berg was first employed by Ericsson in June 1946. In January 1973 he joined the editorial staff of Ericsson Review. With the aid or artists, photographers and translators he has produced the magazine, at first in five languages and later in four, being responsible for the whole process from the initial stage with Swedish manuscripts and illustration scetches up to the finished product. Ericsson Review presents the Ericsson products and is thus the company's face towards the outside world, and the demands made on it are high. With each issue of the magazine Folke Berg has succeded in his ambition to obtain a product that is typographically, pictorially and graphically perfect.

Now that Folke Berg retires from his post the company would like to express its gratitude for his excellent work and also to wish him all the best in the future.

Björn Svedberg *Gösta Lindberg*

The LM Ericsson Prize Winners 1982

This article is an excerpt from the speech given by Dr. Håkan Sterky, chairman of the LM Ericsson Prize Committee, at the awards ceremony of the 1982 LM Ericsson Prize, May 5, 1982. The prize, which consists of a gold medal and as of 1982 the sum of 200,000 Swedish Kronor, was first awarded at the LM Ericsson centennial in 1976, and according to the conditions governing the award of the 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".

By the autumn of 1981, 40 proposals involving 49 candidates working in 13 different fields of activity had been submitted to the Ericsson Prize Committee. The Committee consists of three members and is completely independent of the LM Ericsson's Board of Directors and President. It has sole responsibility for selecting the prize winner among the candidates nominated. 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 1982 Ericsson Prize is being shared by two men who have made fundamental contributions to the development of new forms of information transfer. After 60 years' professional career in telecommunications it is an especial honour and a great pleasure for me to present the winners of the 1982 Ericsson Prize:

Dr. Leonard Kleinrock, born in 1934, Professor and Head of the Computer Science Department Research Laboratory at the University of California (UCLA), Los Angeles, USA and

Dr. Lawrence Roberts, born in 1937, Vice President of Research and Development for GTE Customer Premises Equipment.

The proposals for the Prize covered the greater part of all the fields which today show development within telecommunications and computer technology. After extensive interviews with several experts in various fields, the Committee selected these two candidates which have worked for many years in a very expansive and exciting field of technology, that of communication with and between computers. They have spent much time, energy and imagination on that part of information transfer which has been given the name packet switching. After a decade of development, this method proved at the end of the seventies to be an exceptionally viable solution to the problem of how to transfer information within different telecommunication networks. Through his scientific work, Dr. Kleinrock has created a school for the theoretical treatment of traffic problems within the computer and data network domains. Dr. Roberts as an innovator and designer of packet switching systems has proved the expediency and competitiveness of this technology. Both have contributed brilliantly in speech and writing to making this field of technology known, understood and accepted.

In their presentations, the prize winners will describe in more detail how they began their work in the field of computer technology. They will describe the difficulties they met with, in their efforts to synthesize the advantages that computer technology and telecommunications engineering offer, and the developments they foresee in the future. For my own part I should like to describe, as simply as possible, the background to their activities within computer technology.

Information transfer in the telecommunication network

Today the information that is transferred in a telecommunication network may be divided into two groups. In one of the groups, comprising telephone conversations, no delay in transfer can be tolerated. Up till now this problem has always been solved by allocating a specific communication path in the telephone network to the two persons telephoning. This is called *circuit switching*.

Such stringent time constraints are not imposed on the other group, comprising telegrams, telex, and so on. These messages may be transmitted link by link through a network and a whole communication path does not need to be reserved. Of course, in this case it is required that the information be storable at each intermediate point while waiting for the next free link. This is called *message switching*. A variant of this method especially suitable for data transfer is *packet switching*. In this case the information is divided up into small elements or packets, which are transmitted within the telecommunication network in practically the same way as parcel post is handled but at astonishingly great speed. Even if, in the future, the postal service were to reduce the delivery time to hours for an express letter within a residential area, telecommunication companies with the aid of computer technology could achieve the equivalent objective in a hundredth of a second, i.e. one million times faster.

The principle of packet switching

To help the layman grasp the significance of the principle of packet switching, I will briefly describe what happens when a parcel is transferred by the postal service from sender A to receiver B. The sender A takes the parcel to a Post Office in his home town. The Post Office decides which path the parcel is to take. First it is sent, together with many other parcels, and by relatively slow means of transport, to a central sorting station. There the parcel, together with many others, is packed in a postbag or container which in its turn is loaded into a lorry, railway truck or aircraft. All this is in order to speed up, at a reasonable cost, delivery to the town, in which receiver B is located. After more sorting, the parcel is sent to the Post Office in receiver B's residential area. From there it may be delivered by a postman. Note that where long distances are involved, the parcel is never sent directly from A to B in the care of a postman on a bicycle or in a car.

Starting from this description of the procedure for transfer of a physical object, I will now deal with the procedure when a data message is to be transferred on an electronic path between terminals A and B. My message, as well as those of others, is first assembled in a host computer. There are sorting stations in the data-switching telecommunication network just as in the postal service. These are the nodes of the network. There are also efficient common means of transport, i.e. coaxial cables, radio links or satellite circuits. In the nodes there are computers which guide the messages through the network. My message, which is coded in digital form goes from the host computer to a node computer. This latter is programmed to divide the message up into packets of standard length, furnish each packet with a number, the address to terminal B, information for error correction, etc. The packet is also stored in the nodal computer's memory and there, together with other packets, waits for suitable paths to become free. In this way my message is forwarded from node to node. The computer in the last node on the path to terminal B compiles the various packets, so that the message corresponds to the original. Often the packets have arrived at the address node on different paths and in the wrong order.

Economic aspects

As always when decisions are made about when, where and how new technology is to be introduced in practice, the economic aspects are of great importance. This is also the case when it is to be decided whether circuit switching or packet switching is to be used for data transfer. Development within electronics has played an important part here. The components which are now used within both telecommunication technology and computer technology are today about ten times more reasonable in price than six years ago. Space requirements have also been dramatically reduced, and the functional speed of components has increased considerably. Semiconductor circuits with more than 100,000 components on surface

Fig. 1
Dr. Leonard Kleinrock and Dr. Lawrence Roberts have just received their prizes from His Majesty, Carl XVI Gustaf, King of Sweden. To the right Dr. Håkan Sterky, chairman of the Ericsson Prize Committee

dimensions no bigger than a little fingernail can already be manufactured.

The development in electronics has therefore meant considerable reductions in costs for transfer of information in the telecommunication networks. But the effect has been even greater as regards hardware in computer technology. Dr. Roberts has given in one sentence a guideline for decision-makers. He writes, "If lines are cheap, use circuit switching and if computing is cheap, use packet switching".

The first data networks

During the years 1962–65 the first packet switching systems were proposed by Paul Barran and Lawrence Roberts in the United States and by Donald Davies in the United Kingdom. In 1967 the construction of a network, called ARPANET, was commenced in the USA and some nodes in this network were put into service in 1969. More and more host computers have successively been connected since. In 1977 there were 111 of them.

It is apparent from available statistics that there was a large number of data networks based on the packet switching principle in general use in 1981. Naturally, the number of networks is greatest in the United States. There are five, and some of them are exceedingly large. The method has been introduced in Canada, Europe and Japan and the number outside of the USA is around twenty. In addition there is a considerable number of private networks using packet switching.

The Nordic Data Network which was put into service during the years 1980 and 1981, is based on circuit switching but has been prepared for packet switching according to the Swedish Telecommunications Administration.

The Prize winners' contributions

In connection with the development of package switching, research within many scientific fields has been required. Research into telecommunications traffic and statistical analysis, probability calculations, queueing theory and programming methods, transmission engineering and methods for digital transfer of information, form the basis for successful practical results. It is in several of these fields that Leonard Kleinrock has made outstanding scientific contributions.

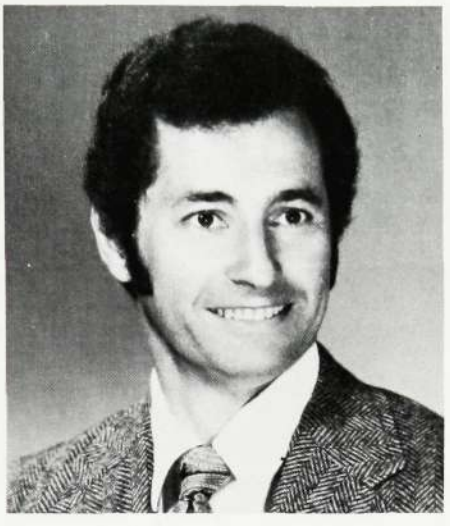
But the sensational results of packet switching have not been achieved by this alone. Repeated trials with limited data networks have been necessary. Failures have been discovered, programming of the controlling computers has been improved, less satisfactory solutions have been rejected, and new methods tested. Reliability, speed and economic viability have been the main criteria for the technical solution that is now generally accepted. It is in this practical field that Lawrence Roberts has made valuable contributions.

Dr. Kleinrock and Dr. Roberts have through their theoretical research work and technical development of packet switching opened promising new areas for transfer of data in telecommunication networks. More and more telecommunication and computer companies in many countries are convinced that this method will enable them to produce better and less expensive systems for data transfer. There is already talk of the possibility of using packet switching technique for transfer of telephone calls.

Fig. 2
Mrs. Roberts, the Prize Winners and Dr. Håkan Sterky after the Prize Ceremony

Packet Switching Principles

Leonard Kleinrock



LEONARD KLEINROCK
Dr. Leonard Kleinrock was born June 13, 1934 in New York City. Dr. Kleinrock received his B.S. from City College of New York in 1957 and the Masters and Ph.D. in Electrical Engineering from MIT in 1959 and 1963 respectively. While at MIT he worked at the Research Laboratory for Electronics and also with the computer research group of Lincoln Laboratory in advanced technology.

He joined the UCLA faculty in 1963. His research interests focus on the modeling, analysis and measurement of finite-capacity resources subject to unpredictable demands. The applications of this research involve multi-access computer systems in isolation as well as in computer networks. Dr. Kleinrock is now the head of the UCLA Computer Science Department Research Laboratory and is a well known lecturer in the computer industry.

He has published over 120 papers and is the author of three books, Communications Nets: Stochastic Message Flow and Delay, 1964; Queueing Systems, Volume I: Theory, 1975 and Queueing Systems, Volume II: Computer Applications, 1976. He is principal investigator of the ARPA Advanced Teleprocessing Systems contract at UCLA.

He was recently elected to the National Academy of Engineering , is a Guggenheim Fellow, an IEEE Fellow and serves on the Boards of Governors of various advisory councils in the computer field. He has been appointed to the Science Advisory Committee for IBM. He has received numerous best paper and teaching awards, including the ICC'78 Prize Winning Paper Award, the 1976 Lanchester Prize for outstanding work in Operations Research and the Communications Society 1975 Leonard G. Abraham Prize Paper Award.

If you attended any of the large computer conferences of the mid-1960s, you surely heard at least one panel heatedly debating the central issue of data communications. On the one side, we heard the representatives from the data processing industry complaining bitterly that no suitable facilities existed for efficient communication of computer-generated data. On the other side, we heard the representatives from the communications industry claiming that each country was essentially a copper mine of interlaced telephone channels which could be used for data communication. However, this telephone network proved to be a completely unsatisfactory solution for computer-communications since the typical use from a computer terminal was the need to send less than one second's worth of data, and the telephone plant required tens of seconds to set up the connection and (in the United States, for example) there was a minimum duration charge for three minutes. This impasse led to a revolutionary new method for using communication channels which has come to be known as packet switching.

Before describing the principles behind packet switching, it is important to elaborate on the nature of computer-generated data. Information processing devices (especially computer terminals) tend to generate data in widely separated bursts. Indeed, computer terminals almost never warn you ahead of time when they need to send data down a communication channel, they seldom tell you how much data they wish to send, most of the time they send nothing at all, but when they do occasionally want the channel, they want it immediately. This nasty combination produced most of the problems we faced in data communications.

In the mid-1960s, the classical technique for assigning voice channels to telephone conversations was circuit switching. With this method, a connected path of channels would be set up for the duration of a "call" in a dedicated fashion. When this technique was applied to data communication, these dedicated communication channels were idle most of the time waiting for occasional bursts of data.

Clearly, what was needed was some method of assigning the communication channels to the terminals only during those few instances when required; this would alleviate the enormous inefficiencies of the classical techniques in this new environment. The solution to the problem required that intelligence be installed in the switches which assigned the channels. The cost of this computerized intelligence had been falling dramatically due to the unbelievable progress in microelectronics – a trend which continues to this day. By 1970, the savings in communications due to rapid dynamic channel assignment by these intelligent switches exceeded the cost of the switches themselves, and so the economic forces made a technology such as packet switching unavoidable. In fact, by 1969, a fledgling packet-switching network (the U.S. Defense Department's ARPANET) was already operational and expanding.

Packet switching works as follows. Imagine that a message, such as the text of the previous sentence, must be sent from a Source to some remote Destination through a network as shown in fig. 1 using the technology of packet switching. The text of that sentence consists of 34 characters (including spaces). Now in packet switching, the text is broken up into segments called *packets*, and these packets are transmitted through the network separately, passing from one node to another node, hop by hop. Each packet has a maximum allowable length which, for our example, we will assume to be 32 characters. Thus, for our sentence, we have a message consisting of two packets, one of length 32 characters and another of length 2 characters. In addition to the data characters, each packet carries with it a coded version of

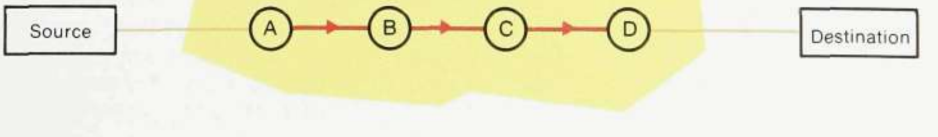


Fig. 1
Data communication network for packet switching from source to destination via nodes A–D

Fig. 2
Dr. Leonard Kleinrock on the rostrum

the address of the Destination. When the first packet is presented to node A, the intelligent switch (in that node) will select the first step in that packet's journey (obviously, it will select node B for this simple example). If the selected channel connecting these two nodes is busy transmitting another packet, then our first packet will wait in a queue until its turn for transmission comes up, at which time transmission begins. When it is received at node B, the process is repeated, and we note that the packet "hops" from node to node through the network using only one channel at a time, possibly queueing at busy channels on its journey. Each packet is treated in this fashion. Thus, many packets of the same message may be in transmission simultaneously, thereby allowing packets to be "pipelined" down the chain.

This procedure differs significantly from circuit switching. In particular, we no longer dedicate any channels ahead of time; they are used only on demand in a dynamic fashion and need be acquired only one at a time instead of in groups. It is this feature of packet switching which provides the efficiency of channel use. The decomposition into packets provides a reduction in transmission time for the entire message through the network due to pipelining. Furthermore, the intelligent switches are constantly sensing conditions to select good paths through the network; this provides considerable improvement in reliability since alternate paths will be used in the case of failure or congestion along primary paths. Since the Source and Destination are not directly connected (as was the case with circuit switching), it is possible for devices of vastly different speeds to communicate with each other through the network; that is, a packet switching network provides the capability of speed conversion. Thus, with packet switching, we have an efficient, rapid, reliable, and flexible communication system.

With any system as complex as a large-scale packet switching network, it is necessary to develop methods for performance evaluation and for system design. If one creates a mathematical model of packet networks, one finds that an exact analysis of the system behavior is hopeless. Our current tools are far too crude for the purpose. In spite of this, we had to find some method of analysis. To our great good fortune, it is possible to introduce certain assumptions which simplify the problem to manageable proportions and which yield excellent tools for approximate system performance evaluation. The same situation exists with regard to network design; we have no adequate methods for leastcost design, but we have developed effective techniques for lowcost design.

It is worthwhile to point out that much of the effectiveness of packet switching comes from the "largeness" of the networks in which we use the technique. Specifically, we have found that two resource-sharing principles come into action here. Recall that the "bursty" behavior of the terminals gave rise to many of our communication problems. There exists a principle, known as the "law of large numbers", which basically states that a large number of bursty data sources will collectively behave in a very "smooth" fashion whereby a predictable and steady flow of data will emanate from the group as a whole. (The insurance companies know this – though they know almost exactly how many people will die next year, they simply don't know which particular individuals will die, so they "bet" with everybody, and they usually win the bet; that is, the mortality tables are extremely accurate due to the law of large numbers.) The second principle is often referred to as the "economy of scale" principle. This principle states that if we begin with a service system which handles a certain load and which is endowed with a certain capacity for handling that load, then doubling the load and doubling the capacity will cause the response time of that system to improve by a factor of two. In order to see how these principles apply to our networks, let us examine the components of a typical system. In fig. 3 we show a more complete picture of such a network. At its periphery we note the various kinds of terminals and computer facilities. It is the purpose of the "communications sub-network" to provide communications among these various devices, and in this sub-network we see the intelligent switching computers which are connected with high speed communication lines. Observe that this sub-network carries traffic from many terminals and computers (hence we expect our first resource-sharing principle, the law of large numbers, to work for us). Since the network is handling so much traffic from this large number of devices, we require large-capacity channels, and so we expect to reap the benefits from the economy of scale (our second resource-sharing principle). In fact, these two resource-sharing principles apply not only to the expensive communication channel capacity, but they also apply to the two other key network resources, namely, the storage capacity and the processing capacity of the intelligent network computers themselves. In summary, then, it is precisely when large populations dynamically share large capacity resources that we enjoy significant performance efficiencies; packet switching networks are prime examples of such systems.

However, we have an unresolved problem. In fig. 3 we note that "remote" terminals must pass through a small network of their own before they reach the communication sub-network in which all the efficiencies of large shared systems are to be found. If we concentrate on the channel labelled "Low-speed line" in that figure, we can inquire regarding the efficiency of its use. We see that only ONE terminal can "share" this line (and one is NOT a large number); similarly, since only one terminal's traffic is passing over this channel, then the load is small and so also will be the capacity. We therefore conclude that neither of our resource-sharing principles applies to this "local access network". Things are fine deep in the sub-network, but we are in trouble at the periphery of our system. Unfortunately, a significant fraction of the system cost is invested in these local access networks. What shall we do in this case? The consideration of this question ushered in the technology of local access, has recently become the major focus of computer-communications. In fact, in the form of local area networks, it has become the critical element in the realization of the automated office.

In order to resolve the problem of local access, we once again take a cue from our two resource-sharing principles. These principles tell us that we must provide a large capacity communication resource to be shared by a large number of users in some fashion. One way to create a common channel to be used by a number of geographically distributed terminals is to provide them with a broadcast radio channel. It is clear that we have moved in the correct direction – we now have a large number of terminals collectively generating

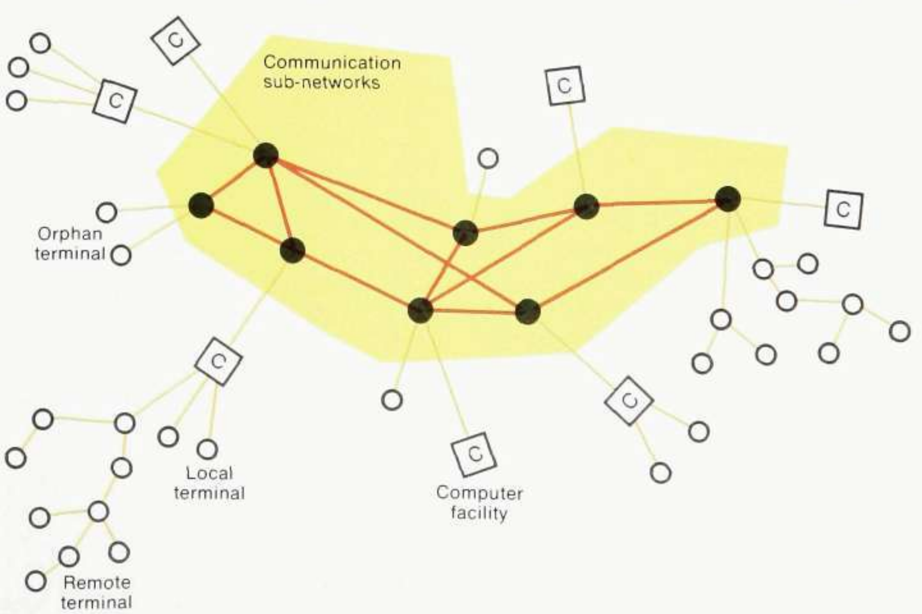


Fig. 3
The structure of a computer communication network

Low speed line
High speed line
Switching computer
C **Concentrator**

a large amount of traffic to be carried over a large capacity channel. The characteristic of this channel is that more than one terminal may attempt to transmit their packets over the common channel at the same time; therefore we refer to this as a "multiaccess" channel. Whenever a transmission is made, all (or most) of the terminals hear the transmission, and so we refer to it as a "broadcast" channel. Lastly since the terminals are distributed in space (specifically, they are not automatically aware of each others' needs to transmit), we say that the channel is "distributed". *Such a channel may therefore be described as a multiaccess broadcast distributed channel.* In the recent past, considerable effort has gone into the design of such channels in order to find ways to use them efficiently. A number of exciting access methods based on packet switching concepts have been developed and experimented with. Of more interest, perhaps, is the application of these multiaccess techniques not to radio but to in-building communication systems commonly known as *local area networks.*

Local area networks are intended to provide high-speed, low-cost communications within a building or between buildings. The typical application is for office automation, namely, to connect together the many information processing devices that one finds in the modern office; these include terminals, work stations, microcomputers, minicomputers, printers, storage devices, electronic blackboards, displays, etc. We have seen a veritable explosion in the number of implementations and products being offered to satisfy the needs of in-building communications. In most of these systems, we see the basic principles of packet switching applied once again, namely, dynamic access from many bursty devices to a common wideband channel. Among the access schemes, two have attracted considerable interest. These are: Carrier Sense Multiple Access with Collision Detection (CSMA/CD) and Token Passing.

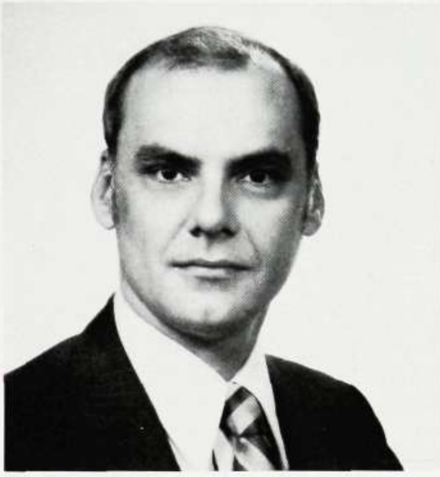
CSMA/CD is a method for using a multiaccess channel in which a device listens to the channel before transmitting and will refrain from transmitting if any other transmission is heard; furthermore, the device listens to the channel while transmitting and if a collision with another device's transmission is detected, both devices will cease transmission and attempt a repeat at a later time. This method is used in the well-known ETHERNET system.

Token passing is a method whereby a special signal known as a token is passed from device to device down the channel. Whenever the token is passed to a given device, that device is permitted to transmit its data on the channel, after which the token is passed to the next device. These are but two of many possible access schemes for sharing the capacity of the channel. Aside from the access method, there is the choice of medium for the communication itself. One common medium is twisted pairs of copper wire; these have been used for the traditional Private Automatic Branch Exchange (PABX) for handling voice switching and have recently been upgraded into Computerized Branch Exchanges (CBX), which are capable of efficiently handling data as well as voice. Coaxial cable is an attractive medium as well, both in its baseband and broadband implementations. Beyond coaxial cable, the remarkable technology of optical fiber channels is extremely exciting; with this medium, enormous bandwidths are available in a medium which is small, lightweight, flexible, low loss, immune to electromagnetic interference, immune to high temperature, etc. The major impediment to the widespread use of optical fiber at present is the lack of a cost-effective method for connecting and tapping the cable; however, when the economics become competitive, optical fibers will find an enormous use in local area networks and in point-to-point long haul communications. These revolutionary developments in local communications all take advantage of the packet switching technology in that they dynamically assign capacity among a large number of bursty devices.

The application of these revolutionary developments in communications does not stop with pure data transmission. The requirement to send video, facsimile, voice, graphics, etc., as well as data certainly exists in today's automated environment. The technology for these applications is currently coming into place. We have already seen a large number of packet-switching networks spring up around the world. Many local area networks are already in place. Satellite data networks are in operation. Many of these separate networks have already been interconnected, and more are being attached every week. We are entering an era of world-wide access to data and to information processing resources, this access being made available through the sophisticated and cost-effective packet-switching computer networks we have been discussing. The 1970s was the era of computer network development. The 1980s will be the decade of network applications and will provide the penetration of the information revolution into many additional areas of activity.

Packet Switching Economics

Lawrence G. Roberts



LAWRENCE G. ROBERTS

Dr. Lawrence G. Roberts was borne December 21, 1937 in Norwalk, Connecticut, received a bachelor's degree in electrical engineering from Massachusetts Institute of Technology in 1959, and a master's and doctor's degree in electrical engineering from the same institution in 1960 and 1963 respectively.

Dr. Roberts is Vice President of Research and Development for GTE Customer Premises Equipment, a division of GTE Communications Products group. Formed in January 1981, this new organization will have responsibility for development of integrated voice/data switching systems and related terminal equipment.

Previously, Dr. Roberts served as President of GTE Telenet Communications Corporation, a company that pioneered in the development of packet switched data communications networks.

He joined Telenet in 1973 and directed the design and implementation of Telenet's "third generation" network, which utilizes advanced multi-microprocessor technology. During this time he also guided the company's rapid growth from a small, privately held business to its 1979 merger with GTE.

Deeply involved in the development of packet switching since the early 1960s, Dr. Roberts is considered to be the architect of packet switching technology.

Formerly the Director of Information Processing Techniques at the Advanced Research Projects Agency (ARPA) of the Department of Defense, Dr. Roberts was responsible for the initiation, planning and development of the ARPANET, the world's first major packet network. In addition, he was responsible for a broad program of computer security, speech compression and understanding, satellite communications, and computer system design.

Upon leaving ARPA in 1973, Dr. Roberts was awarded the Secretary of Defense Meritorious Service Medal in the field of computer science and communications, and in 1976 the Harry Goode Memorial Award in the information processing field. IEEE awarded Dr. Roberts with its Computer Pioneer Award in 1982. As a charter recipient, he was acknowledged as a major contributor to the concepts and development of the Computer Field. Dr. Roberts' award commemorated his outstanding achievements in packet switching. Also in 1982, Dr. Roberts received the Interface Conference Award for extraordinary achievement in data communications.

Dr. Roberts has published widely in professional journals and has lectured throughout the United States and abroad. He is a member of the association of Computing Machinery; The Institute of Electrical and Electronics Engineers, Inc.; Sigma Xi; and the National Academy of Engineering.

Early in 1969 an important event occurred and went by totally unnoticed by the world; packet switched data communication became inherently less expensive than circuit switched data communication. This was a permanent and irreversible change brought about not by invention or development in packet switching but by the immutable action of economic trends. Packet switching optimizes the use of communications bandwidth through the use of considerable computation. With the extremely rapid decrease in the cost of computation brought about by the semiconductor revolution, this computation had become less expensive than the alternative of wasting communications facilities.

This basic change in the economic technology to utilize for data communications went unnoticed however because research on packet switching was just starting—in the US with the initial work on the ARPA Network and in the UK with research at NPL. Over the next four years the basic techniques of packet switching were developed and tested in experimental networks like the ARPA Network[3]. Packet switching proved so successful both technically and economically that public data communications networks were organized on the new technology starting with Telenet Communications in the US in 1975, followed quickly by Datapac in Canada and Transpac in France. From that point forward virtually all data networks, both public and private have been based on packet switching and its cousin statistical concentration rather than circuit switching and non-statistical multiplexing. Today the public packet switching industry has revenues of several hundred million dollars and private equipment sales are of similar size and both are growing vigorously.

Economics of Packet Switched Data

Outside of enthusiasts claiming benefits in various cases, none examined the underlying economic trends of packet vs circuit switching until 1974. In a study I published that year, I examined the trends of both data processing and communications tariffs and concluded that computer electronics had been becoming cheaper much faster than transmission capacity. This resulted in a very sharp decline in the composite cost of packet switched service up until 1969, when the communications component became dominent. The statistical concentration inherent in packet switching provides reductions in the transmission bandwidth used of between 4:1 to 100:1 for various types of data and 7.5:1 for normal 1,200 bit/s interactive calls. Thus, unless the switches in a packet net add a very large cost (as was true before 1969) to the system cost, packet switched data communication is substantially less expensive than circuit switched service.

The trend produced for computation in the 1974 analysis has held very true when updated with the latest switch designs. However, history has shown that the average packet size sent in large packet networks is 32 characters rather than the 128 characters assumed in the prior analysis. This reduces the data moved per packet processed or alternatively, increases the packet processing load for a given data throughput. This factor moved the date forward to 1973, when the cost of computation really equaled the cost of communications.

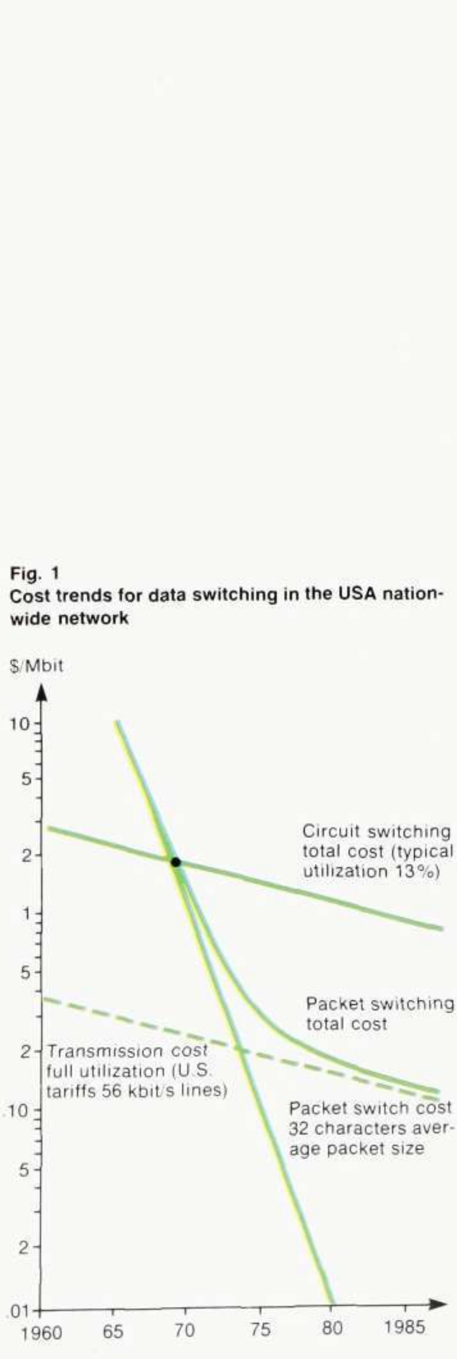
Also, the trend for communications cost was found to be too optimistic in hindsight. It was based on AT&T tariffs for 56 kbit/s speed private lines from 1963 to 1974. They trended down with the introduction of AT&T's Digital Data Service. However, ever since 1974, private line tariffs have increased as has the Digital Service until today it is over twice the 1974 price. Thus, a revised communications trend has been derived which decreases 7%/year rather than 10% as before.

For the data communications trends, a

network is assumed which has two end-office switches and three tandem-office switches. Thus, costs from five switches are included in the switching cost. Packet switches produced from 1970 on have been used to determine the exact cost per million bits of data moved end-to-end through the network. To conform to actual experience, 32 byte packets are assumed. Monthly costs for equipment are assumed to be .04 times the price. Four communication lines are required to link five switches, each one 300 miles long resulting in a path 1,200 miles long. This conforms well to actual network experience. For both lines and equipment, full usage is assumed for each working day or 173 hours per month.

Based on the model network as outlined above, fig. 1 shows the trends of packet switch costs decreasing by 37% per year or a factor of ten every five years. Transmission costs for circuit switching are 7.5 times those of a packet switched net assuming 1,200 bit/s switching and multiplexing and a typical to high actual utilization rate of 160 bit/s. Circuit switch costs are computed using 56 kbit/s tandem trunks and added to the transmission to create a total cost for circuit switching. Total packet switching costs are the sum of transmission and switching.

Fig. 1
Cost trends for data switching in the USA nationwide network



The result as shown is that packet switching crossed over with circuit switching in 1969 and quickly became 7 times less expensive. It is not surprising in light of this large differential that packet switch data networks and services have experienced such large growth since 1975.

Packet Voice

With the large success of packet switching for data applications the logical question is—would it work for voice and if so, what value would it have. The technical application of packet switching to voice has often been questioned for many spurious reasons, largely a result of impressions formed from experience with the original low speed packet networks. Early packet switches were slow and used low speed lines, thus they introduced considerable delay (200 ms) and delay variance, far too much for voice. In fact, except for highly compressed voice, most packet switches being sold today are clearly too slow and expensive for voice traffic. However, these are switches designed in the mid 70s and the inexorable advance of electronics speeds is quickly changing the situation.

The design of packet switches has continuously changed since the first systems. Initial switches had a single processor per switch serving 30–60 lines. Each task for each line had to queue a request for the central processor often resulting in long delays for each packet moving through the switch. Later designs used a microprocessor for each eight lines, reducing the processing delay from tens of milliseconds to a few milliseconds. Modern switches being designed use a single microprocessor per port and specialized LSI hardware switching to move packets between ports. This reduces the delay of a packet moving through the switch to a few hundred microseconds. Also, through the use of internal bus speeds far in excess of possible loads the delay variance can be reduced to tens of microseconds.

In addition to reducing switch delay to well under a millisecond, these design changes permit throughput rates to increase by several orders of magnitude into the several hundred Mbit/s range, substantially exceeding the total full port capacity of a circuit switch. Thus, this new generation of packet switch, permitted with todays semiconductor technology, will be quite capable of packetizing voice and switching it with no throughput restraints and perhaps most importantly, with negligible delay.

Silence Detection

Voice conversations consist of about 67% silence and thus a factor of three savings are potentially in transmission through statistical concentration. This is less than the 7.5 concentration available in data but still quite a saving when transmission costs considerably more than switching in a voice network. Silence detection has been used for many years for analog voice on submarine cables but analog systems tend to clip off the start of words and cause percep-

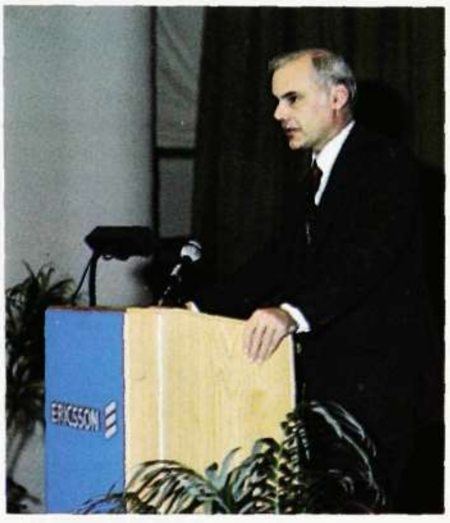


Fig. 2
Dr. Lawrence Roberts on the rostrum

tible degradation of the speech quality. Digital silence detection algorithms however have progressed to where a few dollar microprocessor can find the silence periods extremely accurately with only a few milliseconds delay and even packetize the voice at the same time. As a result silence detected voice is likely to be introduced extremely rapidly into todays voice networks due to the substantial transmission savings possible.

Once voice is silence detected it consists of labeled blocks or strings of data which by definition is packet switching. Two basic alternatives exist for packaging it for transmission; sending a label and then a string of voice samples interspersed with other strings, or to pack a segment of speech from one call into a labeled block. The first technique is typically called Digital Speech Interpolation (DSI) and although it adds very little delay it requires that all the streams be the same data rate, the same restriction which hurts circuit switching. Also, since DSI boxes are used at either end of each channel the cost of DSI increases rapidly as speech travels through a tandem network with many channels. The other alternative is the normal technique of packet switching and creates a delay proportional to the length of the packet. However, using normal packets permits the efficient mixing of data and voice of any speed. Further, it permits a simple standard packet switching technique to be used for tandem switching, considerably reducing the cost compared to the DSI approach.

Packet Size

The size of the voice segment included in a packet must be long enough to keep the header (label) overhead down. The header need only be three bytes long, and packet sizes from 8-64 bytes are being considered by various groups. This represents speech periods of 1-8 ms, far shorter periods than humans would perceive directly but at the longer delays sufficient to require the use of an echo canceller to prevent noticeable echoes from analog hybrids in the current voice net. For nationwide long haul networks, echo cancellors are required in any case due to the inherent propagation delay. Therefore, in this case it makes sense to select one of the longer packet sizes like 32 bytes (4ms) where the overhead is only 10%.

Circuit vs Packet Switching

In order to cost trends of circuit and packet switched voice networks, it is first necessary to hypothize the model networks being considered. Much the same as for the data switching case the model network has 5 switches and 4 tandem trunks. The two end office switches are costed with individual analog voice ports. The tandem trunk lines should be as wideband as traffic would permit for economy. The speed used in the model is T2 or 6.2 Mbit/s. Circuit switches would use a T2 trunk to carry 96 standard 56 kbit/s voice calls. Packet switches using 32 byte packets would be able to put 217 silence detected 64 kbit/s voice calls on the same trunk, an improvement of 126%. The transmission cost for the four transmission links would therefore be 44% of the circuit switched case.

Since the unit of a voice call is a convenient basis to start from, it is attractive to use as a unit of measure for all costs, the capital investment cost of the network per voice call of capacity. Transmission costs have been obtained by using the historically reported AT&T Longlines costs of installed transmission plant per circuit mile and assume a standard call length of 1,000 route miles. Using these well reported numbers this results in a cost decrease trend of 5.2% per year. This trend is shown as a dotted line in fig. 3 and the corresponding packet switched transmission cost trend at 44% of the circuit trend.

The cost trend for circuit switches designed between 1970 and 1982 shows a similarly low decrease per year. This is largely because the technology for digital circuit switches has stabilized and is not computation intensive. This cost can then be added to the transmission trend for an overall circuit switched network cost trend.

The cost of packet switches in the 1970s was determined from the packet throughput and cost of the early packet switches used for data. Voice would

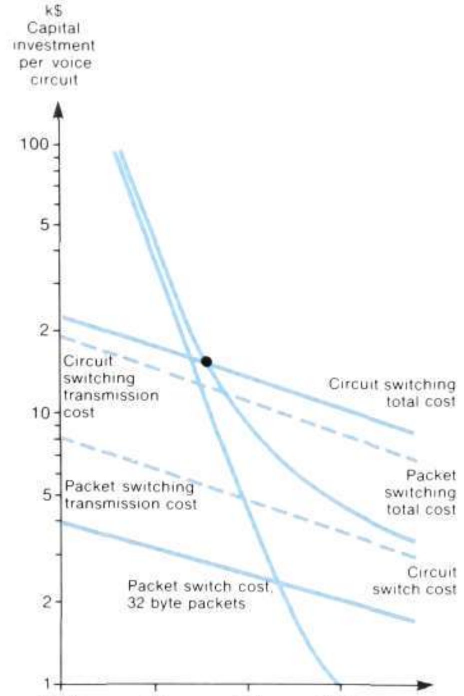


Fig. 3
Cost trend per voice circuit in the USA nationwide network

have overloaded these switches but the cost trend can still be determined. Modern packet switches now being designed or built form the basis of the other end of the packet switch cost trend. The individual voice ports on the end offices remain in the late 1980s as the dominant cost causing a flattening out of the cost curve. However, due to the extremely high efficiency of high speed tandem switching using packet technology, the combined cost of the five packet switches actually falls under the cost of the circuit switches after 1981. This efficiency arises due to the ability to switch 32 bytes with only one decision and memory lookup (label lookup) rather than having to make a decision and memory lookup for each byte in the high speed stream in a circuit switch. This factor also means that packet switches can actually be built to switch at 32 times the maximum speed of a circuit switch, a result which may become important in the building of fiber optic networks.

Adding the packet switch cost trend to the packet transmission cost results in a trend curve for the total cost of a packet voice network. This is seen to cross the circuit network cost in 1977 and as with data, quickly becomes significantly less expensive. As with the data communications crossover in 1969, no one really noticed when it occurred and the commercial reaction was not until 6–7 years later. This would suggest that packet switched voice systems would start appearing in 1984 and become an important worldwide factor in the five years thereafter. The raw economy of the transmission savings plus the advantage of being able to mix voice and data will undoubtedly force the change to occur over time, however, due to the extremely large installed voice plant the change could take considerably more time than was required for the data switching transition. Thus perhaps a more likely prediction based on these trends would be that the transition to packetized voice would occur between 1985 and 1995. By then it might be worthwhile examining the trends for switched video.

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Training within the Saudi Arabia Telephone Expansion Project

Olle Näslund

TEP, the Saudi Arabia Telephone Expansion Project, is coordinated by the Philips–LM Ericsson Joint Venture Consortium. It is extensive and has necessitated a great amount of training. Thousands of people have been trained in several systems and branches of employment, and the training has been conducted in several languages. The instructors had to be able to adjust the training according to the different backgrounds and basic knowledge of the students. Able and experienced instructors and the excellent collaboration with the customer's operation organization, Saudi Telephone, have ensured a good training result.

The author describes the part of the training project that concerns Ericsson products, and gives some examples of how the training was carried out. A brief account is also given of how the training was adapted to suit local conditions.

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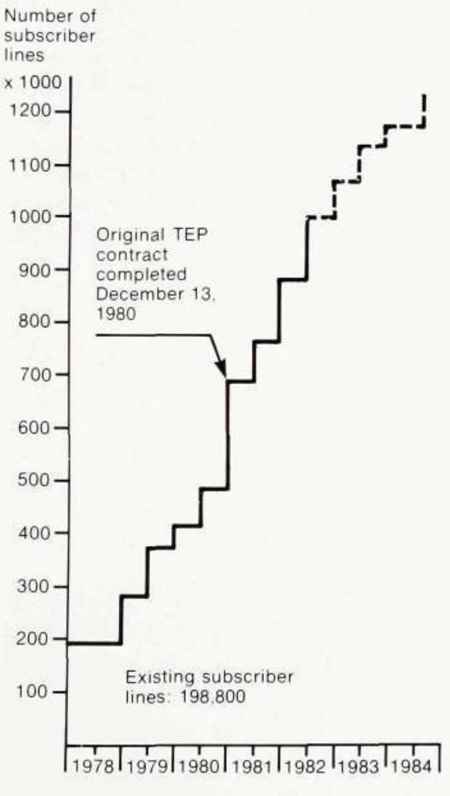


Fig. 1
TEP in figures per December 13, 1980
1,170,000 distribution points
3,500 distribution cabinets
7,850 km cable in
9,000 km duct pipes
21 AXE 10 exchanges, new
24 ARE exchanges, modernized

Saudi Arabia is now undergoing very rapid technical development. Large projects are being initiated one after another, and the Telephone Expansion Project, TEP, is one of the largest and technically most advanced projects to be carried out in the country.

Fig. 1 shows that about half a million telephone lines were installed during the period December 1978 to December 1980. The telephone exchange equipment is completely up to date, with computer-controlled equipment for centralized operation and maintenance. The amount of equipment is almost inconceivably large.

The cables and other equipment in the subscriber network have been adapted to the extreme environmental conditions that are encountered in Saudi Arabia.

About 20 new AXE 10 exchanges have been installed, and the existing local and transit exchanges of types ARF and ARM have been modernized and equipped with computer control to become ARE 11 and ARE 13.

One national and four regional operation and maintenance centres of type AOM 101 have been installed, as well as a repair centre for printed board assemblies and a software centre (for producing exchange software).

The building up of the subscriber network has entailed, for example, laying 8,000 km cable, connecting approximately 1,170,000 subscriber line pairs to distribution points, and jointing about 50 million wire pairs.

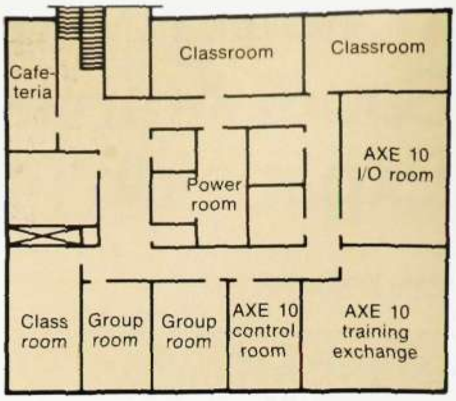
Figs. 2 and 3 show examples of the main activities in the project, namely

- installing and putting the telephone network into operation
- installing and putting the subscriber network into service.

Fig. 2
Installation testing of an AXE 10 exchange

Fig. 3
Installation of ducts for the subscriber network in Riyadh

Fig. 4
Premises for AXE 10 training in Riyadh



The installation and testing of the telephone exchange equipment and the equipment for the operation and maintenance centres was carried out by staff from Ericsson, whereas the work of building up the subscriber network was carried out by the Korean company Dong Ah, a subcontractor of Philips–LM Ericsson Joint Venture.

At one stage approximately 10,000 people were involved in the project, and most of them received some form of training during the course of the project.

The TEP contract also included training of Saudi Telephone staff in the operation and maintenance of the equipment, delivery of the training equipment and setting up the training centres in Riyadh and Jeddah.

Table 1 shows how many people were trained in Ericsson techniques in the different fields of activity included in the project.

Training of Saudi Telephone staff

TRAINING CENTRES ARE BUILT UP IN RIYADH AND JEDDAH
The training of Saudi Telephone staff took place mainly in Saudi Arabia. Only a few groups of students went to Stockholm, Sweden, for training during the time that the training premises were being prepared in Saudi Arabia. The contract included the delivery of training exchanges of types AXE 10, ARE 11 and ARE 13, which were installed in the Saudi Telephone training centre in Riyadh. All the equipment required for the classrooms and group rooms used for the training was also provided.

Fig. 4 shows the layout of the premises for AXE 10 training in Riyadh. Corresponding facilities were available for the ARE 11 and ARE 13 training.

Centres for training in the operation and maintenance of the subscriber network were set up in Riyadh and Jeddah. All furniture and all equipment for these, such as audiovisual equipment, tools, cable and training material, were provided by Ericsson.

Fig. 5 shows cable jointing training in the Jeddah training centre.

COMPLETE COURSE PACKAGES PROVIDED
The contract included the provision of well documented course packages, in order to make it easier for Saudi Telephone to take over the training.

Table 1
The number of people training in Ericsson methods within the framework of TEP

Field of activity	Organization	Number
Operation and maintenance of telephone exchange equipment	Saudi Telephone	354
Operation and maintenance of subscriber networks	Saudi Telephone	220
Installation of subscriber networks	Dong Ah	2000
Planning and quality control o subscriber networks	Philips–LM Ericsson Joint Venture	400
Installation and testing of AXE 10 exchanges in Mecca and Medina	Ericsson	30

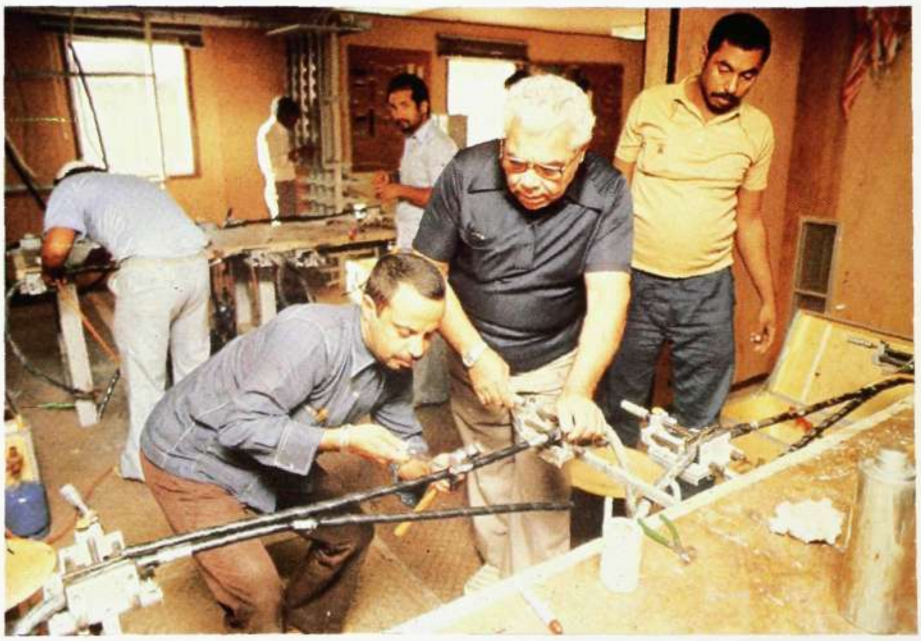


Fig. 5
Practicing cable jointing at the Jeddah training centre

Table 2
Training of Saudi Telephone personnel in the operation and maintenance of exchange equipment

Abbreviations in table:
- **TT** Equipment for automatic charging (toll ticketing)
- **ATME** Equipment for automatic transmission measurements
- **CSO** Central sales office (for connecting in subscribers)
- **SCC** Subscriber complaint centre
- **TMC** Transmission measurement centre
- **RC** Repair centre (for printed board assemblies)
- **SC** Software centre (for producing exchange data)

System/ Operating centre	Personnel category	Number of trainess	Weeks of theoretical training	Months of praktical training
AXE 10	Maintenance technicians	80	16	6–12
	System technicians	42	28	9–18
ARE 11	Maintenance technicians	60	16	6–12
	System technicians	24	28	9–18
ARE 13 incl. TT and ATME	System technicians	20	38	9–18
Exchange	Maintenance technicians	12	1	1– 3
power supply	System technicians	6	4	3– 6
CSO	Operating staff	6	1	1– 3
SCC	Operating staff	60	1	1– 3
TMC	Operating staff	33	1	1– 3
RC	Operating and maintenance technicians	11	17–24	6–12
SC	Operating staff	4	8	3– 6

Fig. 6
Operation and maintenance of the exchanges is carried out via the AOM 101 system

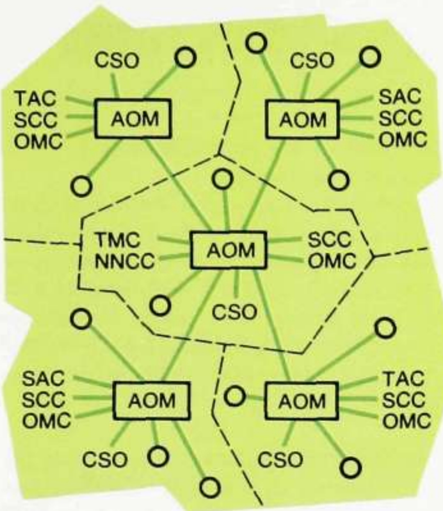
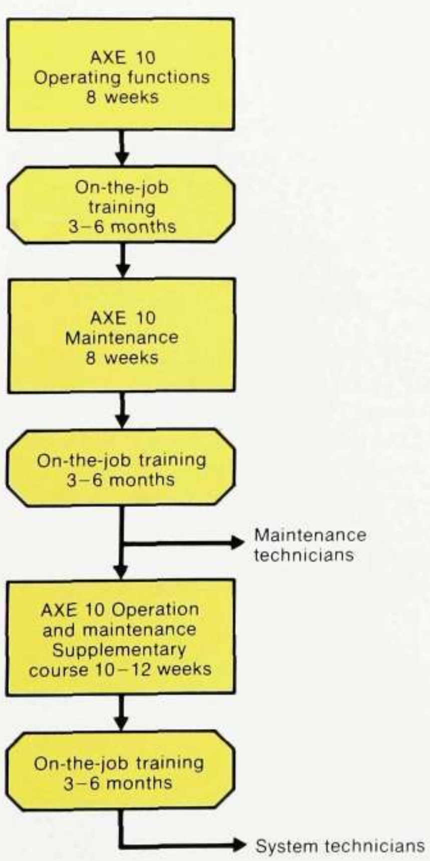


Fig. 7
Training paths for operation and maintenance staff for AXE 10 exchanges



Required basic knowledge: English, basic telephony, science of electricity and digital technology

The course packages supplied by Ericsson contained all necessary information for the instructors, such as course descriptions, syllabuses, lesson schedules, exercises and tests with solutions, and audiovisual material.

TRAINING IN THE OPERATION AND MAINTENANCE OF EXCHANGE EQUIPMENT

Centralized operation and maintenance

Operation and maintenance of exchange equipment is administered from operation and maintenance centres via the computer-controlled equipment AOM 101, fig. 6. The maintenance functions are carried out at OMC (Operation and Maintenance Centre), whereas the operating functions are carried out at operating centres, such as:
- Central Sales Office (CSO), for connecting and disconnecting subscribers
- Subscriber Complaint Centre (SCC), for locating faults on subscriber lines
- Traffic Administration Centre (TAC), for traffic recordings and statistics.

Personnel categories

Two types of personnel were trained for the operation and maintenance work at OMC, namely maintenance technicians and system technicians.

A maintenance technician must be able to carry out, with the aid of the operating manual, routine operation and maintenance work in an exchange.

A system technician must be able to handle the tasks of a maintenance technician and also to correct the types of faults that are not covered by the routines given in the operating manual. Personnel who are to work at operating centres of the types CSA, SSC and TAC are trained specifically for these jobs.

Training paths—operation and maintenance

Table 2 shows how many of Saudi Telephone's personnel have been trained in operation and maintenance and the length of the theoretical and practical periods. The length of the practice periods varies depending on the ability of the students to assimilate knowledge. The training plan for maintenance and system technicians for AXE 10 was designed in accordance with fig. 7. It started with an 8-week course on operating functions in AXE 10, such as connection of subscribers, traffic recordings and collection of statistical data.

The students were then given a break from the classroom and allowed to practice their knowledge in the field as assistant technicians. This was followed by another 8-week course on maintenance, and then another period of on-the-job training.

Students for the system technician training were then selected from the group of maintenance technicians. They completed a course of 10–12 weeks, followed by on-the-job training.

Training of staff for operating centres

It was found that the personnel at the operating centres of types Central Sales Office, Subscriber Complaint Office and Traffic Aministration Centre were not usually technicians and that they normally used Arabic as their working language.

It was therefore necessary to prepare special operation manuals in Arabic for this personnel category. The manuals had to cover the operation of terminals and the commands used in the different operating centres. The training staff of Ericsson and Saudi Telephone developed these work aids jointly by first analyzing the jobs at different operating centres, then preparing operating manuals in English and finally having them translated into Arabic.

The operating manuals were self-instructing, which made them very suitable for both training and use in the daily work.

On-the-job training

On-the-job training constitutes an important complement to the formal training in the classroom and at the training exchange.

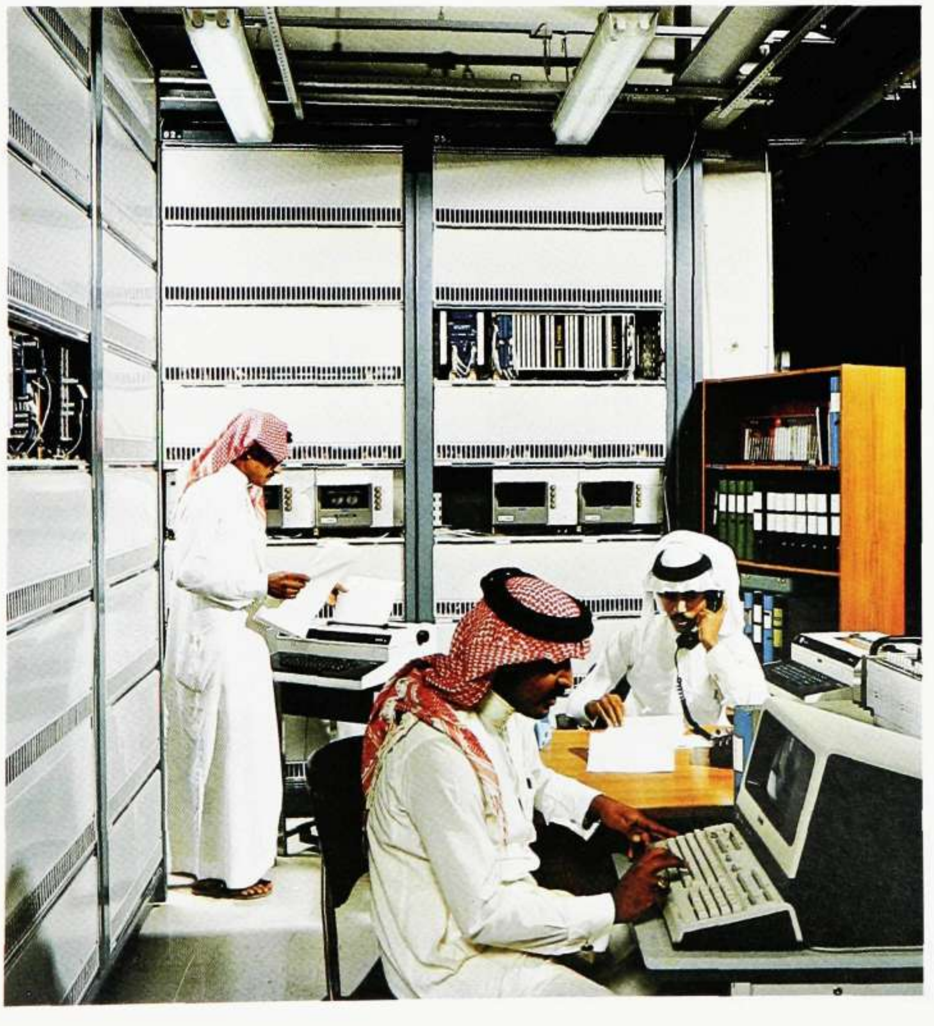
Course	Course length, weeks
Basic telephone engineering	1–4
Installing subscriber lines	5
Fault finding	6–7
Indoor installation	8
Pressurization methods	9–10
Line and network construction	11–14
Cable pulling	15

Staff category	Number	1–4	5	6–7	8	9–10	11–14	15
Ducted cable installers	65	×						×
Aerial cable installers	35	×	×		×		×	×
Cable jointers	90	×	×	×	×	×	×	×
Pressurization technicians	10	×				×		×
Total number of students at the training centres	200							

Fig. 9
Contracted training for Saudi Telephone concerning the operation and maintenance of subscriber networks

In spite of the emphasis placed on practical training (the training exchange was an exact replica of an ordinary exchange and great importace was laid on the use of the exchange documentation), the students were often lost when they were to start work in the operation and maintenance centre or an exchange. It was evident that for several tasks additional training was required with the aid of an instructor, and that the students needed help in getting to know their exchanges. Ericsson and Saudi Telephone jointly prepared a program for on-the-job training.

Fig. 8
Great emphasis was placed on practical work in the operation and maintenance training



The ability of the students to carry out their tasks was first tested. Deficiencies as regards skills and knowledge were mapped and the students were then given individual training programs in order to remedy the shortcomings.

Adaption of the operation and mainteance training

During the three years that training of maintenance and system technicians was carried out, a number of measures were taken to adapt the training to suit the background and needs of the students. Such adaptions were carried out in close collaboration with Saudi Telephone.

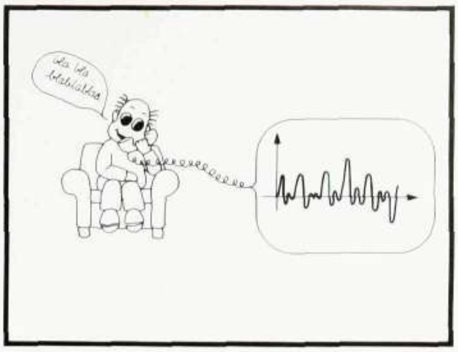
During the first courses the training results proved to be unsatisfactory because of gaps in the basic knowledge of the students. Saudi Telephone then started an education program comprising English, electronics and digital technology. This led to a rapid improvement in the training results.

The initial courses also showed that the theoretical parts ought to be kept as short as possible in order to allow more practice at the training exchanges. The courses were therefore modified so that the theoretical parts were divided into short sections, alternating with practical training. The effect of this modification was very positive and the great interest the students evinced in the practice sessions is illustrated in fig. 8.

Useful experience was also gained as regards training methods. For example, the instructors found that forming small groups for the purpose of problem solving did not work very well. This was because of the strong Saudi tradition that a whole group of students discuss and solve problems together, under the supervision of the instructor.

The interaction between instructors and students, and between the training staffs of Ericsson and Saudi Telephone, benifited greatly from the fact that most instructors worked in Saudi Arabia for as long as 2–3 years. The instructors then learned to adapt the training methods to the different backgrounds and level of basic knowledge of their students.

اشارة أكثر تعقيدا مثل اشارة الكلام والتي تحتوى على نغمات
مختلفة واتساعات وترددات مختلفة فى نفس الوقت.
ولذلك يبدو الرسم البياني مربكا .
لا يمكن تحديد النغمات المختلفة فى الرسم البياني .



الاشارات A ، B لها اتساع وذبذبة مختلفتين .
الاشارة C هى اشارة تحتوى على اشارتى A ، B
(فرق الجهد للاشارة A والاشارة B يضاف الى بعضه فى كل
لحظة) .
ولذلك الاشارة C تحتوى على نغمتين أو موجتين جيبيتين
باتساعين وترددين مختلفين .

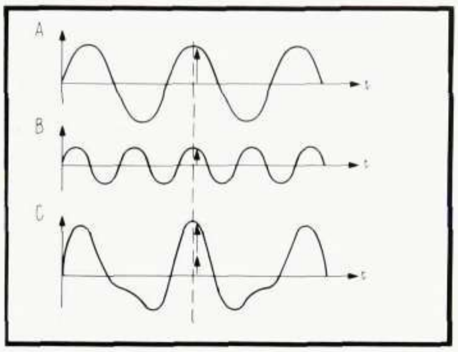


Fig. 10
Examples of student material in Arabic

Fig. 11
Training in the function of the telephone set

TRAINING IN OPERATION AND MAINTENANCE OF THE SUBSCRIBER NETWORK

Training in operation and maintenance of the subscriber network was carried out at the same time as the training activities described above. Fig. 9 shows the contracted training of 200 technicians at training centres set up in Riyadh and Jeddah. At each of these centres the premises were equipped for training in

- installing telephone sets
- jointing cables
- locating cable faults
- cable pressurization
- basic telephone engineering.

In addition personnel were trained in line and network construction and cable pulling. Special training areas for this purpose were built at each training centre.

Twenty Saudi engineers took part in a 12-week training course in Stockholm, Sweden. Their training comprised all types of network equipment used in the project.

Adapting the training

The initial discussions with Saudi Telephone showed that it was necessary to carry out certain modifications to the proposed training.

The training was intended to be carried out in English, but it was soon clear that the students' knowledge of English was not sufficient, and the training had to be conducted in Arabic. Saudi Telephone also wanted to have som courses modified so that they were better adapted to the Saudi Telephone orginazation. For example, the basic cable jointing course intended for all trainees was cancelled and replaced by a special 5-weeks course intended only for cable jointers. A preparatory course in basic telephone engineering was also prepared.

Preparation of course material in Arabic

The decision to carry out the training in Arabic had far-reaching consequences. Locally recruited instructors would have to conduct courses, first with support form Ericsson's instructors and later wholly on their own. This made new demands on the preparation of the material intended for the instructors. It also proved necessary to rework a large part of the material intended for the students. The course development was carried out by a group of course developers, who developed the English course material. The course material was largely based on video material prepared by Ericsson in Stockholm. It consisted of video programs that showed the work methods for operation and maintanance of the subscriber network. The translation of the course material into Arabic was carried out partly by professional translators and partly by the locally employed instructors, who also tried out the Arabic course material. The translation work proved to be more complicated than expected, since it was difficult to find Arabic equivalents for many ot the technical terms. Many times it was necessary to include English words in the Arabic text.

The instructor material that was prepared consisted of detailed instructions for carrying out the lessons. Fig. 10 shows an example of the student material, in Arabic, for the preparatory course in basic telephone engineering, and fig. 11 shows the corresponding practical experiment.

Recruiting and training Arabic-speaking instructors

There were technicians available in Saudi Arabia who had long experience of Ericsson's network equipment, since such equipment had been installed before TEP started. This made it possible to quickly recruit and train some fifteen Arabic-speaking instructors who were also proficient in English.

The instrutors who were recruited underwent thorough training in the new cable types and the new work methods during the time that the course material in Arabic was being prepared.

Training conducted in Arabic

About a year after the start of the project the training centres in Riyadh and Jeddah had been equipped, the Arabic-

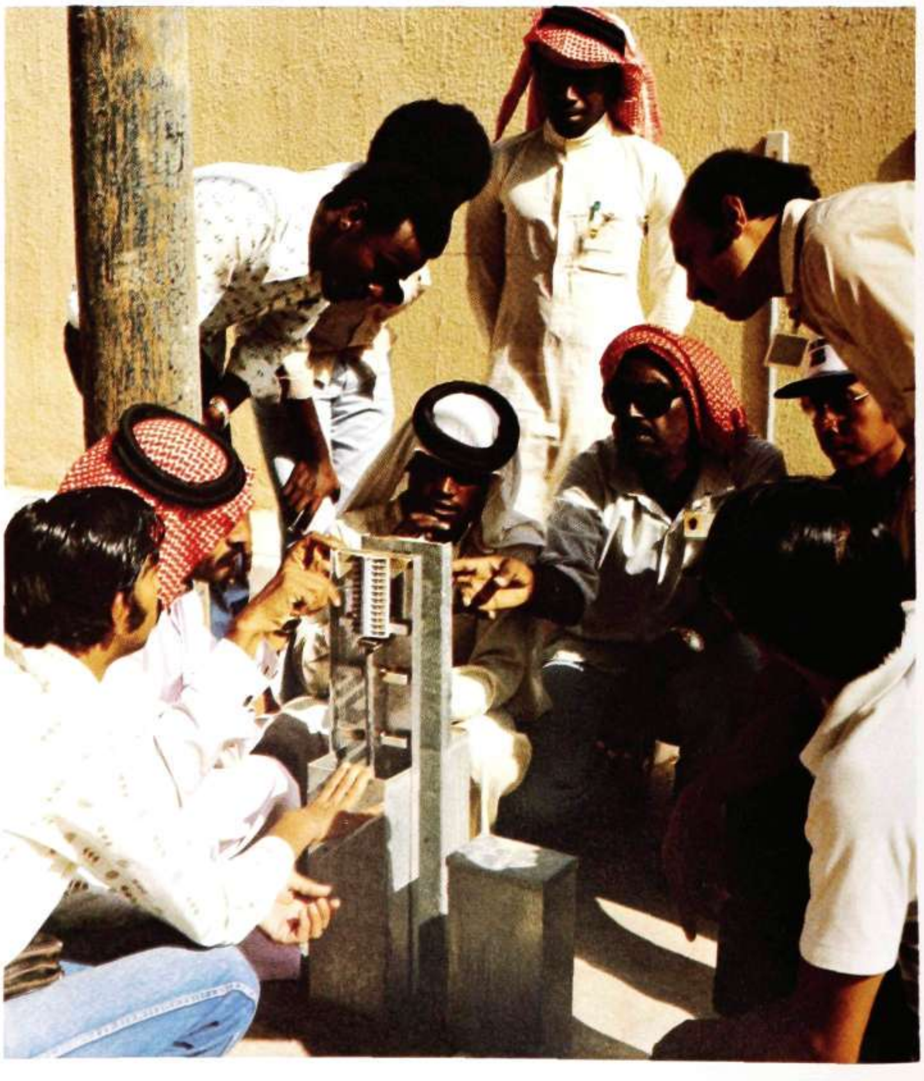


Fig. 12
Connecting up a subscriber line to a distribution point

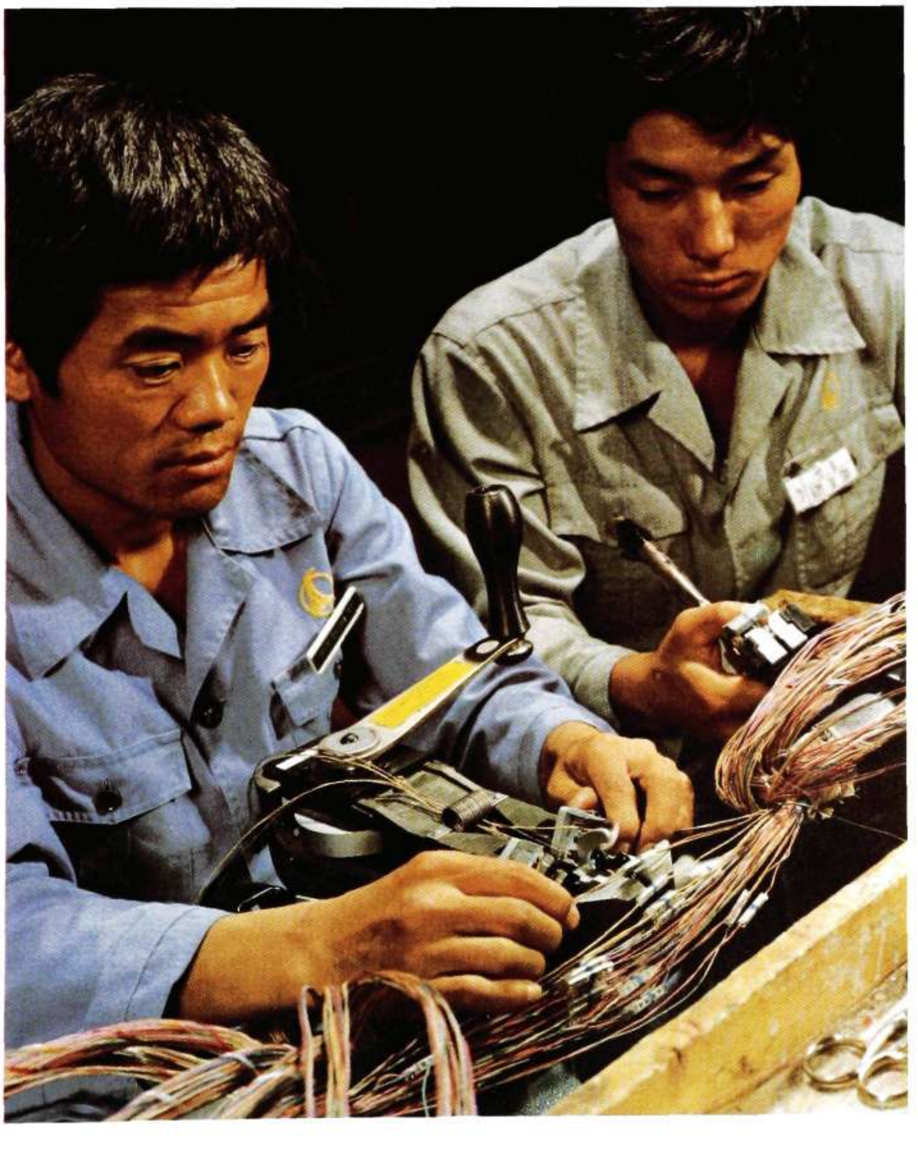
speaking instructors had been trained and all course material was available in Arabic.

The contracted training of the Saudi Telephone personnel could then be started on a large scale. At the peak of the training ten courses were conducted in parallel in Riyadh and Jeddah.

On the whole the training was free from problems thanks to the great amount of work put into adapting the courses. They were well adapted to the Saudi Telephone operation and maintenance organization, and the materials, tools and instruments used during the courses were the same as were used in the field. The detailed guides for the instructors and the frequent use of video in the training were a great help to the instructors and contributed to an even level of quality in the training.

Figs. 11 and 12 show that the training included a considerable amount of practical work.

Fig. 13
Practicing cable jointing at the Dong Ah training centre



Training of personnel in the project organization

As was mentioned initially, at one stage about 10,000 people were involved in the project, and most participants received some form of training during the project. Table 1 and the examples given below illustrate the extent of this type of training.

Approximately 400 employees of Philips – LM Ericsson Joint Venture were trained in the planning and quality control of the network installation work. About 2000 employees of the Korean firm Dong Ah were trained in the following fields:
- building ducts for the primary network
- installing distribution cabinets and boxes
- pulling cables
- jointing cables
- installing cable pressurization equipment
- testing the completed network.

For example, 500 cable jointers were always needed during the course of the project, and a total of more than 1000 cable jointers were trained.

In addition some thirty Muslims were trained for the installation and testing of AXE 10 exchanges in the holy cities of Mecca and Medina.

Training Dong Ah employees in network installation

Ericsson and Dong Ah collaborated in a large training scheme during the initial stage of the project. For example, the scheme included a plan to train 40 engineers and 300 cable jointers in 4 months, with a further 200 cable jointers to be trained within 6 months of the start of the scheme.

The training started in Stockholm, where 40 Korean engineers underwent taining conducted in English. These engineers then led the installation work by Dong Ah in Saudi Arabia.

The next step was to train 16 cable jointing instructors in Stockholm. Twelve of these then went to Saudi Arabia to train cable jointers there. The remaining four instructors held preliminary courses for cable jointers in Korea. This training comprised eight weeks and was conducted with groups of 30 students. An instructor from Ericsson assisted the Korean instructors.

The 8-week preliminary course in Korea was followed by 3 weeks' training in Saudi Arabia, where Dong Ah had, in record time, built its own training centre for 200 students.

Fig. 13 shows students practicing cable jointing at this training centre. 60–70 students participated in each course.

A demonstration network was built up at the training centre which contained all relevant types of cables, cable joints, distribution cabinets and distribution points.

Training in the field of line and network construction, cable pulling and testing of the completed network was usually carried out on the job by Ericsson's instructors. Fig. 14 shows personnel being trained in pulling of optical fibre cable in Riyadh.

Video as a training aid
As has already been described, the training of the Dong Ah personnel was carried out in several stages, some of which required the presence of an interpreter translating from English into Korean. It proved to be difficult to transmit technical knowledge without the loss of quality when not all work methods were well documented in Korean. It was therefore decided that all work methods for jointing and sealing cables, for connecting cables to the terminal blocks in cabinets and distribution points etc., were to be recorded on video tape. This resulted in some twenty video programs being prepared with the speech in Korean. These tapes proved to be of invaluable assistance to the instructors during the training.

The need for written instructions for the installation work was reduced when the video programs were available, but in spite of this a large part of the documentation was translated into Korean.

After about one year Dong Ah was able to take over full responsibility for the training of its personnel. In this way a few instructors from Ericsson were able

Fig. 14
Students learning how to install optical fibre cable

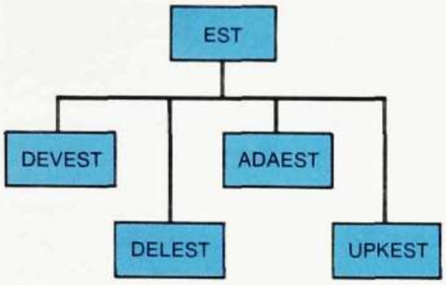


Fig. 15
The Ericsson System for Training, EST

to impart their technical knowledge very quickly and efficiently to a large number of Korean technicians.

Summary

The Saudi project has given Ericsson very valuable experience in the field of training. It has shown the value of adapting the training to the local conditions.

The investment in developing comprehensive course material has also been profitable, both during the contracted training and when Saudi Telephone took over full resposibility for the training.

The collaboration with Saudi Telephone during the adaption work has given Ericsson's instructors deeper insight into the working methods and problems of an administration.

The practical periods, i.e. on-the-job training, provided a measure of the effect of the theoretical training, enabling the instructors to check the skill of the students. It provided feedback which led to further improvement of the courses.

The project has also been valuable for Ericsson's work on improving its standards and its methods for developing, adapting, carrying out and administering courses. These activities are collectively designated the Ericsson System for Training (EST). EST consists of four parts (see fig. 15):
- DEVEST, Development according to EST
- ADAEST, Adaption according to EST
- DELEST, Delivery of training according to EST
- UPKEST, Upkeeping according to EST.

The course development method DEVEST follows closely the ITU Training Development Guidelines.

ADAEST is a method for adapting a course, which has been developed in accordance with DEVEST, to the local conditions prevailing in a particular country.

DELEST comprises policy and routines for carrying out training projects which have been developed in accordance with DEVEST or adapted in accordance with ADAEST. It is important for the instructors to have support in the form of policy and routines during the implementation, for example when testing basic knowledge, evaluating the students, checking attendance and grading results.

UPKEST describes the rules that apply for the updating and improvement of courses.

On December 13, 1980 Ericsson had fulfilled its contracted assignment and Saudi Telephone took over full responsibility for the training. At the same time Saudi Telephone received the complete instructor and student documentation for all courses, prepared in accordance with the Ericsson standard. This completed the largest training project Ericsson has undertaken hitherto.

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Optimizing Telecommunication Networks

Staffan Bergsten and Anders Rudberg

Ericsson has long been developing methods and digital processing systems for optimizing telecommunication networks.
The authors give a general description of network optimization and discuss forecasts, quality requirements, problems and different approaches. They describe briefly the digital processing systems that are now in use or being developed and indicate the general trend of future development of systems and methods.

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Today the total investment in telecommunication networks throughout the world amounts to about $ 650 billion. Forecasts indicate a rise in the rate of investment, and during the period 1980-1990 equipment worth about $ 450 billion will be installed. The corresponding increase in the number of lines is shown in fig. 1.

It is not only the magnitude of the investment that is characteristic for telecommunication networks but also the fact that it is a long-term investment. Economic optimization is an essential part of the planning required when extending a network. The optimization is based on forecasts for the increase of the number of subscribers and for traffic growth, routing and transmission plans, the cost of different network components etc.

Ericsson has long been concerned with these problems. A study of the Paris network was carried out as early as 1950, and since the beginning of the 1970s some hundred different networks have been planned with the aid of the methods and digital processing (DP) systems developed by Ericsson. The methods and systems are continually being developed further as new technology, new services, new routing philosophies etc. are introduced.

Network optimization is performed within Ericsson not only in connection with tenders for systems and networks, but also as a consultant service to customers. The methods and DP systems are also useful when developing new products and systems, and when analyzing the economical and structural effects that the introduction of new products will have on the network.

Forecasts

The demand for telecommunication can be described by means of two parameters:
- the number of subscribers to be connected to the network
- the traffic generated by the subscribers.

The planning requires forecasts describing this demand in both the time and space domains.

The quality of the planning is to a great extent dependent on the quality of these forecasts. The longer the period covered by a forecast the more uncertain the forecast. This means that the long-term planning need not be as detailed as the short-term.

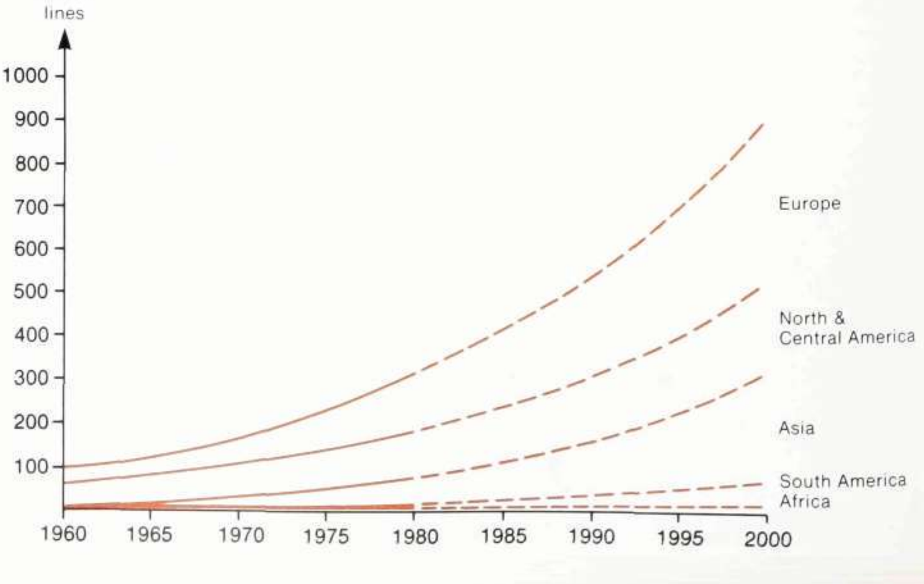

Fig. 1
The total number of lines in operation throughout the world during the period 1960–1980 and a forecast for the period 1980–2000. Each line can be assumed to correspond to an avarage investment of $ 2,000.

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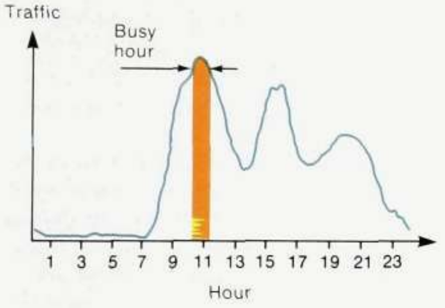


Fig. 3
Variations in the traffic during the 24 hours of the day

The forecasts should comprise any other services for which the network may be used in the future as well as telephony.

The type of information required in the forecasts is described below.

Subscriber forecast

The subscriber forecast should describe the change in the number of subscribers during the planning period. It may be necessary to divide the subscribers into different categories, such as home, business and office subscribers, both in the subscriber and the traffic forecasts.

The accuracy required in describing the geographical distribution of the subscribers varies depending on the problem to be solved. For example, one subscriber forecast per exchange area is sufficient when calculating the junction network, whereas a detailed description of the geographical distribution of the subscribers is necessary when siting exchanges in a multi-exchange area.

There are different methods for preparing the detailed description. One way is to consider existing and planned cable distribution cabinet areas in the network. To each such area a number of subscribers are assigned in accordance with the forecast, fig. 2a. This method is suitable in the case of low growth or a short planning period, i.e. when consideration must be taken of the existing and the planned subscriber network. This method has certain limitations when the growth is more rapid or when the planning concerns a completely new area, i.e. when the subscriber network must be changed extensively or when a new network must be established. A more detailed description of the geographical distribution of the subscribers is obtained if the area is divided into squares and the forecasted number of subscribers is given for each square, fig. 2b.

In rural areas, where the subscriber distribution is characterized by several subscriber concentrations scattered over a large area, a mixture of the grid

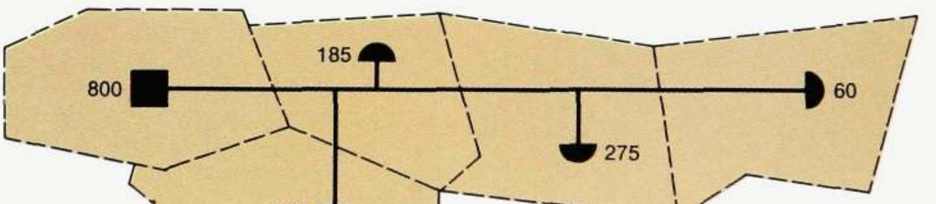


Fig. 2a
The subscriber forecast defined with the aid of planned and existing cabinet areas. In each line cabinet distribution area the number of subscribers is given for different years and different types of subscribers. In the optimization all subscribers in a line cabinet distribution area will belong to the same exchange area

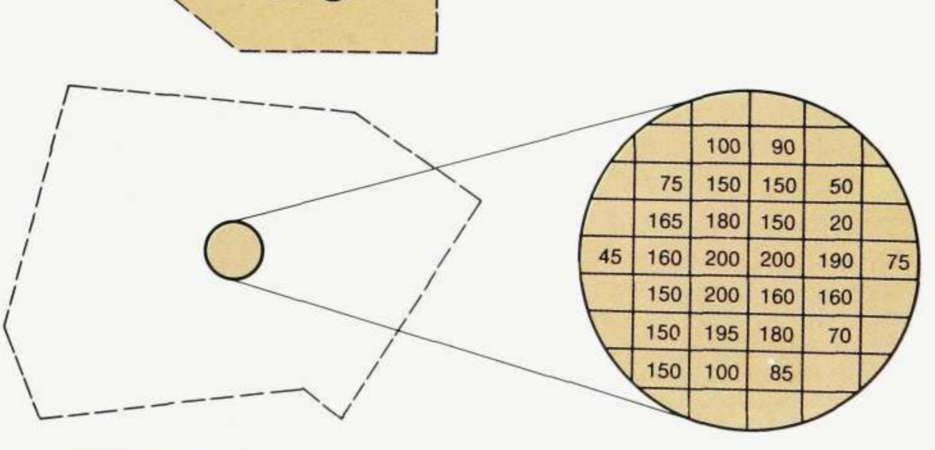


Fig. 2b
The subscriber forecast defined with the aid of a grid. Each grid square has the dimensions of about 250×250 m². The number of subscribers is given for each square, possibly for different years and subscriber categories

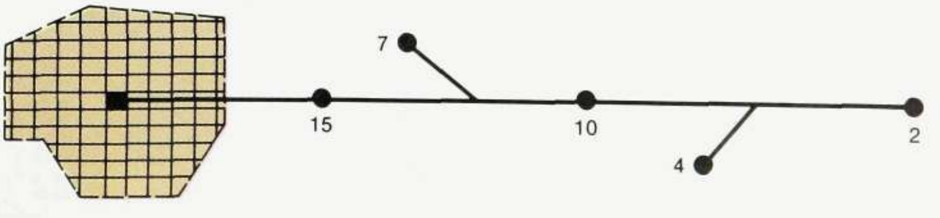


Fig. 2c
The subscriber forecast described with the aid of a combination of a grid and discrete points

Box no. 1

Some simple traffic relationships

Fig. A
The utilization of the lines in a route as a function of the traffic offered to the route at a given level of congestion.
The more traffic offered to the route, the better the utilization of the lines in the route. This fact is used in, for example, networks with alternative routing, where the rejected traffic from a number of direct high usage routes is directed to a common overflow route.

Fig. B
The utilization of the lines in a route and the congestion on the route as a function of the number of lines in the route, for a given amount of offered traffic.
The optimum number of lines will in most cases fall within the section marked in the figure, since both high utilization and low congestion are desired.

Fig. C
The overload capability of a route, as a function of the size of the route.
The figure shows an example where the congestion on the route must not exceed 1% with normal load, which gives the traffic volume at normal load, and where the congestion must not exceed 10% with overload, which gives the traffic volume at overload. The overload capability is here defined as the ratio between the increase in traffic during overload and the normal traffic.
The figure shows that small routes are better able to handle overload than large routes. However, when considering this fact it must be recognized that larger routes are not likely to be offered such large relative overloads as smaller routes.

Fig. D
The remaining traffic handling capability of a route as a function of the number of faulty lines in the route.
The remaining traffic handling capability is expressed as a percentage of the traffic the route normally handles. With an increasing number of faulty lines in the route the congestion will increase, and the traffic the route can handle will decrease.

Congestion = 1%

Route size	Offered traffic
10 lines	4.46 erlang
20 lines	12.03 erlang
100 lines	84.06 erlang
500 lines	474.04 erlang

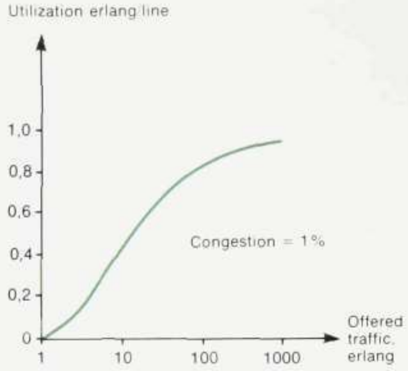


Fig. A

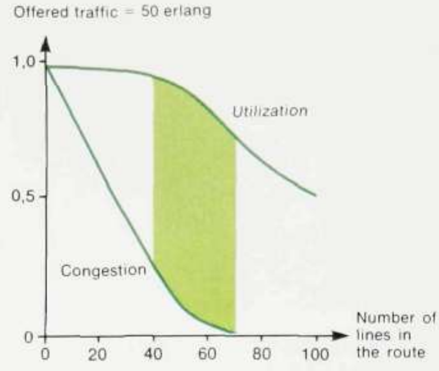


Fig. B

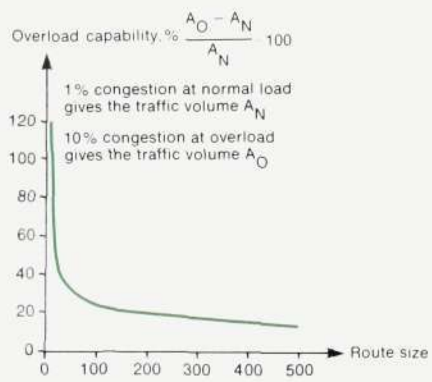


Fig. C

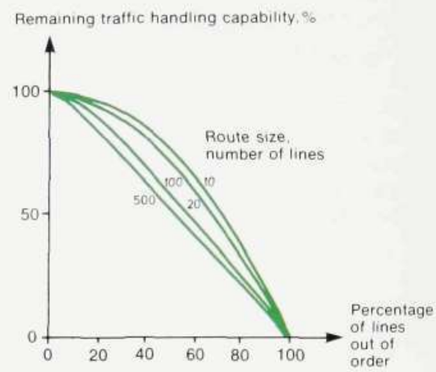


Fig. D

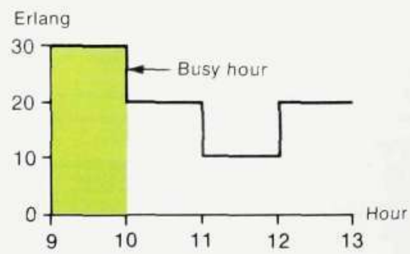
method and dots representing the number of subscribers is often the most suitable method, fig. 2c.

Traffic forecast

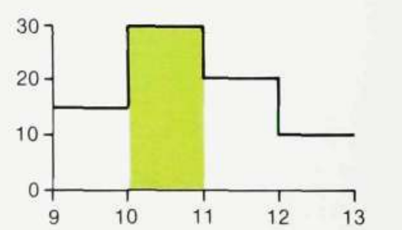
The traffic intensity varies during the day, fig. 3, as well as from one day to another during the week. There are also seasonal variations, with more traffic before major public holidays and less traffic during holiday periods etc. These traffic variations are used to determine when the busy hour occurs, and also the amount of traffic during this hour. All dimensioning is performed on the basis of the traffic during the busy hour. Box no. 1 describes some simple traffic relationships.

The busy hour can occur at different times of the day for different traffic categories. These traffic categories may use common routes. If the busy hour traffic from the different traffic categories are simply added during the dimensioning of such a common route the result obtained may be unnecessarily high. In order to obtain correct dimensioning of the network it is very useful to define traffic profiles that give the variations in the traffic during the whole 24 hours of the day. On the basis of these profiles the busy hour for each route can be calculated, fig. 4.

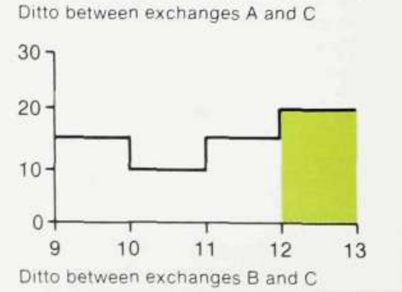
The traffic variations depend on the different traffic intensities and traffic inter-



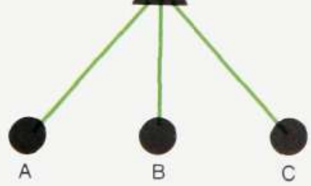
Traffic profile for the traffic interest between exchanges A and B



Ditto between exchanges A and C



Ditto between exchanges B and C



Busy hour traffic in erlang on the routes

Route	Disregarding the traffic profiles	Considering the traffic profiles	Time (a.m.)
A-T	30+30=60	20+30=50	10-11
B-T	30+20=50	30+15=45	9-10
C-T	30+20=50	30+10=40	10-11

Fig. 4
An example of how non-coincident busy hours in a network can be considered. The traffic between exchanges A, B and C is routed via tandem exchange T. If non-coincident busy hours are considered, the routes A–T, B–T and C–T can be dimensioned for a lower traffic than the traffic indicated by the sum of the busy hours for the three routes

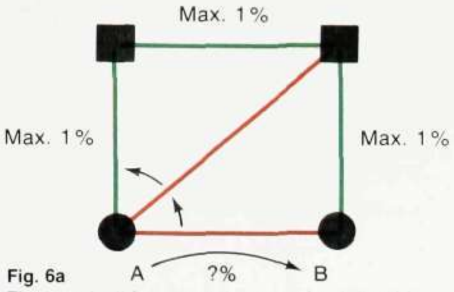


Fig. 6a
The permissible congestion must be defined for each final route in the network. The figure shows an example where these values have been set to 1%. Since, for economic reasons, high usage routes may be changed into final routes during the optimization process, the congestion limits must also be defined for such routes. The resultant congestion for traffic from A to B is more or less arbitrary since it is dependent on the number of lines in the high usage route. However, it cannot exceed the sum of the congestions on the final routes used for this traffic case

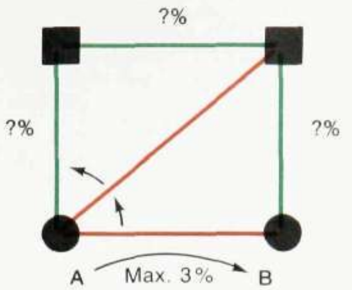
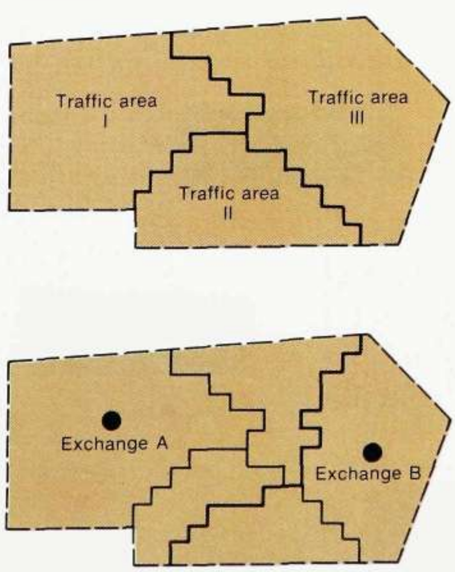


Fig. 6b
Point-to-point congestion must be defined for all traffic cases in the network. This value has been set to 3% in the figure. The congestion requirements that will affect the dimensioning of the final routes will be dependent on the proportion of the traffic that is handled by the high usage routes. Thus the congestion on the final routes becomes arbitrary but cannot exceed the set point-to-point value

—— Low loss route, green
—— High usage route, red

Fig. 5
A network with two exchanges, A and B, is divided into three traffic areas. The figure shows an example of a traffic matrix for the three traffic areas. To calculate the inter-exchange traffic matrix it is also necessary to know the number of subscribers per exchange area, per traffic area and per combination of exchange and traffic area



ests among the subscribers in the network. These traffic characteristics also vary from one subscriber category to another. The subscriber forecasts often show different growth rates for different subscriber categories. This necessitates a traffic forecast for each category if a correct forecast is to be obtained for the whole network.

The traffic forecasts should describe the traffic growth in the network during the planning period. These forecasts form the basis for traffic matrices which define the traffic distribution in the network. In order to be able to dimension the junction network, a traffic matrix which describes the traffic requirements between exchanges is required. This matrix is based on the traffic offered by the subscribers and their exchange assignment. If the optimization also concerns the number of exchanges and the exchange areas, the boundaries of the exchange areas will be altered during the optimization process, and thus the traffic requirements between the exchanges will also change. A method is therefore needed which makes it possible to calculate, on the basis of the subscribers' traffic, the inter-exchange traffic requirements.

It is not possible to describe the traffic requirement of each individual subscriber. However, it is too great a simplification to assume that all subscribers in the network have the same traffic requirements. The network is therefore preferably divided into traffic areas, within which the subscribers are assumed to have similar traffic characteristics.

↱	I	II	III	Σ
I	50	50	150	250
II	50	250	200	500
III	75	175	500	750
Σ	175	475	850	1500

Traffic between the traffic areas

↱	A	B	Σ
A	578	356	934
B	379	187	566
Σ	957	543	1500

Traffic between the exchange areas

The traffic requirement between the traffic areas can then be specified in a traffic matrix, fig. 5. As the traffic areas remain unchanged during the optimization process, it is possible to calculate the traffic requirements between the exchanges when the exchange area boundaries are altered.

Quality requirements

Service quality is a general measure of the ability of the network to satisfy the needs of the subscribers. Economic optimization would be meaningless without quality requirements, since there would then be no criteria for how well the network should meet the needs. The service quality concept covers a wide range. From the point of view of optimization the most important factors are availability (grade of service) and intelligibility (transmission quality).

Grade of service

The grade of service is determined by the highest permissible congestion in the network. This is either defined for each final route (last choice route) or as permissible point-to-point congestion, fig. 6. Some types of traffic routing are described in box no. 3. Point-to-point congestion must be considered a better measure of the service quality than route congestion since it is closer to the congestion experienced by the subscriber. In the optimization process the permissible point-to-point congestion is distributed over the routes used for the traffic case being considered. Routes that are used for several traffic cases are then dimensioned for the traffic case with the most stringent requirements. This means that the final routes in the network will have more or less arbitrary congestion values. If the method of final route congestion is used instead, the congestion value will be defined in advance for each such route. This simplifies control and supervision of the network.

Transmission quality

Intelligibility is defined by means of the so-called reference equivalent, which is a subjective measure of the average quality of a connection set up between

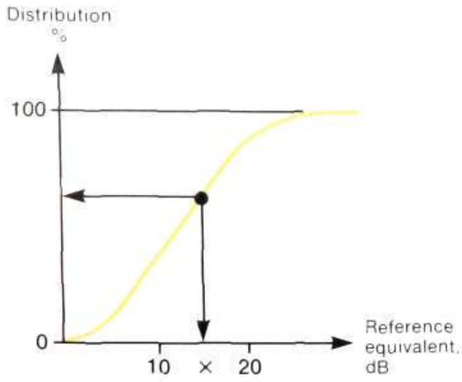


Fig. 7
The reference equivalent is a subjective measure of the quality of a connection set up between two subscribers. It defines the sound level of the speech expressed in decibels (dB) relative an international standard (NOSFER) set by CCITT. The reference equivalent is effected by all attenuation and amplification on the connection and by the electrical and acoustic characteristics of the microphone and the receiver. It should be neither too hign (=too low sound level in the called subscriber's receiver) nor too low (=too hingh sound level). The optimum value is about 10 dB.
The distribution of the reference equivalent for the calls set up during the busy hour gives a picture of the transmission quality of the network. The distribution takes into account the type of traffic and the traffic intensity, i.e. common types of connections have greater effect than rare types

two subscribers, fig. 7. The intelligibility requirement therefore leads to requirements for limit values for permissible attenuation in different parts of the network and will partly govern the choice of transmission equipment.

The congestion and transmission requirements control the availability and intelligibility in the network. The network is assumed to work under ideal conditions, i.e. the equipment is in good working order and the traffic does not exceed the forecasted values. However, the quality requirements described above do not guarantee an acceptable service level under abnormal conditions, e.g. when overload or faults occur. There are no internationally agreed criteria for such cases that are suitable for network optimization. Generally speaking, such abnormal situations should be considered when dimensioning networks.

The quality requirements will of course affect the cost of the network. The various requirements are often defined independently of each other and without taking cost into account. The difficulty is to balance all factors, since too high requirements result in an unnecessarily expensive network, whereas too low requirements result in a network which does not meet all customers' needs.

Problem definition

The aim of network optimization is to determine the amount of equipment needed in the network, where the equipment is to be installed, and when the investment is to be made in order to obtain the minimum network cost, fig. 8.

The network optimization should determine
- the number of exchanges of different types
- the geographical position of these exchanges
- the size of the exchanges and the exchange areas
- the size and structure of the junction network
- the type of transmission equipment to be used in the network
- the structure of the subscriber network.

This applies for urban and rural networks as welll as for long-distance networks. However, certain items are more important in one type of network than in another. For example, in urban networks the number of exchanges and their locations are of vital importance, whereas in rural networks it is more important to choose the right type of transmission equipment between the subscribers and the local exchange.

The method of solving the various problems is also dependent on the type of network. The junction network in a city may consist of a complicated network with alternative routing, which requires special optimization methods, whereas a rural network often consists of a few direct routes that are easy to dimension.

Methodology

Fundamental questions when developing methods and programs are whether

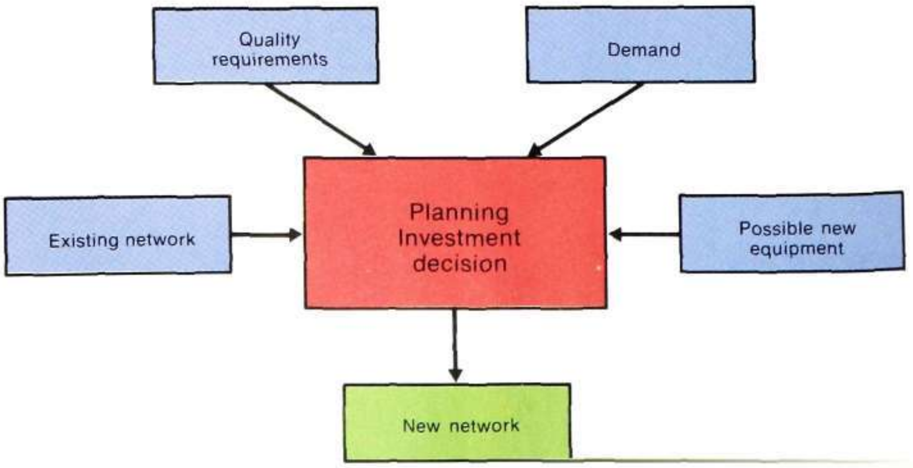


Fig. 8
Forecast demands and quality requirements form the basis for the planning of extensions in the telecommunication network. The planning must take existing networks into consideration to an appropriate extent. It must also define the types of exchanges and transmission equipment that can be introduced in the network. Network optimization is an essential part of the planning process.

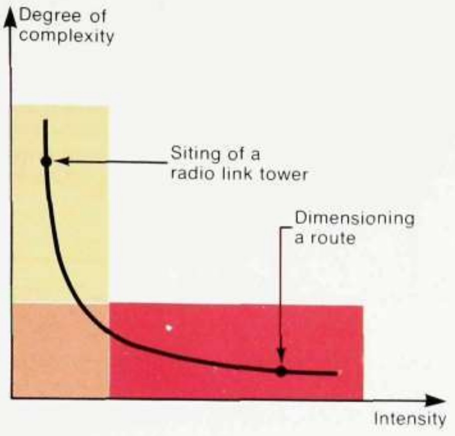


Fig. 9
The principle for the distribution of tasks between the planner and the computer in interactive programs. By degree of complexity is meant the difficulty in describing the problem mathematically

Decision by the planner

Decision by the computer

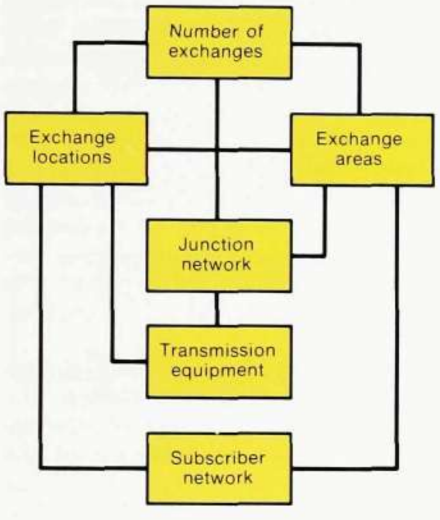


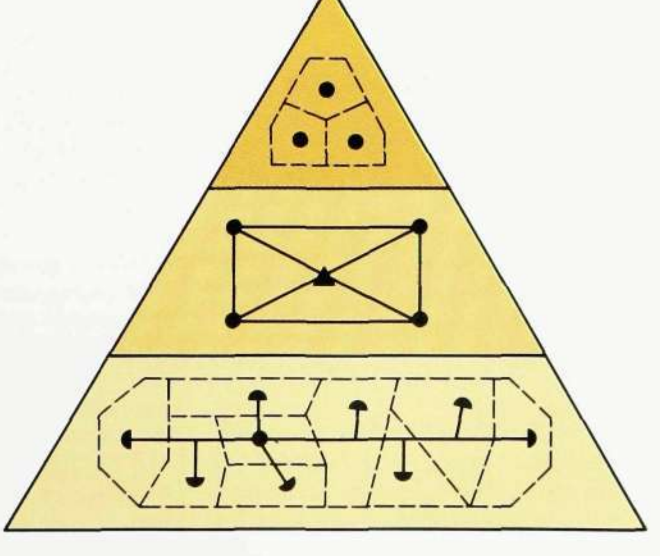
Fig. 10
The realtion between the various problem fields. No problem can be solved properly without considering the solution of the other problems

Fig. 11
Problem pyramid dividing the planning problems into levels. Each level requires a certain degree of detail when describing the model

Number of exhanges
Exchange locations
Exchange areas

Junction network
Transmission network

Subscriber network



cross-sectional or dynamic models should be used, to which extent consideration should be paid to the existing network, and whether the computer programs should be designed as batch or interactive programs.

A dynamic batch program, which takes the existing network into consideration to the maximum extent, would be the ideal aid. However, such a program would be very difficult to develop, and is not even necessary for specific problems.

To take the existing network into consideration to the maximum extent would lead to very complicated methods and require a great amount of input data for describing the network. Most methods therefore only consider existing equipment to a limited extent. For example, the distance between exchanges can be calculated on the basis of the geographical layout of the existing cable routes, whereas the actual supply of cables and capacity of the routes might not be regarded. Networks with a high growth rate and existing equipment that is well utilized may, in principle, be considered "virgin territories" and may thus be described by simpler models without reducing the quality of the result. With a low growth rate and a lower degree of utilization it is essential to take existing equipment into account.

With cross-sectional models the network is optimized for a given demand at a certain point of time. With dynamic models, on the other hand, the network extension is optimized over a specified planning period. Dynamic models require existing networks to be taken into account, since it is necessary, at each extension stage, to consider the state of the network after previous stages have been completed.

Cross-sectional models may also be used for dynamic studies. The network is then optimized for a number of points of time between the present time and the end of the planning period. One difficulty is then to decide to what extent solutions for later points of time should be allowed to govern the solutions for the earlier points of time, since the uncertainty of the forecasts increases with time.

Solutions to different problems have different effects in the long term. For example, the decision where to place exchanges has a more long-lasting effect than the decision concerning the sizes of the routes in the junction network.

A dynamic approach leads to more complicated models and algorithms, and means that a number of alternative extension strategies have to be investigated. In order to limit the calculation work the optimization aid can be designed as an interactive program in which the planner's experience is utilized. Interactive programs are also used when it is not possible to frame the planner's experience in a simple mathematical form or to describe the problem in mathematical terms. In an interactive program the dialogue between the planner and the computer has to be designed with great care. The dificult decisions should be made by the planner, whereas frequent but uncomplicated decisions can be made by the computer, fig. 9. The development cost of an interactive program is higher than the cost of the corresponding batch program.

Solution methods

The different problem fields discussed above, fig. 10, are closely related. No

Fig. 12a
The cost of the equipment can often be described as a step function, where the size of each step depends on the modularity of the equipment

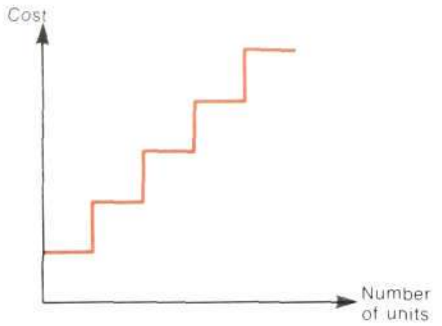
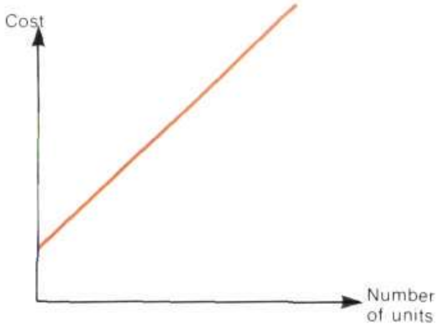


Fig. 12 b
Linear approximations are often used in optimization



Box no. 2

INFORMATION REQUIRED FOR NETWORK OPTIMIZATION
How the information is described and the degree of detail required are dependent on the type of optimization to be performed.

Forecasts
- Subscriber forecast
- Traffic forecast

Quality requirements
- Transmission plan
- Grade of service plan

Existing networks
- Exchange locations
- Exchange areas
- Transmission equipment (location, capacity)
- Cable routes (junction network and primary network)
- Buildings (location, capacity)

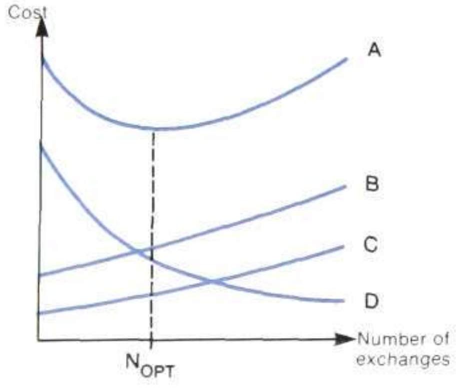
Exchange equipment (existing and new)
- Type
- Capacity
- Routing possibilities

Transmission equipment (existing and new)
- Type
- Capacity
- Transmission characteristics

Cost information
- Exchange equipment
- Transmission equipment
- Buildings
- Sites
- Ducting
- Rate of interest
- Planning horizon

Geographical data

Fig. 13
The network cost as a function of the number of exchanges. When the number of exchanges increases the subscriber network cost decreases because of shorter distances between subscribers and exchanges. At the same time the cost of the junction network increases, since the traffic is distributed over several routes, which leads to lower utilization of each line. A larger number of exchanges will of course also increase the cost for exchange buildings, sites etc.
A Total network cost
B Junction network
C Exchange equipment, buildings etc.
D Subscriber network



N_{OPT} = Optimum number of exchanges

problem can be solved without considering the other problems. However, simultaneous solving of all problems is not feasible, for both practical and theoretical reasons.

The set of problems can be divided into different levels, where each level requires a certain degree of detail in the mathematical model, fig. 11. The description of the studied section of the network must of course be more detailed the further down into the network the study is allowed to penetrate. For example, at the lowest level, the subscriber network, it is necessary in the model to take into account the locations of the subscriber line distribution cabinets, cable dimensions etc. However, such a detailed description is quite unnecessary when deciding the number of exchanges, at the top level. This circumstance may be exploited when determining methods of approach and preparing models.

In models for solving problems at a lower level it is assumed that the problems at the higher levels have been solved, but the converse is not true. This means that in models for solving problems at a higher level it cannot be assumed that problems at the lower levels have been solved. Simplified models may, however, be used. For example, in the mathematical model used for optimizing the subscriber network the location of the exchange can be assumed to be known, whereas the model used for optimizing the location of this exchange must include a description of the subscriber network.

To minimize the network cost to a suitably defined level a mathematical model is required which formulates the relation between, on the one hand, the demand and the quality requirements and, on the other hand, the cost of the network. Network optimization means minimizing each cost function with regard to the variables that are applicable to the problem.

The network cost can be divided into three main parts:
- the cost of the subscriber network
- the cost of the junction network
- the cost of the exchange equipment, sites, buildings etc.

Each part may be further divided, with a degree of detail that is determined by the problem level, fig. 11. The cost is a function of the amount of equipment in the network. Mathematically the cost can usually be expressed as a step function, fig. 12a. The algorithms that are used in the optimization models should therefore permit arbitrary cost functions. However, the algorithms used today are often based on linear approximations, fig. 12b. The development trend towards integrated digital networks means that it will be increasingly important to consider the modularity of the equipment. The algorithms that are now being developed therefore pay greater attention to non-linear cost functions of the type shown in fig. 12a.

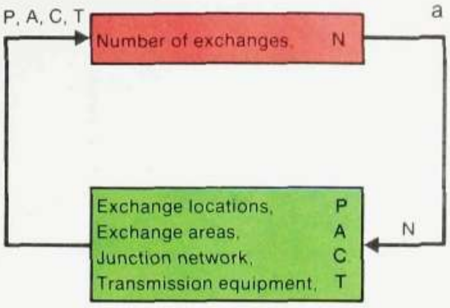
Some methods for solving the optimization problems are discussed below. Box no. 2 summarizes the type of information that should be available for the optimization.

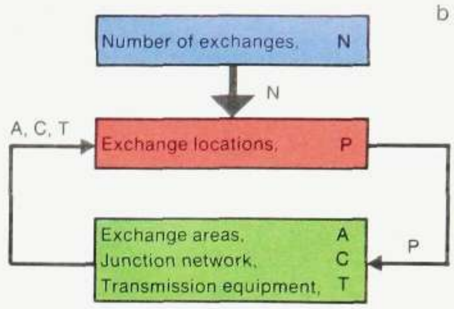
Number of exchanges

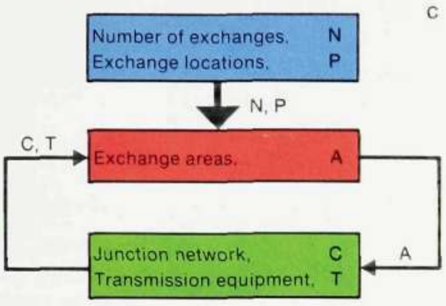
The cost of the subscriber network, the junction network and the exchange equipment is a function of the number of local exchanges in the network, fig. 13. The sum of these costs has a minimum value which corresponds to the optimum number of exchanges. In this context the concept local exchange also signifies remote subscriber switches. For each number of exchanges there is a best solution as regards exchange locations, exchange areas, junction network etc. Since all these factors are related, fig. 10, some type of iterative procedure must be used for the optimization, fig. 14a. The optimum number of exchanges can be determined by successively increasing the number of exchanges until the lowest possible network cost has been obtained.

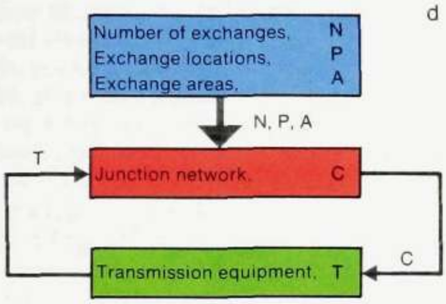
Similar methods could be used to decide the number of tandem and transit exchanges. In most cases this is not necessary since the number of possible alternatives is limited. A simple method that can be used instead is to optimize the junction network for a few alternative numbers and locations, and then choose the most economical alternative

Fig. 14a–d
The principle of the iterative process. The problems to be solved are represented by the contents of the red boxes. When solving these problems the variables in the blue boxes are assumed to be given. The interdependence between the red and green boxes makes iteration necessary. Each green box represents an iterative process









Exchange locations

The location of each exchange in the network should be determined so that the network cost is as low as possible, fig. 15. Such a location does not usually coincide with the center of gravity of the subscriber locations within the exchange area, since the transmission requirements can mean that distant subscribers have to be connected by means of a more expensive type of cable, which affects the exchange location and moves it towards these subscribers. Routes to and from the exchange can also affect the location in a similar way.

The choice of exchange location is seldom completely free. The availability of sites, buildings etc. limits the choice. The iterative process is illustrated in fig. 14b.

The location of tandem and transit exchanges is primarily decided by the availability of sites and exchange buildings, since the location of these exchanges affect the cost of the junction network to a lesser degree.

Exchange areas

For given exchange locations, the exchange areas must be defined, i.e. the subscribers must be assigned to the exchanges. The cost of connecting a subscriber to an exchange is mainly dependent on the length and type of the subscriber line. However, the subscribers will not always be assigned to the closest exchange, since the junction network cost for handling the traffic from the subscribers may be different depending on in which exchange the traffic is handled. When determining exchange areas, consideration must also be paid to natural boundaries, such as rivers etc., and to the existing subscriber network. The iterative process is shown in fig. 14c.

Junction network

When the traffic between the local exchanges is known, and the locations of the local and tandem exchanges have been decided, the junction network can be dimensioned. For networks with alternative routing this means determining the most economical sizes of the high usage routes, fig. 16. The final routes can then be dimensioned with the aid of the defined congestion criteria. As the traffic rejected from a high usage route has a different characteristic compared with pure chance traffic, this must be taken into consideration when dimensioning the final routes.

If there is no alternative routing the optimization is limited to determining, for each traffic case, the most economical routing, i.e. a choice between a direct low loss route or a tandem route.

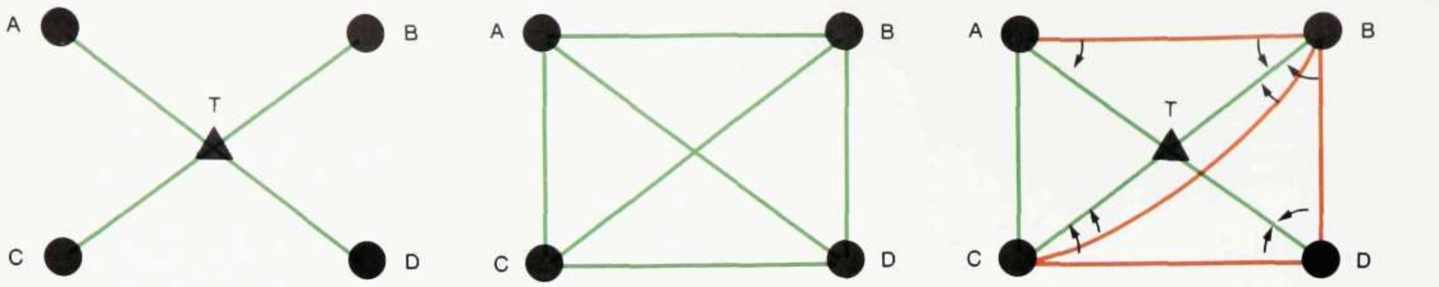
Similar methods may be used for all types of junction networks, urban net-

												Row total	Accumulated total
					50	50						100	
					50	50	40					140	240
				50	75	50	70	30	10			285	525
			50	60	75	80	90	80	10	10		455	980
	75	100	100	200	100	150	100	75	50	10	10	970	1950
	100	100	150	200	200	200	175	150	50	20	20	1365	3315
	75	50	100	100	200	200	150	150	50	20	3382	1095	4410
	20	50	100	150	125	10	25	100	50	20		650	5060
		40	50	150	100	100	100	75	75			690	5750
		40	50	75	30	75	150	100	75			595	6345
			20	20	20	50	10	30	50			200	6545
				20	20	20	10	50	20			140	6685
					15	15	10	20				60	6745
					3382	10	10					20	6765
Column total	270	380	620	1025	1060	1060	940	860	440	80	30		
Accumulated total		650	1270	2295	3355	4415	5355	6215	6655	6735	6765		

$$\frac{6765}{2} = 3382.5$$

■ = Theoretically optimum exchange location

Fig. 15
Calculation of the optimum exchange location in an exchange area. It is assumed in this example that the same type of cable is used throughout the subscriber network and that the cable laying follows the coordinate system. The effect of the junction network on the exchange location is disregarded. The theoretically optimum exchange location then coincides with the point of intersection of a vertical and a horizontal line which are placed so that there is the same number of subscribers on each side of each line



Box no. 3

SOME TYPES OF TRAFFIC ROUTING
IN THE JUNCTION NETWORK

Left:
Star network
Centre:
Mesh network
Right:
Alternative routing
Low loss route
High usage route

Direct route:
A general expression for a route between two exchanges. It can be a high usage or low loss route. The expression is usually applied for routes between local exchanges.

High usage route:
A route whose rejected traffic is offered to another route. Thus a high usage route is never a final route.

Final route (Last choice route):
A route whose rejected traffic is not offered to another route.

Low loss route:
Synonomous with final route, with special emphasis placed on low congestion.

Tandem route:
A route between a local exchange and its tandem exchange. Normally a final route.

works, national long-distance networks etc. The iterative process is illustrated in fig. 14d.

Transmission equipment

The dimensioning of the network gives the number of circuits required between the exchanges. How this demand should be met depends on the capacity of the existing transmission network, the extent to which it can be extended and what new types of transmission systems and media that can be introduced.

Optimizing the transmission equipment means calculating, for each route in the network, the combination of transmission systems and transmission media and the geographical route that give the lowest overall network cost. What system or systems and media are allocated to a route depends on how the existing transmission network can be utilized, the size of the route and the distance between the exchanges. A link between two nodes in the transmission network usually contains several routes, which makes it possible to use common transmission equipment, fig. 17.

Subscriber network

The subscriber network should be optimized for each exchange area. This means deciding the locations of the subscriber line distribution cabinets, the cabinet distribution areas, the amount of ducting and the type of transmission equipment. The use of remote subscriber switches affects the design of the subscriber network since the traffic concentration point is moved out into the network.

Many of the above-mentioned problems may be solved by manual methods, but the amount of data to be processed, the iterative nature of the problems, and the number of alternatives to be studied implies that access to computers and efficient DP systems is essential for a serious treatment of the optimization problems.

DP systems

Ericsson performed the first network optimization on computer as early as the end of the 1950s. Since then both the network optimization methods and the DP systems have been radically improved. Today Ericsson has a range of DP systems for optimizing all parts of the telecommunication network, fig. 18. The predominant DP system is called PLANET. It comprises over 15,000 FORTRAN statements, divided into approximately 160 subprograms. PLANET has a modular structure which makes it possible to combine subprograms in order to obtain programs for different purposes. Fig. 18 shows some examples.

PLANET is based on a cross-sectional approach. The optimization algorithms utilize the concepts of marginal cost and marginal utilization. PLANET can take into consideration such factors as the modularity of PCM systems, the limited availability of the switching systems and non-coincident busy hours. When dimensioning the junction network the measure of the grade of service can either be point-to-point congestion or final route congestion. PLANET allows great freedom when describing the routing in the network. The routing can also be optimized to a certain extent. The existing network can be taken into account by defining the cable routes, by fixing the number of exchanges, exchange locations or exchange areas, by limiting the capacity of exchange buildings etc. The PLANET programs are usually of the batch type, but there are some interactive variants.

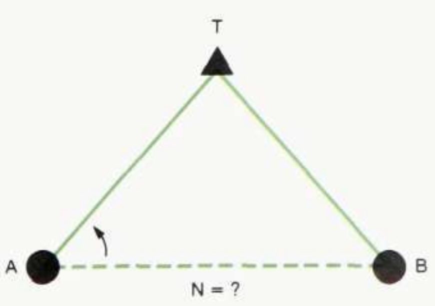


Fig. 16a
The high usage route between exchange A and exchange B is to be optimized

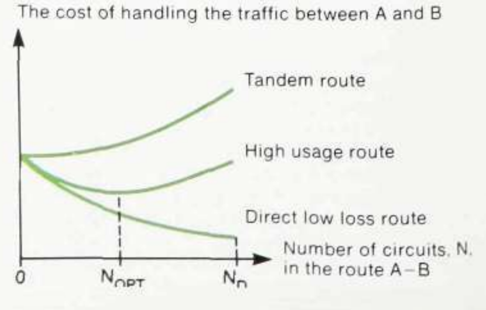


Fig. 16b
The optimization may result in three different basic solutions:
Tandem route
It is not economical to have a direct route between A and B. All traffic is routed via the tandem exchange T. N = 0.
Direct low loss route
It is not economical to have alternative routing. All traffic is carried by the route A – B. The number of circuits, $N = N_D$, is determined by the congestion limit.
High usage route
A certain proportion of the traffic is carried on the direct route A – B. The rejected traffic is routed via the tandem exchange T. $N = N_{OPT}$

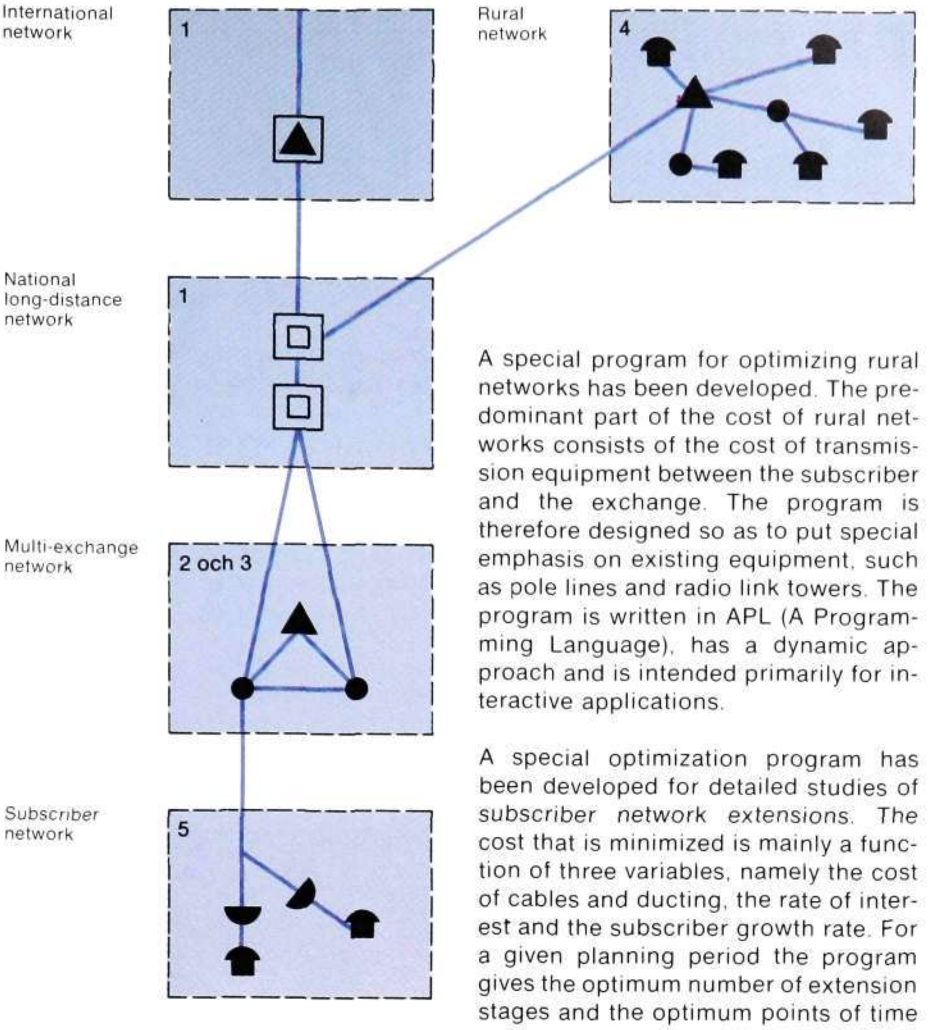


Fig. 18
Network diagram with corresponding DP aids. Programs 1, 2 and 3 are included in PLANET, whereas programs 4 and 5 are special aids for optimizing rural and subscriber networks

A special program for optimizing rural networks has been developed. The predominant part of the cost of rural networks consists of the cost of transmission equipment between the subscriber and the exchange. The program is therefore designed so as to put special emphasis on existing equipment, such as pole lines and radio link towers. The program is written in APL (A Programming Language), has a dynamic approach and is intended primarily for interactive applications.

A special optimization program has been developed for detailed studies of *subscriber network extensions.* The cost that is minimized is mainly a function of three variables, namely the cost of cables and ducting, the rate of interest and the subscriber growth rate. For a given planning period the program gives the optimum number of extension stages and the optimum points of time for these. A detailed specification is obtained for the cable types and ducting to be installed during each extension stage. The program is a dynamic batch program written in FORTRAN.

Future developent

The development of new methods and DP systems is mainly governed by three factors:
- technical development
- demand for new telecommunication services
- new quality requirements.

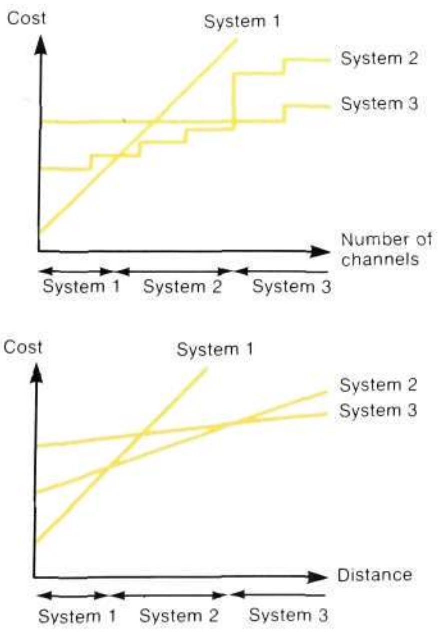
The general technical development results in new telecommunication products, for example higher order PCM systems and digital radio link systems, optical systems and subscriber multiplexors. Stored program control and digital technology also provide new routing and traffic control facilities in the network.

The digitalization of the telecommunication network will make the development towards *integrated services* digital networks possible. In addition to telephone traffic, the network will be able to carry different types of data traffic.

With digital technology the transmission requirements will have less influence on the optimization. On the other hand the reliability and overload characteristics of the network will be more important. Integrated services digital networks will also result in new quality requirements.

These changes will of course affect the cost models and algorithms used in network optimization, and the DP systems are therefore continually being adapted and further developed.

Fig. 17a–b
The cost as a function of the number of channels (top) and the distance (bottom) for different types of transmission systems. The economical ranges for the various systems are indicated



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A New Electronic PMBX, ADD 20

Åke Andersson

In spite of the fact that automatization of telephone traffic has reached a very advanced stage all over the world there is still a growing market for small PMBXs. Ericsson has therefore developed a PMBX system, ADD 20, for 10 and 20 extensions.

The author discusses the demands made on small PMBXs, describes the mechanical and electrical design of the new system and the facilities and functions it offers.

UDC 621.395.23

The PMBX system ADD 20 is intended for small hotels, hospitals, department stores, offices etc. where personal service and easy communication is very important. The opration is easy and uncomplicated for the users of the extensions, since it is only necessary to lift the receiver in order to reach the operator. For outgoing calls the extensions can request an external line and then just dial the wanted number.

ADD 20 is compact, space saving and has all the equipment neatly covered. The modern design of the PMBX means that it fits well into most environments.

The electronic PMBX ADD 20 is available in two sizes:

- ADD 20101 for 10 extensions, three trunk lines and a maximum of five simultaneous calls, fig. 1

Fig. 1
Electronic PMBX ADD 20101 for ten extensions, three trunk lines and five simultaneous calls

Fig. 2
Electronic PMBX ADD 20201 for twenty extensions, five trunk lines and seven simultaneous calls

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- ADD 20201 for 20 extensions, five trunk lines and a maximum of seven simultaneous calls, fig. 2.

The call connections are set up by means of a push-button panel, where the lines and switching possibilities are arranged in a coordinate system. A depressed button is indicated through the button top. The other optical signals, calling signal, clearing signal, engaged switching row etc., are indicated by light emitting diods in different colours.

ADD 20 can interwork with all types of manual and automatic public telephone systems. The PMBX can also be arranged for connection to manul local battery exchanges by equipping it with auxiliary line relay sets.

All lines are connected to screw clips in a wall-mounted connection box, which is connected to the PMBX via a plug-in cable.

Mechanical construction

The PMBX is very compact, being built up of reed relays, miniature relays, push-button switches and electronic components, and can be placed on an ordinary desk. The standard version of the PMBX is built up on three printed boards, which are connected together via a wired unit, to which the boards are plugged in, and also via connectors between the boards. If call secrecy is required, a fourth printed board assembly is connected into the PMBX.

When designing the PMBX great consideration was paid to the ergonomic aspects, such as the angle of the push-button panel to the horizontal plane, the colouring and the relative positions of the equipment in the panel.

The push-button switches are mounted between aluminium profiles, which take the load off the printed board assembly and protects the push-buttons against heavy objects, fig. 3, right.

Each push-button contains an indicator that shows when the button is depressed. Each button top consists of a lens, which, when the button is depressed, is pushed down towards a base with two yellow sections. The sections will then be visible through the lens as two stripes. Fig. 3 illustrates how the push-button indicator works, showing the button in the neutral position (left) and depressed (centre).

Electrical design

The electronic PMBX ADD 20 is dimensioned in accordance with the same guidelines that apply for all Ericsson's exchanges as regards reliability and environmental requirements.

Some examples of the interesting circuit designs used in ADD 20 are the call circuits for trunk lines and extension.

Trunk line
The call circuit consists of two optocouplers for sensing the ringing signal

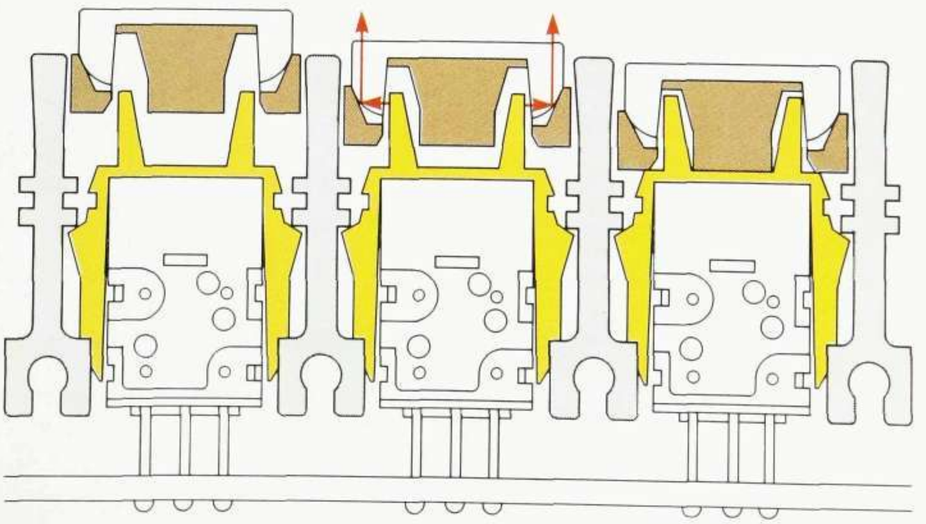

Fig. 3
Right:
The push-button switches are mounted between aluminium profiles, which provide protection if a heavy object should inadvertently be placed on the PMBX panel

Left:
The push-button switch in the neutral position. The edge of the lens that forms the top of the button is above the yellow stripes in the switch

Centre:
The push-button depressed. The yellow sections are visible throug the lens as two yellow stripes in the button

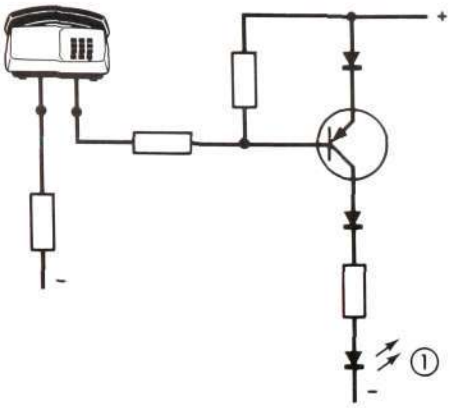

Fig. 5
Call circuit for an extension
1 Calling indicator

from the trunk line, fig. 4. Opto-couplers have been used in order to ensure galvanic isolation between the line and the indicator circuit. This has the following advantages:
- The attenuation introduced by the PMBX is negligible
- It is possible to test whether a call is finished by testing the current through the opto-coupler.

Extension line
The call circuit consists of a transistor which is controlled by the d.c. loop that is formed through the telephone when the extension handset is lifted, fig. 5. Negative polarity is connected via the telephone to the transistor, which then starts to conduct. The diode connected to the emitter protects the transistor against positive overvoltages on the a-wire. The advantage of an electronic call circuit is that the light emitting diod which indicates a call has the same light intensity regardless of the length of the line.

Secrecy board
ADD 20 can be supplemented by an extra printed board assembly if call secrecy is required. An intermittent warning tone is then sent out over any established call connection if the operator enters or connects in a third party on the line.

Facilities
ADD 20 is equipped with all facilities and functions required by a modern PMBX. In the fact panel on the opposite page the facilities and functions are divided into extension facilities, operator facilities and general functions.

Accessories

Power supply
The electronic equipment in the PMBX puts certain requirements on the power supply. The following alternatives are recommended:
1. Battery 20 Ah with an automatic charging rectifier BMJ 21101/1
2. Battery eliminator BMN 2122.

Telephone sets
The design of the superior exchange decides whether the extensions to the PMBX are to have a dial or a push-button set. Telephone sets without dial or push-buttons can also be used since the operator can provide assistance.

Connection boxes
Extensions, trunk lines and the power supply are connected to the PMBX via a 20 or 30-pair cable from the wall-mounted connection box.

Line protection
It is recommended that trunk lines and

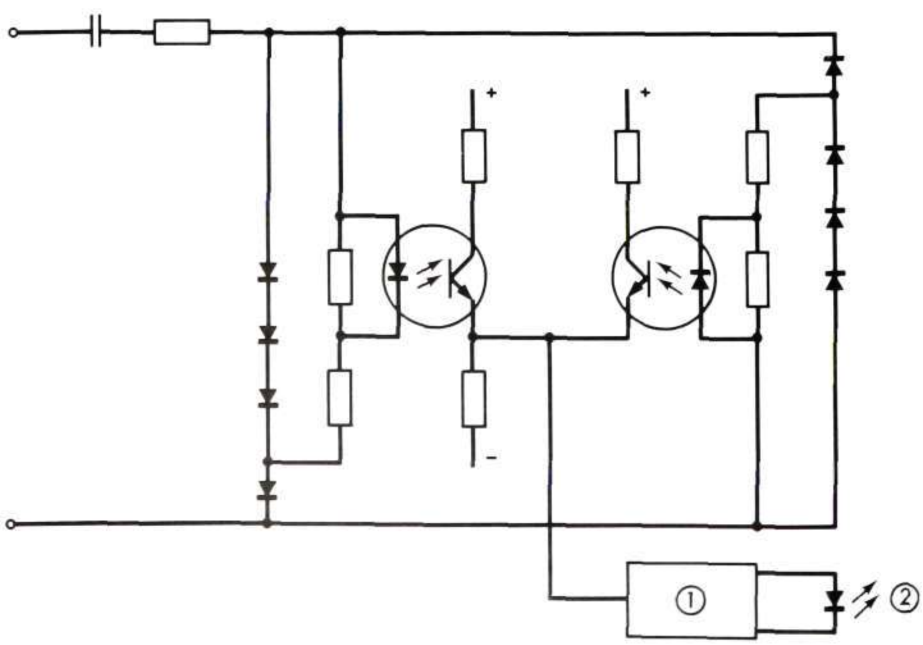

Fig. 4
Call circuit for a trunk line
1 Delay circuit
2 Call indicator

Fact panel

Extension facilities

- Automatic disconnection of the ringing signal when the extension answers
- Throug-dialling, which makes it possible to dial calls without the aid of the operator
- Automatic holding of the trunk line for incoming external calls
- Transfer of incoming external calls to the operator or another extension.

Operator facilities

- Push-button set for dialling
- Splitting key permits the operator to talk to any party while the other is disconnected
- Possibility of answering new calls on the trunk lines even if all call circuits are occupied
- Night service connection of the trunk lines to arbitrary extensions
- Light emitting diode for indicating that a ringing signal is sent out
- Supervisory circuit with a light emitting diode for calling and clerаring signals
- A discrete buzzing signal is heard when the supervisory circuit is activated, but it is not repeated for new calling and clerаring signals until the operator has cleared the previous connection
- The buzzer can be swithed to give a stronger, continuous signal in order to draw the operator's attention if she has to leave her position
- Light emitting diodes that indicate engaged switching rows
- Ringing towards the A-extension.

General functions

- A power break does not affect calls in progress on the external lines
- No power consumption when the PMBX is connected up for night service
- Two-wire connection of all lines
- Transistorized ringing generator
- Local battery type trunk lines can be connected with the aid of auxiliary equipment
- Call secrecy can be guaranteed by arranging for a warning tone to be sent to the callers when the operator connects in on a circuit. This facility requires extra equipment.

other external lines connected to the PMBX are equipped with overvoltage protection, particularly in areas where thunderstorms are frequent.

Call meters

ADD 20 can be equipped with call meters on the trunk lines. The public exchange must then be equipped to give subscriber's call-fee meter pulses having a frequency of 50 Hz, 12 kHz or 16 kHz.

Local battery trunk lines

Additional equipment can be provided for connecting trunk lines to local battery exchanges.

Technical data

		ADD 20101	ADD 20201
Capacity			
Extensions		10	20
Trunk lines		3	5
Simultaneous calls		5	7
Dimensions			
Width	mm	535	750
Height	mm	149	158
Depth	mm	308	343
Weight	kg	12.0	16.2
Electrical data			
Operating voltage	V DC		24 ± 4
Loop resistanse for extensions, including the telephone set	ohms		700
Leakage resistance, minimum	ohms		15.000
The data for the trunk lines are dependent on the limit values for the public exchange.			
Current consumption			
Internal call	mA		50
Call between operator and an extension	mA		130
External call	mA		0
Ringing towards an extension	mA		110

Microcomputer Controlled Interlocking System

Niels Siggaard and L. Norbert Sørensen

The Danish company, Dansk Signal Industri A/S, a member of the Ericsson Group, has developed a microcomputer controlled interlocking system. The system was developed in collaboration with the Danish State Railways (DSB), and the first system is now in operation on the Vejle–Holstebro line in Jutland. The authors outline the circumstances that have made it possible to use microcomputers in interlocking systems and describe the structure and functions of the system. They also describe briefly how it is possible to plan and test the system using a general purpose computer.

UDK 656.25:681.31-181.4

Interlocking systems are used on railways to ensure the safety of train movements. The primary aim is to prevent collisions and derailments. Another purpose of the signalling systems, of which the interlocking system forms part, is to ensure the traffic flow in accordance with the timetable and to permit easy operation at the lowest overall cost.

For more than 40 years safety relays have been the predominant component in railway fail-safe systems. Many attempts have been made to replace the clumsy and expensive safety relay with electronic components, but the cost has previously been too high. The increasing use of microcomputers has led to a drastic price reduction as a result of mass production. This has made the introduction of solid state technology into railway signalling attractive.

Fig. 1
Local control of a station with the aid of a colour VDU and an alphanumeric keyboard

Dansk Signal Industri A/S has developed a microcomputer controlled interlocking system, which was first commissioned in December 1980. Today the system is in operation at 14 stations in Denmark ranging from 20 devices (signals, point machines etc.) to more than 170 per station.

Development of the new interlocking system began in 1976, initiated by the need to modernize signalling on the line between Vejle and Holstebro in Jutland, so as to obtain reliable track circuit detection from a new generation of lightweight trains introduced by the Danish State Railways.

The interlocking function is computer controlled and relays are only used to interface the track devices. The computer processes the interlocking functions by means of two program systems which are independent of each other. The devices communicate continuously with the interlocking computer via concentrator computers common to several devices.

With the microcomputer interlocking, most of the wired logic specific to one installation is replaced by stored data. As manual preparation of these data would be as cumbersome as the planning and implementation of traditional installations, an off-line support program has been developed. It is run on a general purpose computer to produce the engineering and interlocking data needed. The resulting outputs is data ready for direct loading into the computers of the interlocking installations, with printouts to be read as signal and route control tables.

The dispatcher's control of the interlocking system at each station is carried out either locally or remotely (CTC) through microcomputerized interfaces. The equipment for local control consists of a colour video display unit (VDU) showing the track diagram with an indication of the actual status of all track devices, including track circuit occupancy and route setting. Provision has been made, especially at small stations, for the alternative use of a local control panel with keys and lamps.

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Dansk Signal Industri A/S

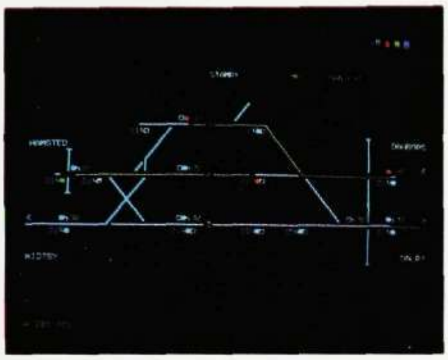


Fig. 2
Track-diagram, train routes, train movements etc. are presented on the VDU

Fig. 3
Block diagram of interlocking system JZSD 770

Dispatcher's equipment
Keyboard and semigraphic colour video display unit with 80 characters per line in a 48-line format. Eight different colours are used and each character is displayed by means of an 8x6 matrix

Alternative train dispatcher's equipment for small stations
Panel with lamps for indications and keys for setting of train routes

Computer CAPP for local control
Microcomputer Intel 8085 with 56 kbit memory is used for local control. The computer controls the exchange of information between the dispatcher's equipment and the interlocking computer. It records alarms and is also used for functional testing of the VDU

Computer FU for remote control
Microcomputer Intel 8085 is also used for remote control. This computer controls the exchange of information between the interlocking computer and the remote control system

Computer SID for interlocking
The Ericsson computer APN 163 with microprocessor circuits AMD 2901, 64 k word memory and 16-bit word length is used for the interlocking

Computer KC for concentrator function
Microcomputer Intel 8085 with 2.25 kbit memory is used for the concentrators. KC controls the exchange of information between the relay equipment and the interlocking computer. A maximum of 31 controlled devices and 28 track circuits can be connected to each concentrator

Relay sets RS
The circuits with safety relays are mounted in standard relay sets. The track circuits have free-wired relays connected directly to KC

Track devices
The track devices comprise different types of signals, points, track circuits, automatic blocks, level crossing plants etc.

System JZSD 770

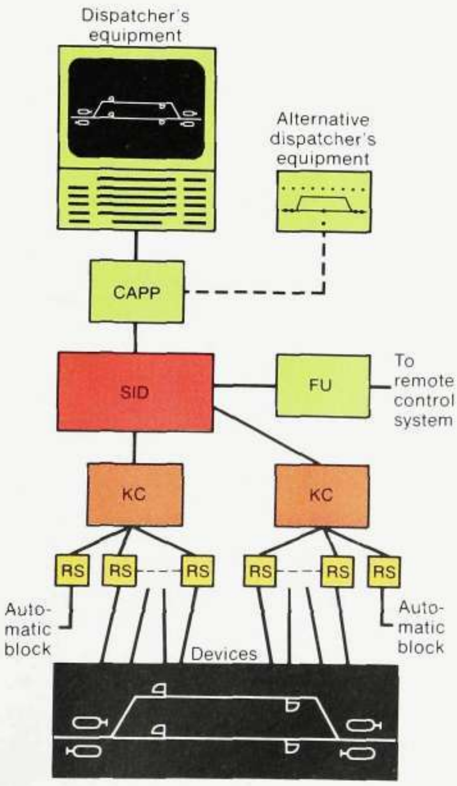
Interlocking system JZSD 770 consists of a number of subsystems, fig. 3:

- A subsystem for local control consisting of the VDU with keyboard and a computer CAPP
- A subsystem for remote control consisting of a computer FU
- An interlocking computer SID
- A number of concentrators each consisting of a concentrator computer KC with associated relay equipment RS.

The track devices (signals, points, track circuits etc.) are connected to the concentrators. Transmission of data between the different subsystems takes place via transmission links. The data messages are transmitted in serial form, and each message is supplemented with redundant information in order to ensure faile-safe function.

Subsystem for local control

The subsystem for local control consists of a VDU with a standard keyboard and a microcomputer, CAPP, which handles the exchange of commands between the the dispatcher's equipment and the interlocking computer. CAPP processes and stores commands from the keyboard and transmits them to the interlocking computer, SID.



Status information from the devices is transmitted continuously in the opposite direction, from SID to CAPP. This information is processed in CAPP and then used for updating the information displayed on the screen.

Alarms received from the interlocking computer, for example burn-out of a filament in a signal lamp, and internal alarms from CAPP are recorded and stored for display on the screen and for later printout. All control commands and changes in the status of devices are also logged continuously, and the overall state of the system is logged at specific time intervals. Printout of logged information is used in connection with investigation of accidents and for system fault finding.

The local control computer contains facilities for adjusting and testing the VDU. The presentation on the display is refreshed alternately from the two data systems A and B. The computer continuously monitors receipt of status information from all track devices.

Subsystem for remote control

A microcomputer, FU, is used as interface to a remote control system. This computer handles the exchange of information between the interlocking computer and the remote control system. Commands are received, processed and transmitted to the interlocking computer. In the opposite direction there is a continuous flow of indications, which are processed and stored before being transmitted via the remote control system.

Interlocking subsystem

All information about the geographical layout of the station is stored in the interlocking computer, together with all possible train routes and the corresponding positions of all track devices. The computer processes commands from the control computer such as:

Establish train route from signal no. 01 to signal no. 02.

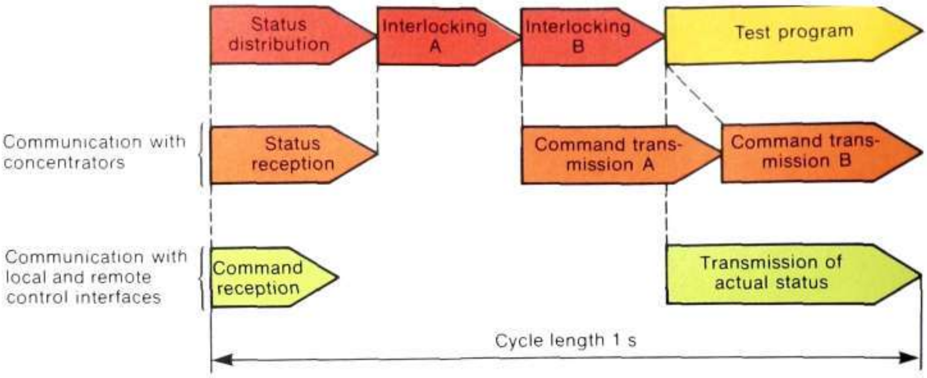


Fig. 5
Program execution in an interlocking computer

Information received from the devices via the concentrators is simultaneously processed – status information such as:
Signal no. 01 shows stop.

The interlocking computer performs a number of tasks such as verification of commands, prevention of set-up of conflicting routes, setting train routes with the associated operation of points, signals etc.

Concentrator subsystem

Each concentrator consists of a computer with associated relay equipment controlling the function of the devices. Commands from the interlocking computer are stored in the concentrator computer. The commands are checked with respect to correct concentrator address (for example) and then fed to the relay equipment.

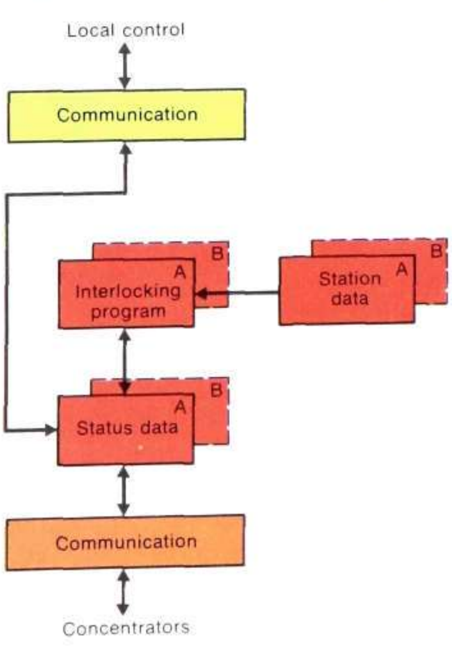
The concentrator computer continuously records the status of all connected devices during each operating cycle and transmits the corresponding data to the interlocking computer. Changes of the status of the track circuits are recorded more often, and any occupancy longer than 30 ms is detected.

The relay equipment consists of relay sets for signals, points and other devices, and track relays for the track circuits. The relays used are safety relays. Each device normally has a relay set of its own.

Fail-safe interlocking

Fail-safe interlocking is obtained by means of an interlocking process (A118) proposed by ORE (Office de Recherches et d'Essais) of UICF (Union International des Chemins de Fer). The interlocking process is carried out by a single computer with two completely independent programs, A and B, fig. 4.

Fig. 4
Software structure in the interlocking computer



The system concept adopted is that the whole interlocking process be divided into two separate channels, A and B. The diversity is in space and in time as well as in coding. Track devices cannot be activated unless the output from the two channels agree, which is determined by fail-safe comparison in the relay sets.

A common functional specification forms the basis of the programs, and the coding as well as the testing of these programs uses separate data sets (also designated A and B) in which corresponding data bits are mutually inverted and the address bits mutually reversed.

The programs contain the general interlocking rules applicable for any installation, whereas the data define characteristics specific to each station.

Interlocking programs A and B are executed sequentially once every operating cycle of 1 s. Based on updated information on the actual status of track devices, and depending on the dispatcher's commands, and routes already established, programs A and B process the respective A and B commands which are then transmitted to the relay sets controlling the points and signals in question. If commands for a device processed during a cycle are in disagreement, transmission is blocked by the computer. The program execution of the interlocking computer is shown in fig. 5.

The relay set for a particular device performs a fail-safe comparison between the A and B commands transmitted to it at any time. If the commands are in agreement and legal, the relay set activates the device. If the commands are either illegal or in disagreement, the relay set shifts to a locked state keeping the device in a restrictive position.

For a signal to maintain a "proceed" aspect, this activation must take place every second cycle. Lack of command information for more than 2.5 s will automatically switch the signal to "stop".

In a similar way every relay set generates A and B indications representing the actual status of the device to which they correspond. All devices, including the track circuits, are scanned cyclically by the computer. The A and B indications must agree if they are to be accepted by the computer.

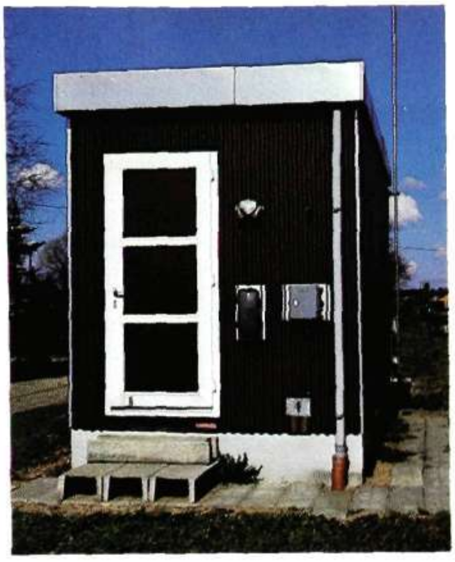


Fig. 7
Cabin in the station area containing the equipment for one track sector

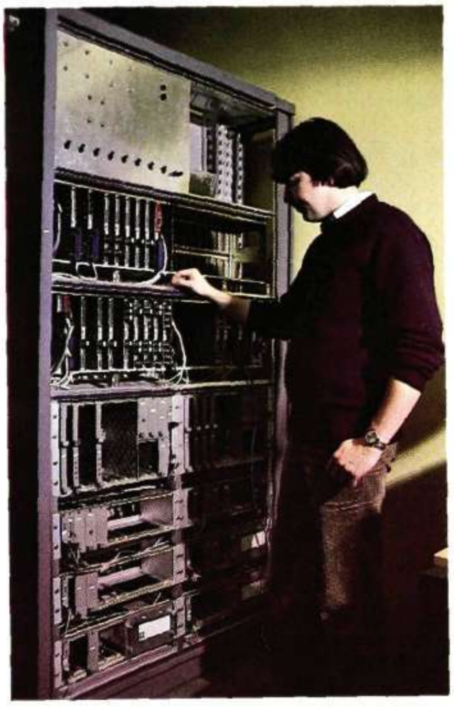


Fig. 8
Interior from a cabin showing SID, CAPP, FU, and a number of KCs

Project planning

When planning a system for a station it is usually best to divide the station area into sectors in order to keep the cables between the devices and their relay sets short, fig. 6. Cable costs as well as the risk of malfunction due to induced electrical interference are thereby reduced.

A concentrator, controlled by a microcomputer, is installed together with the relay sets for each sector. It handles the exchange of commands and status information with the interlocking computer.

The equipment for each sector is placed in a cabin of the type shown in figs. 7 and 8. One of these cabins also contains a separate room for the dispatcher's control equipment.

Control facilities

The devices at a station can be controlled either locally or remotely from a remote control centre.

For local control a colour VDU is used, showing the track diagram with an indication of the actual status of all track devices, including track circuit occupancy and route settings. The VDU echoes commands fed from an alphanumeric keyboard. Stored commands and error messages can be displayed on the VDU by means of special commands.

Research studies as well as experience gained in the field have proven that colour displays convey the information more efficiently than any other visual method. Furthermore, colour dispalys lead to easier information recognition, thus permitting faster dispatcher response.

The dispatcher uses a standard type keyboard. Commands are given in the form of abbreviations, consisting of letters and figures, and built up in a standard pattern that is easy to learn. The symbol for the selected device flashes when the command is entered, so that the dispatcher can immediately check that the correct track device is addressed.

The system automatically checks that a command does not contain any syntax errors. The number of characters must be correct, the command must be correct, the addressed device must exist etc. Each command is also checked in the interlocking computer before it is accepted. A red star is displayed on the screen and an acoustic alarm is given if a faulty command is entered.

It is possible to edit a command during entering by erasing either the last character or the whole line. When a

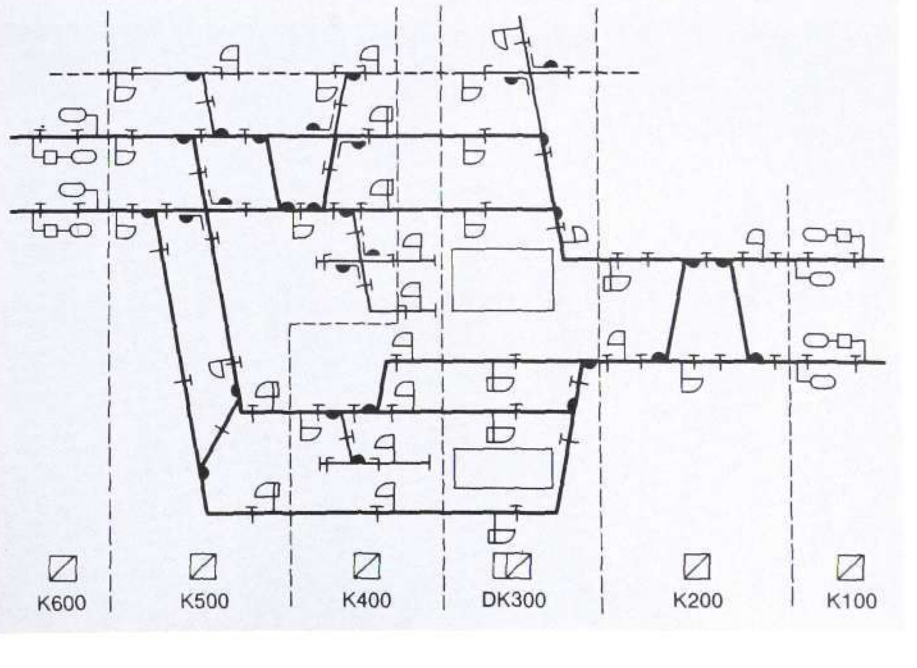


Fig. 6
The track network at Herning station has been divided into six sectors, each with a cabin for the concentrator, relay sets, track relays, power supply etc.

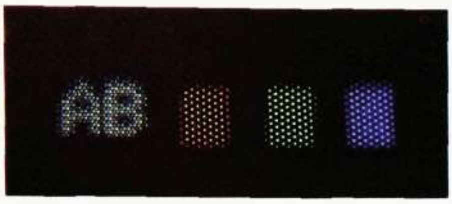


Fig. 9
Shifting between the letters A and B on the VDU indicates that the presentation is refreshed from both data systems. The red, green and blue colour symbols are used to check correct colour information

correct command has been entered it will be executed when the key "Carriage return" is depressed. If a route cannot be established immediately, the command is stored until all conditions are met.

The system also allows the dispatcher to enter commands in advance. Priority rules and input order determine the order in which the commands are executed. It is also possible to store previously prepared command sequences, which can be released for execution by means of a single command.

In order to ensure high reliability the indications on the display screen are updated alternately from the two independent computer programs. In addition there is continuous monitoring to ensure that indications are obtained from all devices. Symbols for devices of great importance to safety, such as signals and points, are always indicated by at least two characters.

A number of check indications are continuously displayd on the screen. In the top right-hand corner three symbols, one red, one green and one blue, show whether the colours are properly adjusted. Next to them another indication shows alternately A and B, proving that the display is being updated from both computer programs, fig. 9. In the top left-hand corner a flashing yellow star is shown as an indication that the transmission between the interlocking computer and the control computer functions properly.

Different types of train routes and train movements are indicated by different colours. For example, shunting operations have a special colour indication. As regards the signals, the display indicates which signal aspect they show, and for the points it indicates both their positon and whether they are controlled locally or centrally. The significance of the various colours is:

Turquoise	Neutral devices (not part of any train route)
Blue, flashing	Faulty device
Green	Established main route
Green, broken line	Overlap route
Yellow	Established shunting route
Red	Occupied track circuit
White	Signal aspects for shunting
Lilac	Point released for local control
Black	Key-locked point and line block.

System planning

In a microcomputer controlled interlocking system all information that is individual to a specific system (station) is stored in the computer memory.

A project planning system has been developed which, on the basis of certain fundamental conditions, generates all information that is required for a station. The program is run on a general purpose computer.

The project planning system is of the interactive type and based on the geographical concept. An operator feeds the computer with information regarding the track network, the type and position of signals and points, signalling conditions, release conditions for train routes, overlaps and suitable positions for the concentrators and relay sets in a system.

On the basis of this information the project planning system provides a plan of the individual system data, i.e. the data that determine the safety logic in

Fig. 10
Control of 66 level crossing plants on the line is integrated in the interlocking functions

the interlocking subsystem and the command and indication data for the local control equipment.

These individual system data are fed into test equipment having the same computer configuration as is to be used in the final interlocking system. The traffic and control functions of the whole system can then be tested by simulating the functions of devices, device relay sets and track circuits. All this can be done long before the real planning of the installation work starts.

When the characteristics of the system have thus been tested and accepted, the individual system data are fed into the computer memories.

All equipment for a station can be supplied mounted and installed in prefabricated cabins. Before the equipment is delivered it undergoes a computer-controlled test procedure with artificial devices. The function of the concentrators and devices, and the rack installation are then tested before the interlocking computer is connected.

The only testing required when the system is installed consists of checking that the devices are connected correctly, adjusting the signal lamp currents and possibly also carrying out a functional test with the devices connected up, after which the system can be put into operation.

If an interlocking installation has to be modified at a later date, the operation of the modified system can be tested in advance in the test equipment. In this way the time required for the modification of the equipment in the field can be considerably reduced.

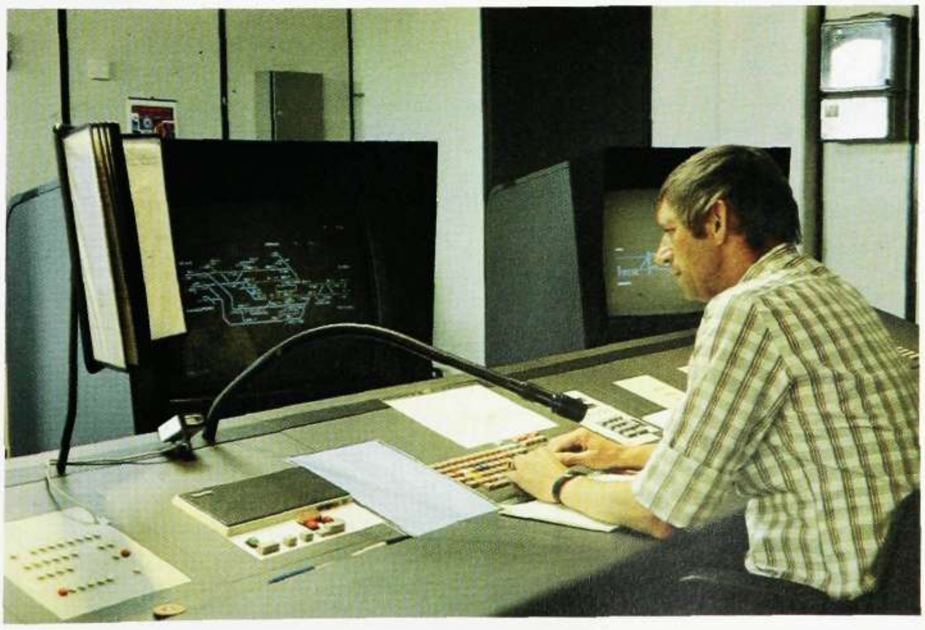
Summary

The use of microcomputers in interlocking systems has opened up new possiblities. The new technology has many advantages over the old:
- greater immunity against electrical interference
- improved dispatcher control facilities
- automatic event logging
- project planning time greatly reduced
- simple and efficient testing of installations
- functional test off-site in advance of delivery
- modifications easy to carry out
- investment costs reduced.

The microcomputer controlled interlocking systems that are now in operation in 14 Danish stations have fulfilled all expectations. The experience gained hitherto proves that this new generation of interlocking systems is reliable and will be very attractive also to Railway Administrations outside Denmark.

The success of the system has further been confirmed by the fact that the Danish State Railways have placed another order for delivery of similar equipment to be installed at seven stations on the line Roskilde–Køge–Naestved in Zealand.

Fig. 11
Herning is equipped for two control positions. One is used for local control of Herning station. The other position is used for remote control of Holstebro station



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Emergency Telephone System SOS 512

Ilkka Jäntti

Ericsson's Finnish subsidiary has developed an emergency telephone system with such a high degree of flexibility that it can also be used as a general regional alarm system.
The author discusses the reasons for equipping roads with emergency telephones and describes the structure and function of the new system.

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Fig. 1
The shape, colour and reflecting labels of the emergency telephone column makes it clearly visible

During the years 1972–1979 the road and telecommunication authorities in Finland carried out an investigation with the aim of determining the need for alarm telephones along public roads, and the properties an emergency telephone system should have.

The public road network in Finland comprises 74,000 km of roads and is frequented by 1.8 million vehicles.

The most important reason for installing emergency telephones is that they can help to save lives and mitigate the effects of personal injuries because expert aid can be summoned quickly to traffic accidents. Speedy aid at the site of an accident is more important than fast transport to the hospital. According to doctors the number of fatalities at traffic accidents could be reduced by 20% if first aid could be given immediately after an accident.

Although the value of human lives or human suffering cannot be assessed in terms of money it is possible to determine where and under what circumstances it is profitable to install emergency telephones. It can be estimated how much society can save if a death or a permanent handicap can be avoided and a person can remain an active member of society. These savings can be compared with the cost of installing, operating and maintaining an emergency telephone system.

On the basis of such calculations the Finnish invesigation recommended that 2,000 km of the major roads should be equipped with emergency telephones without delay, and a further 3,000 km should be provided with emergency telephones within a few years. According to the investigation the value of these telephones is 2.5 times their cost.

During the investigation a prototype of the emergency telephone system was tested in Kuusamo in north-east Finland. It proved to be very successful under very difficult climatic conditions. The experience gained during this trial has been used to improve the design and software of the system further. This has resulted in a very advanced and reliable emergency telephone system, SOS 512.

One of the main aims when designing SOS 512 was to obtain the greatest possible degree of flexibility. The system can use circuits in the switched network as well as point-to-point circuits of its own. The latter facility means that the system offers an economical solution even for areas without any previous telecommunication facilities.

SOS 512 has not been designed solely as a road emergency telephone system. The system also meets the need for alarm transmission and communication in many other areas, such as harbours, railway stations, oil refineries and oil fields. The system can also receive calls from subscribers in the public telephone network. This makes it possible to use the alarm centre of the emergency telephone system as a general alarm centre for a region. This centre can then connect all alarm calls to the correct emergency service unit in the case of accidents and other emergency situations.

The main parts of the system

The system consists of
- emergency telephones
- concentrators placed between the emergency telephones and the receiving unit
- the receiving unit and operator's unit in the alarm centre.

Emergency telephones

The emergency telephones, which are loudspeaking telephones, consist of the following parts:
- column
- telephone cassette
- line connection unit.

The columns are approximately 2 m high. In order to make them as clearly visible as possible they are orange and tapering downwards. The columns are also equipped with light-reflecting symbols

ILKKA JANTTI
Oy LM Ericsson Ab
Finland

The telephone cassettes are replaceable, which makes for quick and easy maintenance. The sending and receiving sensitivity for each telephone are adjusted in the line connection unit, and therefore remain unaltered when the telephone cassette is changed.

Concentrators

The concentrators used when the circuits are connected up via the public telephone network are designated remote concentrators, CNR-R. The concentrators in a separate network are designated local concentrators, CNR-L. Remote concentrators are installed in telephone exchanges, where they occupy a subscriber position each. Local concentrators are installed either in connection with the receiving unit or, in order to reduce the cable cost, further out in the network.

A call from an emergency telephone automatically initiates a call from the connected remote concentrator to the public telephone network. When the dialling tone is detected the telephone number of the receiving unit is transmitted. If the number is free, the call is connected up and the caller, who has heard a tone signal during the connection procedure, is through to the alarm centre.

The receiving unit is connected to the telephone network via at least two and usually more subscriber positions. If the remote concentrator receives the busy tone or a congestion signal, the call attempt will be repeated to the corresponding numbers a preset number of times. If, in exceptional cases, the call cannot be set up in spite of these repeated attempts, the remote concentrator is automatically returned to the idle state after which the user can initiate a new call.

The local concentrators are connected to the receiving unit via point-to-point circuits, which makes for quick connection of calls from the emergency telephones.

Receiving unit and operator's unit

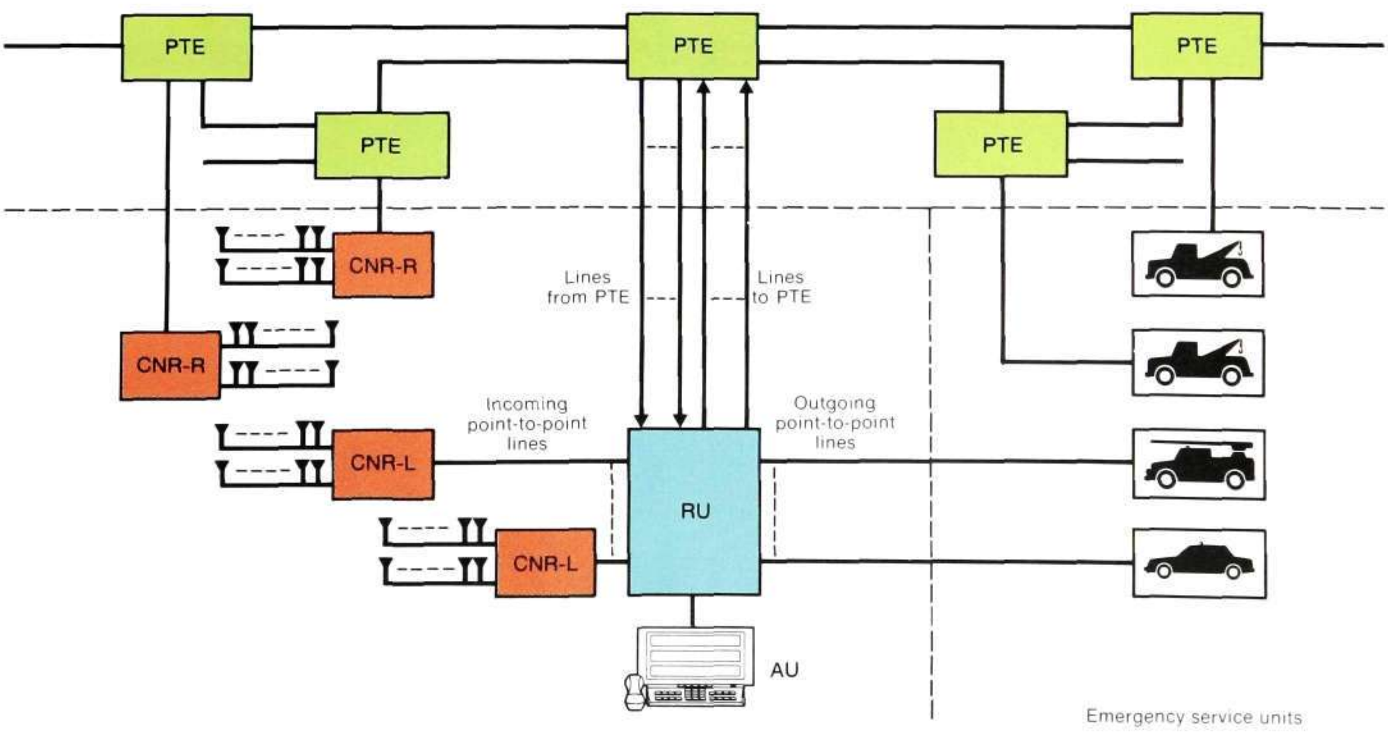
The receiving unit and operator's unit are placed in the alarm centre.

The lines from the separte emergency telephone network and the lines from the public telephone network are connected to the receiving unit (RU). The receiving unit also controls outgoing calls from the operator's unit (AU) to the emergency telephones and subscribers in the public telephone network.

The operator's unit is used to answer

Fig. 2
Block digram of emergency telephone system SOS 512 and interworking units

PTE	Public telephone exchange
CNR-R	Remote concentrators connected via the public network
CNR-L	Local concentrators connected to the receiving unit RU via point-to-point lines
RU	Receiving unit
AU	Operator's unit



incoming calls and, when necessary, to forward them to other emergency service units, for example the police or fire brigade, via point-to-point circuits or via the public telephone network. The identity of a calling emergency telephone is displayed in a digit indicator in the operator's unit.

Function

The loudspeaking emergency telephone is very easy to use. The caller only has to turn the handle on the front and wait for an answer. A tone signal is received as a confirmation that the call is being set up.

The call from the emergency telephone is indicated in the operator's unit both optically and acoustically.

When the operator depresses the answering button a two-way speech connection is set up to the emergency telephone, and the number of the telephone is displayed on the digit indicator. The operator can then find out what type of aid is required, and connect up the call to the unit in question, i.e. police, fire brigade, ambulance, garage etc.

If the call cannot be cleared immediately it can be switched to camp-on. This makes it possible to supervise several calls simultaneously. The operator's unit shows which calls are in the camp-on state, to what service unit they are waiting to be forwarded and which have had the forward connection established. The number of calls that have not yet been answered is also indicated.

The operator can listen in on or enter a forwarded call. In both cases a warning tone is heard. Calls between emergency telephones and the operator's unit can be recorded on tape, but not forwarded calls. Emergency telephone calls are disconnected by the operator.

The emergency telephones can also be called individually from the operator's unit.

Cable network

The emergency telephones can be connected either over individual lines or with several telephones in parallel on the same line. Individually connected telephones require one wire pair per telephone, whereas a line with telephones in parallel requires two wire pairs.

A loaded quad with a maximum loop resistance of 2,000 ohms is normally used. It permits 44 telephones to be

Fig. 3
The operator's unit used for receiving emergency calls has a simple construction and is easy to handle

connected in parallel on one line. The emergency telephones are usually placed in pairs along the road, with a distance of 2 kilometres between the pairs.

Mechanical construction

The columns are made of glass fibre reinforced plastic laminate further reinforced by four steel bars. They are mounted on a concrete base and dimensioned to withstand the pressure exerted by snow and ice when the roads are ploughed. The material and finish have been chosen so that the columns can withstand high and low temperatures, moisture, sunshine and normal cleaning. The low weight of the columns means that very little damage is caused to any vehicle that hits a column. In very difficult environments the electronic part of the telephone unit can be installed in a waterproof container in the ground.

Construction practice BYB is used for the concentrator and the receiving unit. The concentrator consists of one magazine. It can be mounted in the same cabinet as the receiving unit unless the concentrators are installed along the line. The receiving unit consists of between three and six magazines. Its equipment varies depending on the system structure and the number of telephones.

The operator's unit is built with push-button set and digit indicator in the same unit. The unit contains internal 5 V conversion and can therefore be placed at some distance from the receiving unit.

Service supervision and fault tracing

An emergency telephone system requires a high grade of service. This has been achieved partly by choosing components and a design that ensure high reliability and partly by designing the supervisory system to give immediate indication of disturbances and easy fault tracing.

The operation of the system is monitored continuously. The equipment at the alarm centre has its own test routine. An alarm is obtained immediately a fault occurs. The aim is to be able to track the fault down to a replaceable unit.

The emergency telephone functions for sending, receiving and signalling are automatically tested every ten minutes by the concentrator when in the idle state. The concentrator switches the

Fig. 4
The emergency telephone is easy to operate. The caller only has to turn the calling handle. When the operator answers the call a two-way loud-speaking connection is set up

telephone to the test state, and then transmits and detects test signals. If a fault is detected the concentrator transmits information regarding the type of fault and faulty unit to the receiving unit. This information is displayed on the digit indicator in the operator's unit.

The concentrators also establish contact with the receiving unit at regular intervals. If any such call from a concentrator is not forthcoming, the receiving unit tries to call the concentrator. If the call attempt fails, the concentrator is assumed to be faulty and an alarm is initiated. When a fault alarm is obtained, the operator can have the identity of the faulty unit displayed by depressing a button.

The operator can also test an emergency telephone by calling it from the operator's unit and listening to the traffic noise. However, the built-in automatic testing of the system is more comprehensive than such manual testing.

Fig. 5
The telephone cassettes are easily replaced, which simplifies fault clearing. The individual settings that specify the identity of the emergency telephone are made in the connection unit and are not affected when a cassette is changed

Summary

System SOS 512
- consists of emergency telephones, concentrators and receiving units with operator's units
- has high capacity and it is easy to connect in further receiving units
- has the emergency telephones connected to the receiving units via point-to-point cables or via the public telephone network
- can also receive emergency telephone calls from the public network
- permits forwarding of the emergency calls
- permits tape recording of the emergency calls.

The emergency telephones
- are loudspeaking telephones
- permit full duplex speech transmission
- can have 2-wire or 4-wire connection
- are powered via the speech wires
- permit parallel connection of up to 44 telephones on one 4-wire line
- are mounted in columns made of glass fibre reinforced plastic laminate
- have a replaceable cassette for the electronic part.

The concentrators
- have a capacity of 32 emergency telephones with 2-wire connection or 96 telephones with 4-wire connection
- have a line connected to a receiving unit or to the public telephone network
- test the telephones automatically
- are controlled by microprocessors
- are powered by 41 – 70 V
- are assembled in BYB magazines.

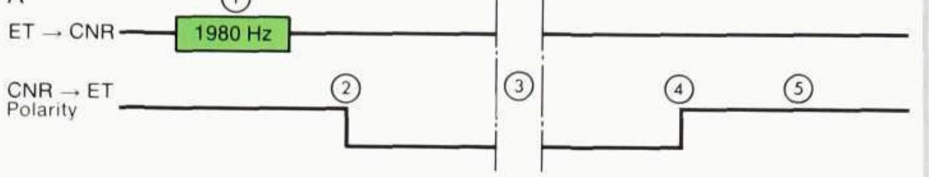
The receiving units
- have a capacity of 16 concentrators, 1536 emergency telephones and a maximum of 18 incoming and outgoing lines
- are controlled by a system of several microprocessors, in which the main processor is duplicated
- are powered by 41 – 70 V
- are assembled in BYB magazines.

Signalling in SOS 512

Call from a two-wire emergency telephone, ET-2W, to the concentrator, CNR, fig. A

1 Call from the emergency telephone
2 Acknowledgement by means of polarity reversal from the concentrator
3 Speech connection
4 Disconnection
5 The emergency telephone in the idle state

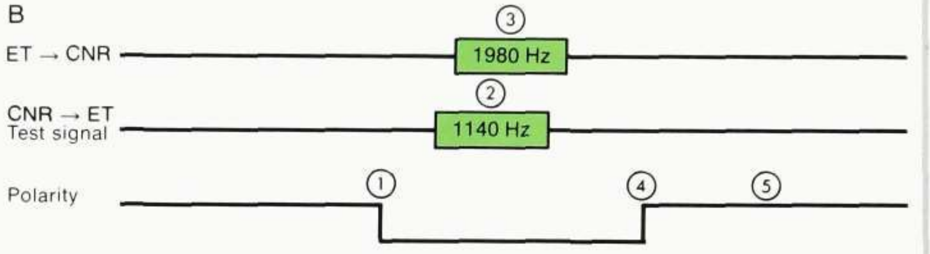
A



Automatic testing of a two-wire emergency telepnone, fig. B

1 A polarity reversal switches the telephone unit to the active state
2 Test signal to the emergency telephone
3 Acknowledgement from the telephone in the form of a call
4 Disconnection by means of polarity reversal
5 The emergency telephone in the idle state

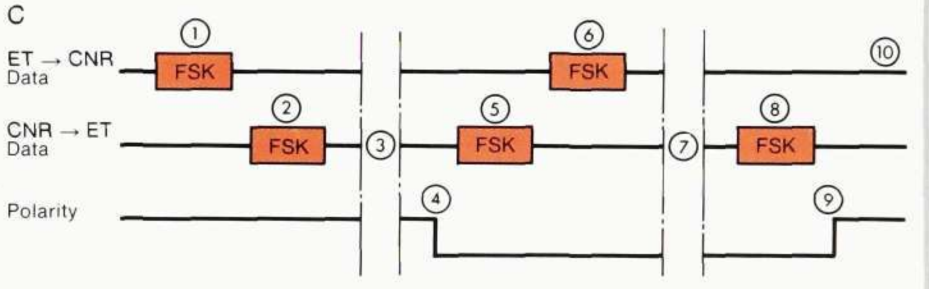
B



Call from a four-wire emergency telephone, ET-4W, to the concentrator, fig. C

1 Call from the emergency telephone
2 Acknowledgement from the concentrator, command for queueing state
3 The telephone in the queueing state
4 Polarity reversal to speech polarity
5 Command for speech state
6 Acknowledgement from the telephone
7 The emergency telephone in the speech state
8 Disconnection command
9 Polarity reversal to idle polarity
10 The emergency telephone in the idle state

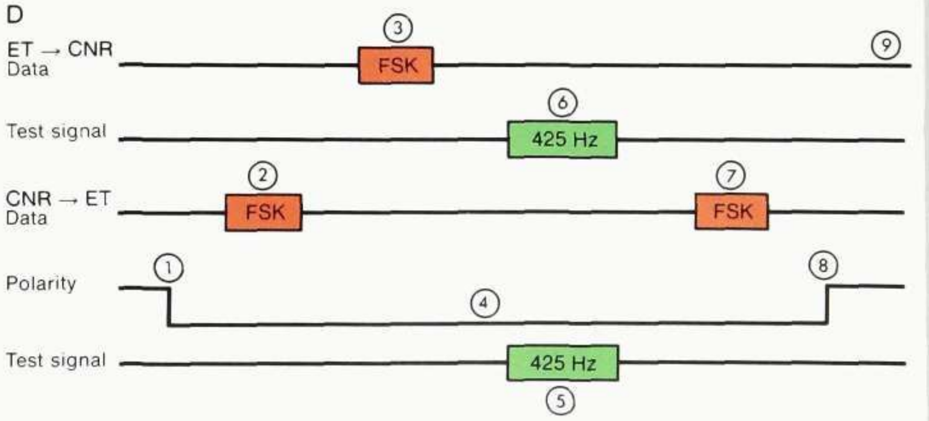
C



Automatic testing of a four-wire emergency telephone, fig. D

1 Polarity reversal to speech polarity
2 Test command to the telephone
3 Acknowledgement from the telephone in the form of a call
4 The emergency telephone in the test state
5 Test signal to the telephone
6 The test signal goes back to the concentrator via the speech path
7 Disconnection command
8 Polarity reversal to idle polarity
9 The emergency telephone in the idle state

D



Signalling format, fig. E

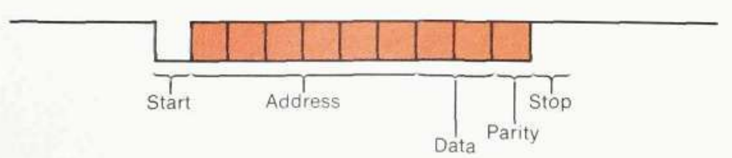
All signals between a four-wire emergency telephone and the concentrator have the format shown in fig. E.

Signalling between the concentrator and the receiving unit

A digital frequency shift code in accordance with CCITT Recommendation V.21 is used for the signalling between the concentrators and the receiving unit.

The signalling is carried out in semiduplex at a transmission rate of 100 or 300 bit/s with block synchronization. The length of the blocks is 50–100 bits and each block contains a 16-bit error detection field (CRC).

E



RIFA Capacitors for Telecommunication Electronics

Johnny Blocksjö and Arne Ström

RIFA's activities in the field of semiconductors have been described in a series of articles in Ericsson Review. This and a subsequent article describe RIFA's work in the field of capacitors.
This article first gives some general technical definitions concerning capacitors. RIFA's capacitor program is then presented, and the dielectrics used for the various products are discussed. The authors describe the choice of materials and the construction and characteristics of plastic film capacitors. Finally a summary is given of the main stages in the manufacture of such capacitors.

UDC 621.319.4-11:621.39

Fact panel

Capacitor capacitance
$C = \varepsilon A/d$
C is measured in farad (F)
$\varepsilon = \varepsilon_0\varepsilon_r$ is the dielectric constant
$\varepsilon_0 = 10^{-9}/36\pi$ F/m is the dielectric constant of vacuum
ε_r is the relative dielectric constant
A is the electrode surface area in m^2
d is the thickness of the dielectric in m

Capacitor charge
$Q = C \times U$
Q is measured in coulomb (C) or ampereseconds (As)
U is the voltage across the capacitor and is measured in volts (V)

Capacitor energy
$W = \frac{1}{2} CU^2$
W is measured in joule (J) or wattseconds (Ws)

Capacitor loss factor
$\tan\delta = R\omega C$
R is the equivalent series resistance o the capacitor
ω is the angular frequency in radians/s

The function of the capacitor

A capacitor is a component that consists of two metal electrodes separated by an insulating material (dielectric). A capacitor blocks direct current but lets alternative current through. This function is necessary in many electrical contexts, from power to microelectronics.

The capacitance of a capacitor is determined by the area of the metal electrodes and the thickness and dielectric constant of the dielectric. Details regarding quantities and relations are given in the fact panel to the left.

A capacitor can assimilate, store and give off an electric charge. The ability of the capacitor to assimilate charges is determined by its capacitance and the voltage across it.

The amount of energy a capacitor can store depends on its design. RIFA's product range contains capacitors whose maximum ability to store energy varies from parts of a microjoule to a few hundred joule, fig. 1.

The functional symbol for the capacitor shows an ideal capacitor. In reality a capacitor always has imperfections such as series inductance, L, and loss resistances, R_1 and R_2, as shown in the equivalent diagram in fig. 2.

The inductance can be reduced but not eliminated. At a certain frequency, f_o, the capacitive and inductive reactance are equal, i.e. $1/\omega_o C = \omega_o L$, where the angular frequency $\omega_o = 2\pi f_o$. At frequencies higher than f_o the inductive factor is therefore dominant.

R_1 represents the resistance of the capacitor leads and electrodes, polariza-

Fig. 3
A part of RIFA's range of capacitors

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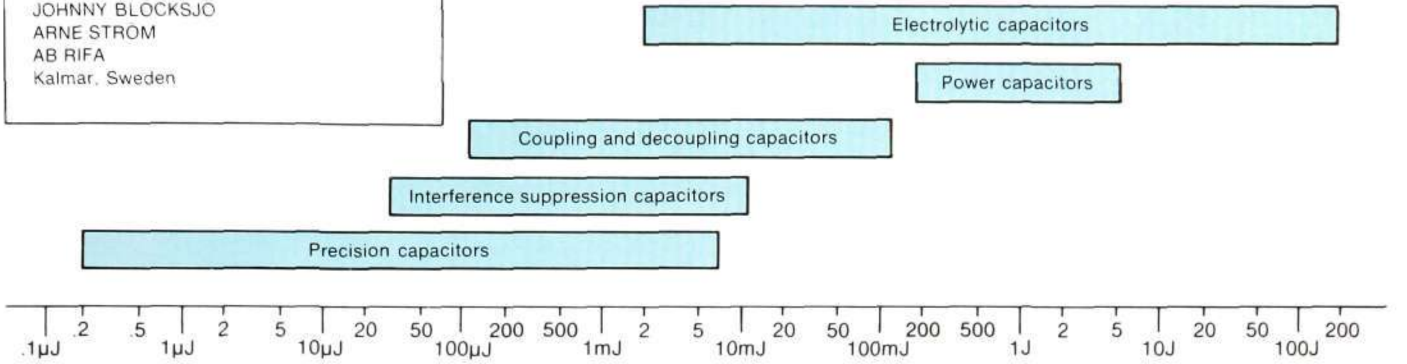


Fig. 1
The amount of energy stored in different types of capacitors manufactured by RIFA

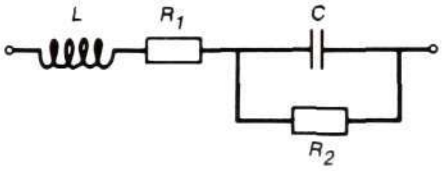


Fig. 2
The equivalent diagram for the capacitor

tion losses in the dielectric and the resistance of the electrolyte in electrolytic capacitors. R_2 determines the leakage current and is dependent on the insulation properties of the capacitor dielectric.

The values of R_1 and R_2 determine the capacitor losses, which ar dependent on temperature, frequency, voltage and capacitance, and which result in heating. Normally R_2 is much larger than R_1, which means that they can with good approximation be replaced by a single resistance, R, the equivalent series resistance of the capacitor.

The ratio between the equivalent series resistance of the capacitor, R, and its reactance, $1/\omega C$, is the loss factor of the capacitor and is designated $\tan\delta$.

RIFA's capacitor range

RIFA's production program for capacitors is divided into the following groups:
- precision capacitors, coupling and decoupling capacitors and pulse capacitors with radial leads in boxes or MINIPRINT encapsulations and with plastic film dielectrics
- interference suppression capacitors and RC units with radial leads in MINIPRINT encapsulations with impregnated paper dielectrics
- can type power capacitors. This group includes not only capacitors for power electronics but also motor capacitors and phase compensating capacitors for flourescent tubes. The dielectric used is plastic film or a mixed dielectric, i.e. plastic film and paper, together with impregnation

Fig. 4
Precision capacitors

	Poly-propylene	Poly-styrene	Poly-carbonate	Polyester*
Minimum thickness, μm	4	6	2	1.5
Maximum temperature, °C	100	85	125	125
ε_r at 1 kHz	2.2	2.4	2.9	3.1
$\tan\delta$ at 1 kHz, %	0.02	0.01	0.1	0.4

*) Polyethylene terephthalate

Table 1
Characteristics of some dielectrics used in plastic film capacitors

– electrolytic capacitors, both axial and can type, and with aluminium oxide dielectrics.

A common characteristic of the above dielectrics is that they can be used in wound capacitors. Winding is a traditional method which ensures a large surface area for the electrodes and thereby high capacitance for a small volume. The first group, RIFA's plastic film capacitors, is described below.

Plastic film capacitors

The group of plastic film capacitors comprises precision capacitors and coupling, decoupling and pulse capacitors.

These components are used for various purposes, such as filters, coupling and decoupling functions and time and pulse circuits. The environments in which the capacitors are used vary from airconditioned rooms through industrial environments with varying temperature and humidity to extreme environments with strong vibrations, e.g. fighter aircraft.

RIFA's range of plastic film capacitors has been planned to meet the requirements of different countries and different applications. It therefore includes products that meet DIN, IEC, CCTU and CECC standards. It also contains capacitors that have been designed to individual customer specifications.

Choice of materials

The choice of materials for capacitors is extremely important. The correct material has to be chosen not only for the dielectric and electrodes but also for the encapsulation and leads.

Dielectrics

The insulating material between the electrodes, the dielectric, is one of the factors that determine the characteristics of a capacitor. There are two main types of dielectrics, polar and nonpolar. The difference between them lies in their material structure. Polar materials normally have a higher relative dielectric constant (ε_r) and a higher loss factor (*tan*δ) than nonpolar. The polar materials are also more frequency and temperature dependent.

Table 1 shows the main characteristics of common dielectrics in plastic film capacitors.

The different dielectrics require different processing and handling in order to give the best possible result, and they are used individually or in combinations in different capacitor designs.

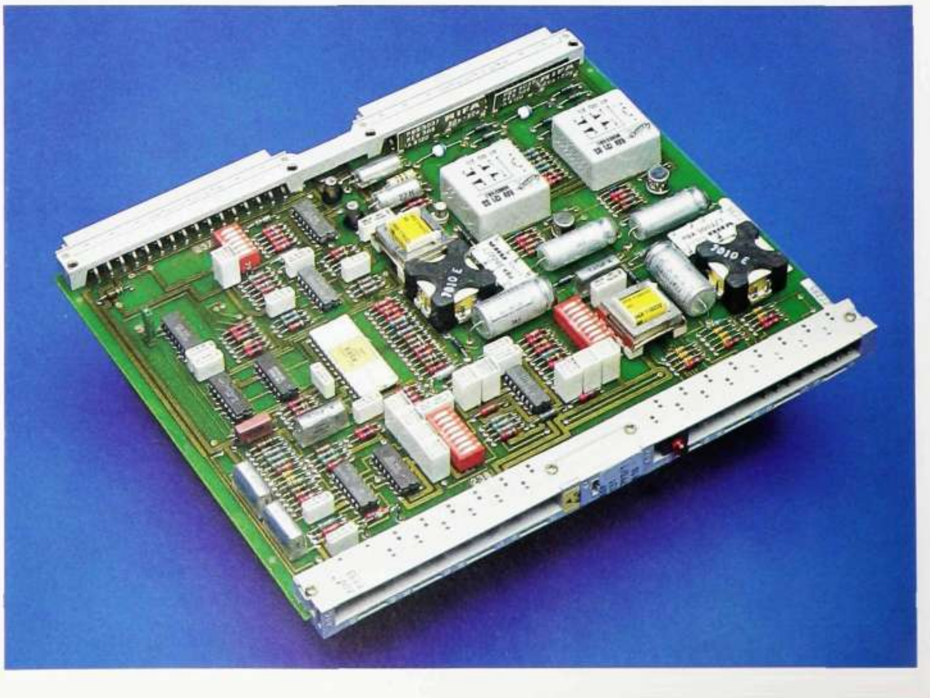


Fig. 5
A printed board assembly containing a number of plastic film capacitors

The types of films described in the table are thermoplastics. They are stretched biaxially during the manufacturing process. During the winding the plastic is in such a condition that very little force counteracts the stretching, but it is an inherent characteristic of the plastic that it tends to return to its original shape. The completed capacitor winding is therefore heated to the glass conversion temperature of the dielectric, which releases a contracting force that remains when the temperature is reduced to the normal temperature range of the capacitor and which gives mechanical stability. The mechanical stability obtained in this way also ensures electrical stability.

Electrodes and leads

The electrodes in the capacitor can be of two types:

- Electrodes made of metal foil, usually a tin-lead alloy or aluminium, approximately 5 μm thick.
- Electrodes made by means of vacuum vaporization, which deposits a 0.03 μm thick metal layer, aluminium or zinc, on the film.

The electrode layers have a resistance per square unit of surface area of some thousandths of ohms for metal foil and between 1.5 and 2.5 ohms for metallized electrodes.

Fig. 6
Supervising the metallizing process

Polystyrene is the only one of the above-mentioned dielectrics that is not suitable for metallizing, partly because of its temperature sensitivity. Metallized electrodes have an obvious advantage as regards volume. However, the thin electrode material limits the performance at high frequencies. Metal foil is therefore used for capacitors that have to be able to withstand high currents and which require a large frequency range.

The leads are made of copper wire with a thick coating of tin. In certain capacitors the wire has a steel core in order to increase its ability to withstand vibrations.

Encapsulation

The encapsulation must protect the capacitor winding against mechanical environmental damage and also be a good electrical insulator. There are different types of encapsulations for different types of climate. The materials used must as far as possible be fire-resistant and self-extinguishing. The dielectrics are affected by external factors and in the long run this affects the capacitor parameters. Penetrating moisture will always affect a capacitor in some way, and it can be devastating for certain dielectrics.

The moulding material used for encapsulation is a thermocuring epoxy resin. Embedding the capacitor winding is a common encapsulation method. Another is to place the winding in a plastic case which is then filled with thermocuring resin. In certain cases a moisture barrier made of metal foil is added.

The requirements for accuracy of size are very stringent since capacitors, particularly the smaller types, are now to an increasing extent being automatically mounted on printed boards. The leads must have good soldering properties to ensure good contact in the finished circuit. The moulding material must withstand the chemical effect of the cleaning liquids normally used after soldering.

Design and characteristics

Precision capacitors

By careful choice of dielectric, design and manufacturing process it is possi-

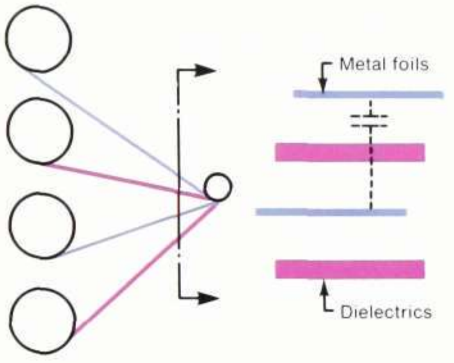


Fig. 7
Winding with metal foil electrodes

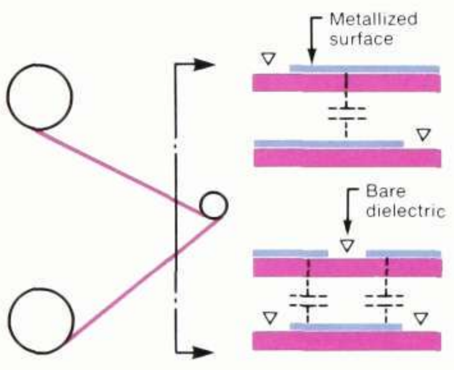
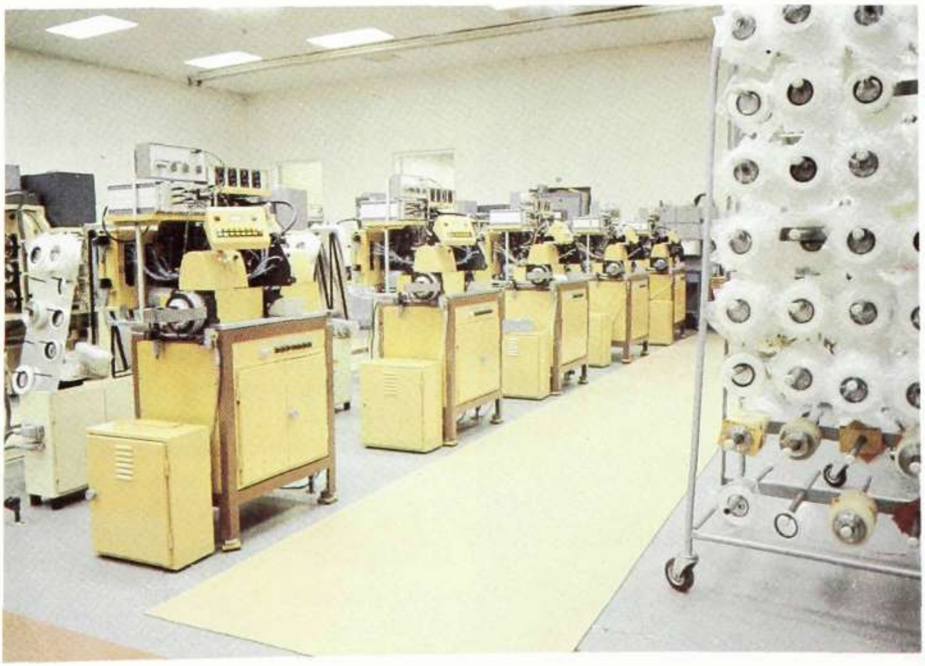


Fig. 8
Winding with metallized plastic film, a single capacitor at the top and a series capacitor below

Fig. 9
Machines for winding precision capacitors



ble to make very stable capacitors. RIFA manufactures precision capacitors in the capacitance range 50 pF to 0.25 μF with the following characteristics:
- narrow capacitance tolerances, down to ±0.5%
- low loss factor, $\tan\delta$ less than 2×10^{-4} at 10 kHz and less than 10×10^{-4} at 1 MHz
- high degree of insulation, with an insulation resistance of between 10^5 and 10^6 Mohms
- large and well defined frequency range, f_0 between 1 MHz (0.25 μF) and 40 MHz (1000 pF)
- low and well defined temperature coefficient, $-(80\pm20)\times10^{-6}$ per °C for low capacitances and $-(160\pm40)\times10^{-6}$ per °C for higher capacitances
- low dielectric absorption, less than 0.01%
- very good long-term stability under specified temperature and climate conditions, $\Delta C<(0.002\ C + 0.2\ \mathrm{pF})$ after 3 years at a temperature of max. +50°C and a relative humidity of max. 70%
- suitable mechanical construction, e.g. adapted to the size of the ferrite cores used for transformers and coils.

Nonpolar dielectrics with low losses provide the means of manufacturing precision capacitors with specified performance. Two such dielectrics are polystyrene and polypropylene.

Polystyrene capacitors are normally designed with metal foil electrodes, which give low losses and good frequency characteristics. Such characteristics are essential for filter applications. Precision capacitors are designed so that all winding turns of the same electrode are connected together. This is done when attaching the leads and gives the capacitor low inductance. Good connection between the turns in the capacitor winding and with the leads is essential to the performance and reliability of the capacitor. The connection resistance must be low to ensure that the capacitor is able to withstand electrical and thermal stresses. Variations in the resistance can cause signal distortion in certain applications.

Coupling and decoupling capacitors

The demands made on coupling and decoupling capacitors as regards certain parameters are moderate compared with the demands made on precision capacitors. These parameters include stability, dielectric losses and capacitance tolerances. However, in modern electronics, where the operating voltages are low, the need for miniaturization is geat. These capacitors are manufactured in the capacitance range 1000 pF to 10 μF and have the following characteristics:
- capacitance tolerances of ±10 to ±20%
- loss factor, $\tan\delta$ less than 8×10^{-4} at 1 kHz and less than 15×10^{-4} at 10 MHz
- high insulation resistance, greater than 10^4 Mohms
- specific frequency range, f_0 between 0.5 MHz (10 μF) and 10 MHz (10 nF)
- moderate ability to withstand high currents with a specific maximum value of the voltage derivative for the pulse edges, 3–30 V/μs depending on the capacitance
- small dimensions
- suitable mechanical construction, with certain types intended for automatic mounting.

Polar dielectrics are normally used for these capacitor types although they give higher losses. Some common thermoplastics that are available in thin films are polyester and polycarbonate.

The process for manufacturing these materials has continuously been refined so that films having a thickness of as little as 1.5 μm and with entirely acceptable electrical characteristics can now be manufactured. The relatively high dielectric constant and extreme thinness of the materials facilitate miniaturization in applications where the voltage rating permits.

The small dimensions also mean that the electrode material, usually aluminium, must be applied directly on the dielectric. Metallized electrodes also give the capacitor a self-healing property. If a disruptive discharge occurs at a weak point in the dielectric, the resultant current from the capacitor charge is normally sufficient to vaporize a small area of the electrode metal around the discharge point and thus remove the short circuit. This process is called self-healing and takes only a fraction of a microsecond. The self-healing ability differs in different dielectrics and is dependent on the chemical composition.

Self-healing reduces the risk of persistent short circuits and is an important feature of the metallized capacitor. A short circuit in a capacitor normally results in an operational fault, for example if it occurs in a decoupling capacitor for the feeding voltage.

Fig. 10
The capacitor ends are sprayed in specially designed metal spraying equipment

In this type of capacitor the connection to the electrode of the winding is normally done by a spraying process. Atomized molten metal is sprayed under pressure on the ends of the capacitor winding in order to give a good connection to the electrodes. Two connecting layers are normally applied, using different metals. The first layer is to provide the best possible connection to the electrodes and the second to ensure good connection to the capacitor leads. The leads are normally attached by means of a special braze welding method.

In certain applications the ability of the capacitors to withstand high currents is to a great extent dependent on the quality of the connection process. All film turns have to be connected in order to obtain a satisfactory result. In metallized capacitors where the film turns are only conneted at intervals there is a risk of current displacement in the contact zones which might result in a break.

The moderat requirements as regards capacitance tolerance and stability mean that the round windings of wound film can be pressed into a flat or oval shape before the connection layers are applied. In the final encapsulation of capacitors with radial leads the winding is placed upright, so that the capacitor occupies the smallest possible area on the printed board.

Pulse capacitors

Pulse capacitors are used in applications where high current pulses occur, for example switched power units, control circuits for picture tubes and protection circuits for various types of electronic equipment. Polar metallized dielectrics can be used in cases where the requirements as regards ability to withstand pulses are moderate. If very high ability to withstand pulses and high frequencies is required, the capacitor is made with metal foil electrodes and a nonpolar dielectric. Consideration must then be paid to the fact that such a capacitor is not self-healing, for example by using a thicker dielectric.

The demands made on pulse capacitors vary considerably in different applications. RIFA's pulse capacitors are made

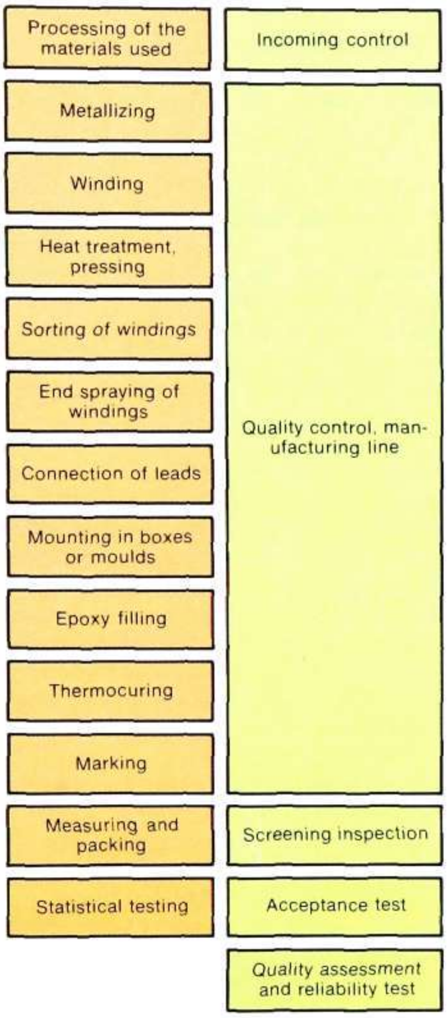


Fig. 11
The main stages in the manufacture of plastic film capacitors

in the capacitance range 1000 pF to 10 μF and have the following characteristics:

- capacitance tolerances of ± 5 to $\pm 10\%$
- low losses, at 100 kHz $\tan\delta$ is between 8×10^{-4} and 50×10^{-4} depending on the type and size of capacitor
- high insulation resistance, greater than 10^5 Mohms
- large and specific frequency range, f_o between 2 MHz (330 nF) and 25 MHz (1000 pF)
- specified temperature coefficient, $-(200 \pm 100) \times 10^{-6}$ per °C
- high capacitance stability, $\Delta C/C$ is less than $\pm 0.5\%$ after 2000 h at +85°C
- large voltage range, 160–2000 V d.c.
- high ability to withstand pulses, up to 2500 V/μs for metallized and up to 15,000 V/μs for metal foil capacitors
- suitable mechanical construction, e.g. with special leads.

The power dissipated in capacitors is one of the factors that have to be taken into account when designing a capacitor. It might necessitate special design of the capacitor elements as well as the leads and encapsulation, so that the internal losses are minimized and the cooling properties of the capacitor are the best possible. In circuits with high currents it is advantageous to use double connections to the capacitor terminals. In circuits with high voltages the capacitors should preferably have series-connected elements. Impregnation is a common method used for increasing the ability of the capacitor to withstand high voltages and also for increasing its ionization threshold.

Manufacture

RIFA's production of plastic film capacitors takes place at the Kalmar factory in Sweden and abroad at factories in Australia, Brazil, France, India and Mexico.

The machines used in the production are mainly designed and manufactured by RIFA because of the unique product designs which have made it possible to use *special production methods.*

The *manufacture* requires a high degree of cleanliness and is divided into zones with controlled environments, having different requirements as regards humidity, temperature and permissible amount of dust. Some of the materials that are used also necessitate a controlled environment for health reasons.

The product range comprises approximately 3000 items, and planning of the material handling is therefore of great *importance for the production.* The diagram in fig. 11 shows the main production stages.

Fig. 12
Equipment for measuring and sorting capacitor windingss

The first stage in the production process consists of the testing and acceptance of the incoming materials. The engineering, production and quality departments are all involved in the delivery testing, since the quality of the materials used is crucial to the manufacturing process.

To consider one example, the dielectric must be of a high and even quality, with very few holes or conducting particles. Its thickness must lie within specified tolerances. The structure and cleanliness of the surface are important factors which affect the life of the capacitor and its ability to withstand high voltages.

The metallizing of the electrods is done by vaporizing the metal on to the plastic film in vacuum. The process must be controlled carefully since it is essential that the metal layer is even and of the specified thickness. No extraneous particles, which could damage the metallized surfaces, are allowed. After the metallizing the material is cut into rolls of suitable width. The production quality is tested for each batch.

The winding process must produce windings with high and even quality. The demands are high, and the braking and pull of the different rollers in the winding machine must be adjusted with great precision to ensure that the windings have the desired characteristics.

The winding machines are placed in special rooms with filtered air supply and other protective measures in order to prevent any dangerous particles from getting into the windings. Winding machines for precision capacitors are equipped with measuring equipment which measures the capacitance and controls the machine so that the winding stops when the desired capacitance value has been reached. In cases when the tolerances are not so stringent the winding is done in simpler machines which count the number of turns and stop at a preset number. The subsequent manufacturing stages, such as the heat treatment and pressing, affect the capacitance of the capacitor windings. This fact must be taken inte consideration during the winding in order to ensure that the finished capacitor has the desired capacitance.

The heat treatment that follows the winding is very important for the stability and life of precision capacitors. The process requires great accuracy as regards temperatures and times.

Precision capacitors with capacitance tolerances of down to $\pm 0.5\%$ are sorted out before the production process continues. If the windings have to be stored

Fig. 13
Like most other machines used in the production process the braze welding and assembly machines have been designed and manufactured by RIFA

Fig. 14
Every capacitor is measured before it leaves the production line

temporarily they are placed in stores having a low relative humidity, which ensures the capacitance stability.

The connection of the metallized electrodes in the winding is made by spraying with metal which has been melted in an electric spark gap. It is then transported to the capacitor winding by an adjustable jet of air. The temperature at the spark gap and the air speed affect the particle size and thereby also the structure of the connection layer. The connection process affects the loss factor of the finished capacitor. The machinery used includes extensive peripheral equipment for the recovery of materials and for environmental protection.

The leads are attached by means of braze welding. The machine that carries out this operation also automatically assembles the capacitor winding in a box or mould. During this manufacturing stage metallized windings are exposed to a voltage higher than the rated voltage in order to remove any defects by means of self-healing.

The next stage in the manufacturing process consists of filling the moulds or boxes with epoxy resin. Resin, hardener and an accelerator in the correct proportions are mixed in an automatic machine, which fills the boxes or moulds round the capacitor windings. This is usually done in vacuum in order to avoid bubbles in the resin after it has cured. The curing takes place in ovens with carefully regulated temperature.

This process determines the environmental properties of the capacitor to a great extent. For example, with certain dielectrics it is essential to ensure moisture proofing.

During the whole production process testing is carried out after important stages. Before the capacitors leave the production line they are also measured in automatic machines which compare their values with set tolerances.

After this testing, which is carried out on every item, all products go to the quality deparment for final testing. Statistical test methods are used for checking whether a batch can be passed. The quality trend is also tested by means of periodic sampling of manufactured and tested batches for, for example, life tests. The chosen manufacturing process and the control method used ensure that the manufactured capacitors have good perfomance both at the time of delivery and after a long period of service.

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Contents

COVER
ERITEX 10, electronic typewriter for teletex communication and word processing

Remote AXE 10 Subscriber Switch in a Container

Björn Norevik

The digital subscriber switch in the AXE 10 system can be placed either in the parent exchange or remotely. In the case of remote subscriber switches it can often be an advantage to have the equipment installed in a container, not only as a temporary solution but also as a more permanent arrangement. A product package has therefore been prepared which can be used for remote units of different sizes up to 2048 subscribers. The product package also includes cooling equipment, which can be of the conventional type or a system where water is used to remove the surplus heat. The latter system offers standby cooling if there is a mains failure.
The author discusses the motives for this type of equipment and what demands can be made on it, and also describes the equipment included in the product package.

UDC 621.395.3

When planning a new telephone network or extending an existing one it is necessary to assess many uncertain factors. However, some basic facts are well known. The subscriber network is responsible for the greater part of the overall cost of the telephone network. Thus an extension often requires large investments in the network, particularly if existing cable routes are already fully utilized.

Consideration must also be paid to increases in the building costs. Uncertain factors, e.g. the rate of growth and the demand for new telephone facilities, necessitate flexibility in the choice of both temporary and permanent solutions.

A digital remote subscriber switch – an economic solution

The digital subscriber switch in AXE 10 has been described in detail in a previous article[1]. The article also described how the digital subscriber switch could be installed remotely from the parent exchange.

A remote digital subscriber switch, RSS, constitutes an integral part of an AXE 10 exchange, and subscribers connected to RSS are thus offered exactly the same facilities as other AXE 10 subscribers.

It is economical to connect remote subscriber switches to their parent exchanges by means of PCM links both when building up new subscriber networks and when extending existing ones. In the case of new networks the

Fig. 1
A container in place and connected to the network in Saudi-Arabia

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need for cables in the primary network will be considerably less than if the subscribers are connected directly to the parent exchange, and in the case of extensions large investments can also be avoided. This applies especially if the cable routes are already fully utilized, since every PCM link installed frees several cable pairs.

Remote subscriber switches in containers

The relatively limited physical size of RSS also makes it possible to install it in a container, fig. 1. This method can be used for subscriber switches with both temporary and permanent positions. The equipment can be moved according to need.

A container is also considerably cheaper than a conventional exchange building.

Applications
The reasons given above make RSS installed in a container particularly suitable for the following applications:
- as a permanent extension to an already fully developed exchange building
- as a permanent extension in an existing network, particularly when the existing cable routes are fully exploited
- as a semi-permanent or permanent solution when building up a new subscriber network and there is some uncertainty regarding future developments
- as a temporary solution when quick service is required, i.e. before the exchange building has been completed or in emergency situations when the ordinary exchange is out of operation because of a fire or for some other reason.

Product package

The type of container chosen for RSS is an insulated refrigeration container, built to ISO standard dimensions[2] (10 or 20 feet long, 9 feet high, 8 feet wide). The shorter container is used for an RSS for up to 512 subscribers, and the 20-foot container for the full 2048-group, fig. 2.

Since the container is insulated, the same type can be used for all installations regardless of the widely differing environments that will be encountered around the world, and adaptations to suit different markets are avoided. An efficient and reliable cooling system ensures a suitable working environment for the telephone equipment inside the container.

The container is built to ISO standard[2] as regards dimensions and handling facilities. The handling requirements relate to, for example its strength and devices for lifting the container. The standard dimensions also facilitate transport (by train, lorry, boat or aricraft), since standardized handling equipment, such as cranes and trucks, can be used. The easy handling and the standardized construction also contribute to speedy installation on site, so that the equipment can be put into operation almost immediately. If necessary, RSS can also just as easily be moved to a new site later on, to meet new demands.

Since the version of RSS intended for container installation is standardized, all cabling within and between magazine groups, as well as to the other

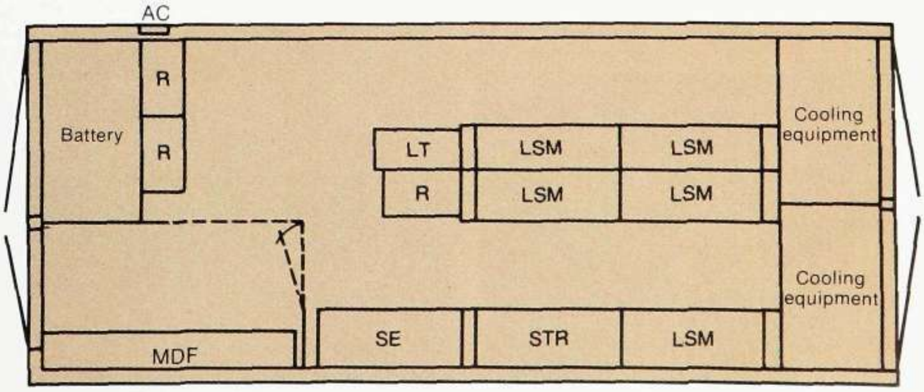


Fig. 2
The layout of a 20-foot container with a remote digital AXE 10 subscriber switch for 2048 subscribers

LSM Line switching modules
LT Line terminal for PCM
R Power supply equipment
STR Signalling terminal equipment
SE Special subscriber equipment for coin box sets, subscribers' private meters etc.
MDF Main distribution frame
AC Alternating current connector

Fig. 3
The container is brought out from one of Ericsson's factories near Stockholm, Sweden

Fig. 4
The container en route

Fig. 5
Transshipment

Fig. 6
Jacks are used at the installation site to lower the container onto its base

equipment in the container, can be prefabricated and installed before delivery. This means a considerable reduction in the amount of installation work required on site.

A complete RSS in a container, i.e. the product package, includes power feeding, cooling, main distribution frame etc., and the equipment in the container can therefore be tested as a complete unit before delivery. This means that the installation testing on site is reduced to a simple verification test before the equipment is taken into service.

Layout

Fig. 2 shows the layout of a container with a full 2048-group RSS. The necessary telephone equipment, line modules LSM 0-15 for the 128-groups and the equipment common for the whole 2048-group, are placed in a single row against the wall and one double row. This layout utilizes the available space efficiently and gives two aisles with sufficient space for installation and maintenance work. The cooling equipment has been placed at one end so that installation as well as any maintenance can be carried out via the doors at that end, without access to the telephone equipment. This also applies for the battery area, which for safety reasons is completely separate, with access via a separate door.

The main distribution frame can also be separated from the telephone equipment, by means of a folding inner door, thereby creating a dust lock. The main distribution frame is also placed against the wall in order to minimize the amount of space required and to ensure sufficient working space for installation and any alterations that might be necessary during operation. The container also has a certain amount of space for transmission equipment, temporarily connected I/O devices, fire extinguishers etc.

Container construction

The container is made of a special steel with the walls welded to a framework so that an airtight and robust construction is obtained. It is then given a finish that provides long-term protection against corrosion even in extreme environments. Ventilation and dehumidification takes place via an air inlet at one end and through an outlet at the other end via the battery area.

The container is equipped with four adjustable feet so that it can be aligned horizontally on a concrete foundation or concrete plinths, fig. 1.

Cooling system

In order to ensure a suitable environment for the telephone equipment, regardless of the external environment, RSS in a container can be equipped with a conventional air-conditioning system or Ericsson's new cooling system which uses water to remove the surplus heat. This system has been described in detail in a previous issue of Ericsson Review[3].

Ericsson's new cooling system is particularly suitable for unmanned equipment of this type, since it has high reliability and requires only a minimum of maintenance and, above all, since it is equipped with standby cooling in the form of a tank of cold water. This tank is dimensioned so that if a mains failure should occur the standby tank will last as long as the batteries of the power equipment. This ensures reliable operation throughout the specified standby time.

Power equipment

The power equipment is of the type BZA 106, a type which has been developed for use in small equipments like RSS and which meets Ericsson's standard requirements for power distribution to electronic exchange systems. The power equipment has a modular structure and its capacity can be adapted for RSS units of different size.

The batteries are placed in a separate battery area, mounted on one to three shelves. There is sufficient space for the large battery capacity required in order to provide the long standby time, which in many situations is needed for a remote subscriber switch in a container.

Main distribution frame

The main distribution frame is of the same type as Ericsson's miniature MDF, BAB 340 with slot connectors, but with the construction practice modified for single-sided installation (BAB 345), fig. 7. This layout provides ample working space for installation and for alterations during operation.

Summary

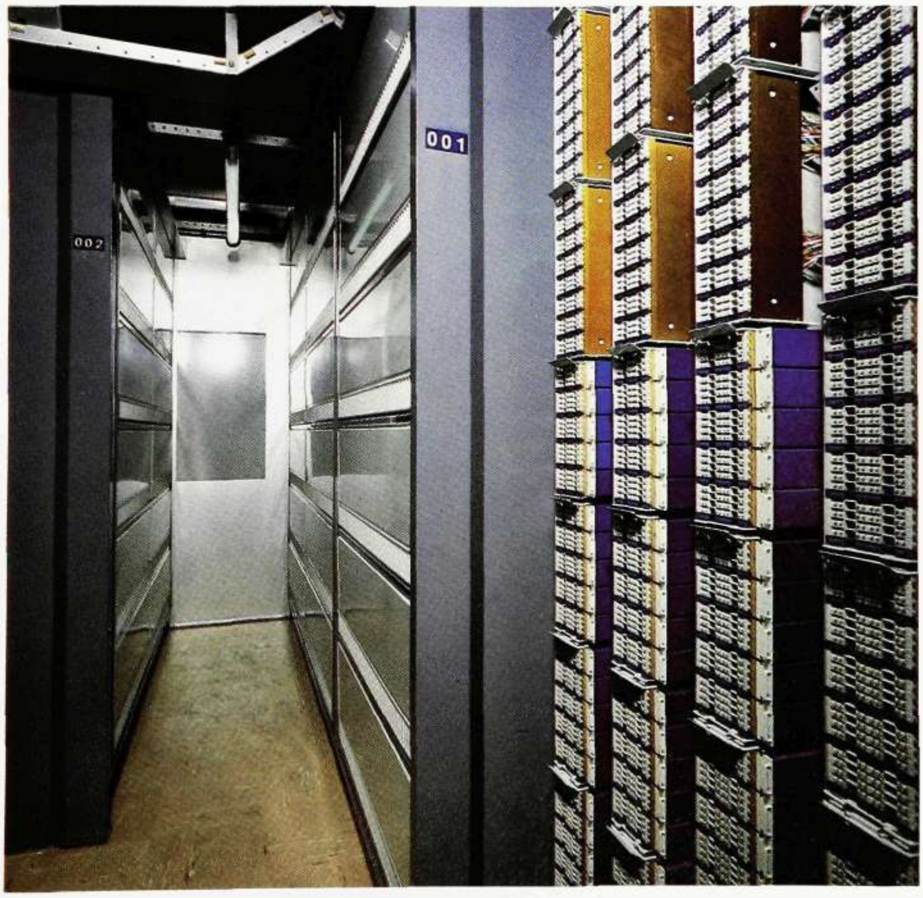
The advantages of placing the AXE 10 remote digital subscriber switch in a container can be summarized as follows:
- In all applications there is a considerable reduction of the installation time, since the equipment can be tested as a complete unit at the factory before delivery.
- The chosen container types are built to ISO standard sizes, which simplifies handling and transport, figs. 3–6.
- The exchange environment for the telephone equipment is maintained with the aid of a conventional air-conditioning system or Ericsson's new cooling system with stanby cooling during mains failures.
- The power distribution meets Ericsson's standard requirements for electronic exchanges.
- The batteries are accessible from outside through a separate door, and the main distribution frame can also be separated from the telephone equipment.
- Two sizes of RSS in container are available:
  - up to 512 subscribers in a 10-foot container
  - up to 2048 subscribers in a 20-foot container.

Other AXE 10 applications

The digital remote subscriber switch is also available in a smaller size for the range 64–128 subscribers. This equipment makes it possible to introduce AXE 10 with its many facilities in the outermost parts of the subscriber network. The volume of this equipment is so small that it has been mounted in a cabinet for installation either indoors or outdoors.

Other AXE 10 products, mainly small exchanges, are also available in standardized container versions. They have the same general structure as described for RSS in this article, and thus also the same specific properties and advantages.

Fig. 7
Interior view of a container with equipment for 2048 subscribers. The main distribution frame can be seen in the foreground



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An Optical Fibre Line System for 34 Mbit/s

Tommy Jansson and Bo Stjernlöf

This article introduces Ericsson's digital line system for optical fibre cable, ZAM 34-2. The transmission rate of the system is 34 Mbit/s, which corresponds to 480 PCM coded telephone channels. The system meets all relevant CCITT and CEPT recommendations. It is primarily intended for use in urban networks, e.g. as the transmission link between exchanges or between an exchange and a number of remote concentrators or as an entrance link to a radio relay link. The system is available in two versions, one with a laser transmitter and the other with a light emitting diode (LED) transmitter. The laser permits a repeater spacing of 12 km as against 5 km for the LED. However, the LED version offers greater reliability.

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621.391.63

During the years 1977–1979 a 34 Mbit/s line system for optical fibre, ZAM 34-1, was developed for field trials. The purpose of this system was to give Ericsson and the telephone administrations the opportunity to gain experience of digital line systems for optical fibre cable. A number of such systems have been delivered to various users, and today several are in regular operation. The new line system, ZAM 34-2, is to a great extent based on the experience gained during the design and installation of ZAM 34-1 field trial systems[4,5].

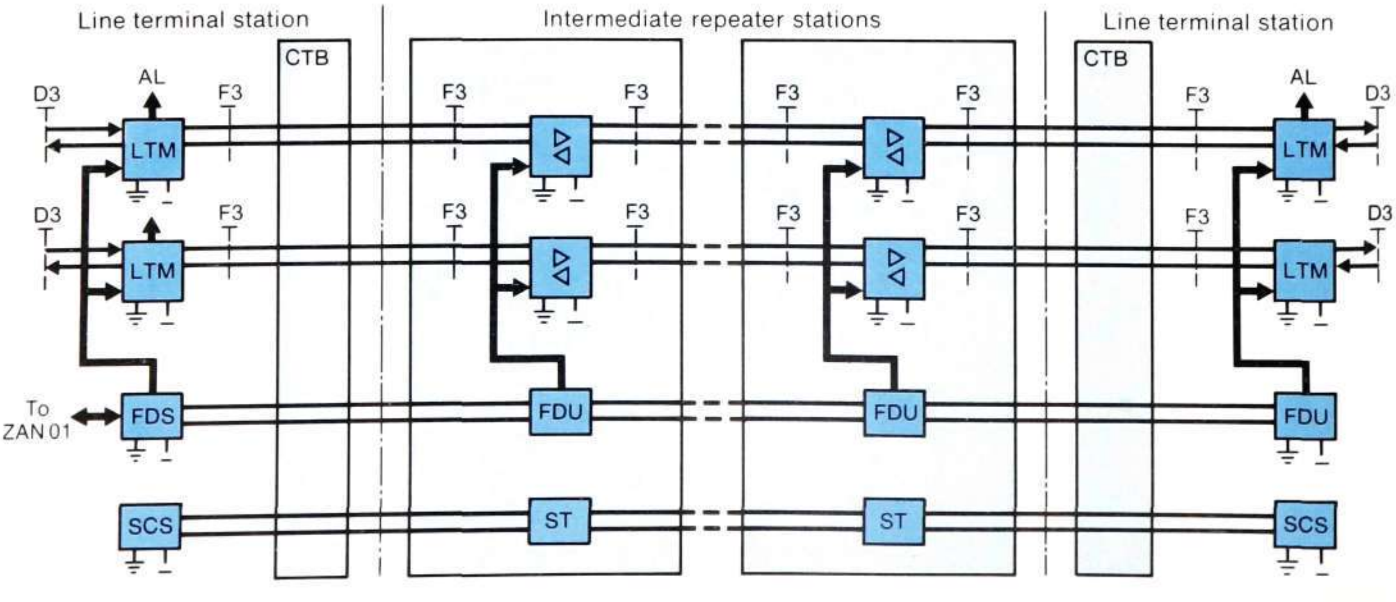
Line system ZAM 34-2 with a transmission rate of 34.368 Mbit/s is intended for digital transmission of 480 PCM coded telephone channels over fibre cable. The light source used is either a laser diode or a light emitting diode (LED), and the photo detector is an avalanche photo diode (APD). The transmission medium is a cable with graded index fibres having an attenuation of 3–4 dB per km and a bandwidth of 300 MHz · km at a wavelength of 850 nm.

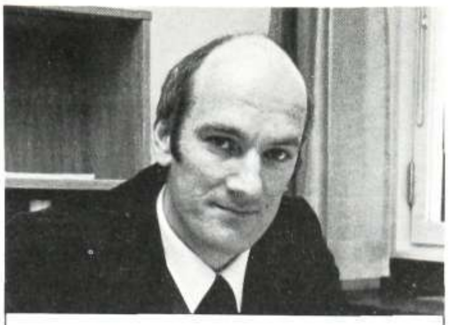
The small spectral width of the laser, together with the large bandwidth of the graded index fibre, result in negligible material and mode dispersions in the laser system, e.g. the system is limited by the attenuation. Because of the relatively large spectral width of the LED, the limitations of this system, is set by the material dispersion of the fibre.

ZAM 34-2 offers the following advantages:
- laser or LED transmitter. The latter, which could be used for repeater spacings less than 5 km, offers higher reliability and lower costs
- large repeater spacings, 12 km with laser transmitters
- thermoelectrical cooling of the laser diode for maximum reliability
- modular structure. The system can easily be converted from a shortwave system for 850 nm to a longwave system for 1300 nm by changing the transmitter and receiver units
- simple connection to the fibre via plug-in optical connectors which do not require any subsequent adjustment
- flexible mechanical construction using Ericsson's BYB construction practice, which simplifies installation and handling. All connections are made with connectors on the fronts of the units

Fig. 1
The equipment in the 34 Mbit/s system

LDM	Line terminating magazine
FDU	Fault detector unit
FDS	Fault detection shelf
SCS	Service circuit shelf
ST	Service telephone
CTB	Cable terminating box
D3	34 Mbit/s interface, CCITT G.703.8
F3	41 MBaud optical fibre interface
AL	Alarm interface





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line transmission
Telefonaktiebolaget LM Ericsson

- the same magazine is used for the line terminal and the intermediate repeater, which makes it easy to change the equipment over from termination to through-connection
- fault location function and facilities for connection to Transmission Maintenance System ZAN 101.

System characteristics

The interface for the 34 Mbit/s line system is D3, and is in accordance with CCITT Rec. G.703. An 8/34 Mbit/s or 2/34 Mbit/s multiplexer, e.g. Ericsson's ZAK 120/480[1] or ZAK 30/480[1], or a digital radio relay link can be connected to this interface.

Line system ZAM 34-2, fig. 1, comprises line terminals and two-way intermediate repeaters for transmission over optical fibre cable.

The equipment is normally mounted in M5/BYB bays and consists of
- line terminating magazines, LTM, where the signal is recoded, the jitter is reduced and the line is supervised
- locally powered two-way intermediate repeaters that regenerate the signal at regular intervals along the fibre cable
- a cable terminating box (CTB) for connecting the fibres in the line cable to the fibres in the bay cables
- fault detection equipment for remote supervision of line terminals and intermediate repeaters
- two or four-wire service circuit equipment in the terminal bay, and connection points in the intermediate repeaters for a service telephone.

The incoming line cable is terminated in a cable terminating box, where the fibres are distributed to the different systems. In the box the fibres of the line cable are welded to the fibres in the bay cables. The other ends of the bay cables are fitted with plug-in optical connectors, for connection to the transmitter and receiver units of the system in question.

Any copper pairs in the fibre cable can also be terminated in the cable terminating box.

The equipment is easy to install and has well defined internal interfaces. The BYB magazines are delivered with the units in place and with all external connections on the front of the units. The equipment is strapped for the most common application, and only a minimum number of straps have to be made during installation.

Terminal equipment

Line terminating magazine

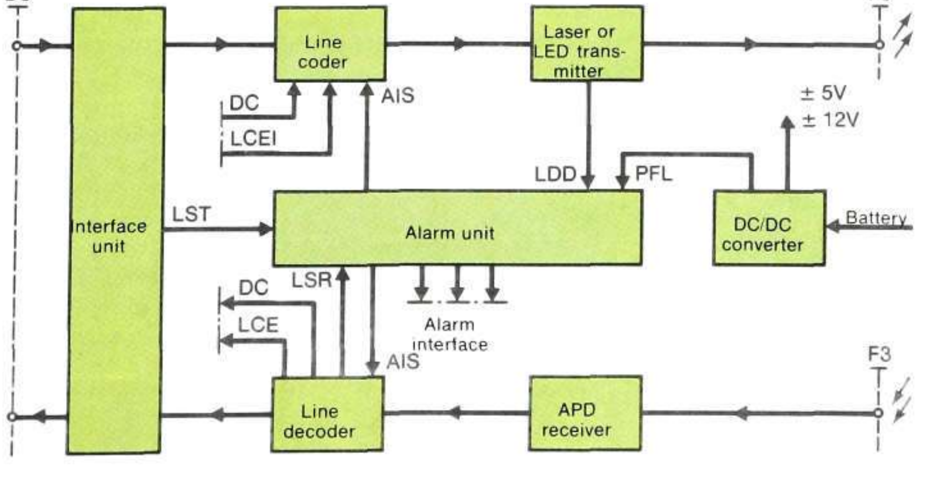
The main functions of the line terminating magazine, LTM, fig. 2, are to
- adapt the signal in the send and receive directions between the D3 interface standardized by CCITT for 34 Mbit/s and the optical fibre interface F3
- detect and indicate alarms.

The basic version of the line terminal consists of seven printed board assemblies:
- interface unit
- line coder
- laser or LED transmitter
- APD receiver
- line decoder
- alarm unit
- d.c./d.c. converter for ±5 V and ±12 V.

In the receive direction of the interface unit the HDB3 signal from the D3 interface is equalized for the attenuation of the connecting cable (max. 12 dB at

Fig. 2
Block diagram of the line terminating magazine

AIS	Alarm indication signal
DC	Data channel
LCE	Line code error detection
LCEI	Line code error injection
LSR	Loss of signal, receive direction
LST	Loss of signal, transmit direction
LDD	Laser diode degradation
PFL	Failure of local power supply



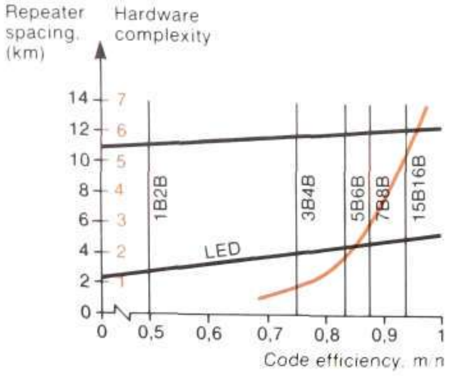


Fig. 3

Repeater spacings and hardware complexity with different types of codes. The choice of the 5B6B code gives a good compromise between hardware complexity and dispersion limiting

17 MHz) and regenerated. The signal is then recoded to a binary serial bit stream and passed to the line coder. In the case of loss of input signal an alarm is given to the alarm unit. Violations of the HDB3 coding law result in an error pulse which is available at a test point.

The *line coder* carries out the code conversion to adapt the data bit stream for a laser or LED transmitter. The chosen code is of type 5B6B, i.e. five binary symbols are recoded as six binary symbols. This gives a symbol rate on the fibre cable of $6 \cdot 34/5 = 41$ MBaud. Since the number of incoming binary states, $2^5 = 32$, is less than the number of outgoing binary states, $2^6 = 64$, the code has built-in redundancy. This redundancy is used to create a code spectrum with a constant d.c. voltage component and a spectral energy distribution that is suited for optical transmission. The redundancy is also used for error supervision of the signal and to get a low-capacity asynchronous data channel.

The choice of code is by necessity a compromise. Other codes, i.e. 3B4B which is used in ZAM 34-1, give a greater dispersion penalty in LED systems. Higher order codes would require too complex equipment, fig. 3.

The bit rate conversion that is needed for the coding is carried out with the aid of a crystal-controlled phase locked loop, PLL. The loop bandwidth has been chosen so that the jitter on the incoming signal is reduced, resulting in a decreased alignment jitter requirement in the receivers of the repeater.

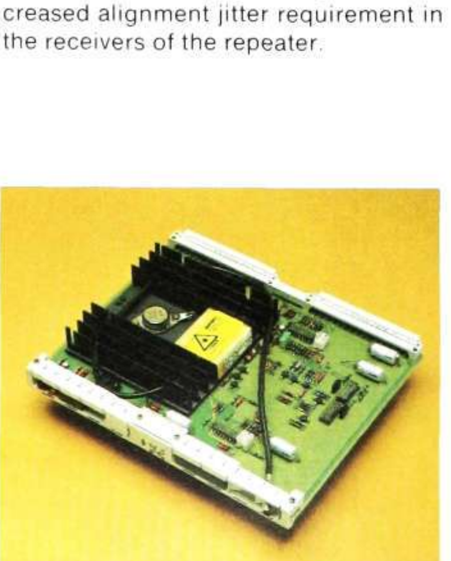


Fig. 4

The laser transmitter with the laser unit. The laser is cooled to a constant temperature in order to obtain the highest possible reliability. The laser is stabilized by means of mean value regulation of the operating point

Line code errors can be injected in the line coder at an optional rate in order to test the fault location system and alarm unit. The coding ensures that none of these bit errors remain after the decoding of the line signal.

The *laser transmitter* receives the coded 41 MBaud NRZ (Non Return to Zero) bit stream and the clock signal from the line coder and converts them to a 41 MBaud RZ (Return to Zero) bit stream. This is done in order to reduce the problems of inter-symbol interference in the receiver. The RZ signal modulates a laser, which provides an optical signal on the fibre cable. The operating point of the laser is mean value regulated by means of optical feedback with a photo diode at the rear mirror of the laser. This stabilizes the laser operation and compensates for temperature variations and ageing effects.

The laser is cooled by a thermoelectric element to a constant temperature in order to obtain maximum reliability. A special laser unit has been developed which meets the system requirements, fig. 4.

The laser temperature can be checked by measuring a voltage in a test point on the front of the unit.

The operating life of the laser system depends mainly on the life of the laser, and the system is therefore equipped with a circuit which supervises the condition of the laser. An alarm is given when the laser has degraded. The alarm is given well in advance, before the system performance is affected.

An alternative to the laser transmitter has been developed, namely an *LED transmitter* which is intended for short transmission distances. It is fully compatible with the laser transmitter as regards its connection. The LED transmitter gives the system even higher reliability than the laser transmitter. Like the latter, the LED transmitter converts the incoming 41 MBaud NRZ flow to an outgoing optical 41 MBaud RZ flow. The LED is mounted on a cooling flange in order to reduce its operating temperature.

The *receiver* detects the incoming optical signal and converts it to a 41 MBaud

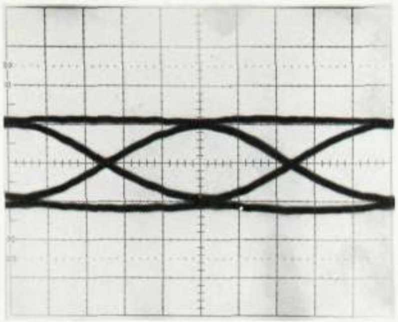


Fig. 6
Oscilloscope picture of the signal at the detection point in the form of an eye diagram for a laser system. Cable length: 10 km

NRZ electrical bit stream. An avalanche photo diode, APD, is used as the photo detector in order to obtain the best possible receiver sensitivity and dynamic range. After the conversion in the APD the signal is amplified in an input stage. This is followed by an amplifier with automatic gain control, AGC. The APD is mounted direct on the hybrid-type input circuit, fig. 5. This minimizes the input capacitance, which would otherwise reduce the sensitivity of the input stage. The use of hybrid technology also minimizes the effects of any electromagnetic disturbances. The sensitivity and dynamic range are further improved by regulation of the APD reverse voltage. When an LED transmitter is used, a dispersion equalizer is automatically connected in after the AGC amplifier. The amplified signal is filtered in a low-pass filter so that the highest possible signal-to-noise ratio is obtained at the detection point. Fig. 6 shows the signal at the detection point in the form of an eye diagram for a laser system with 10 km of fibre cable.

The timing recovery from the signal flow takes place in a PLL containing a voltage controlled crystal oscillator, which provides the necessary compensation for temperature variations and ageing effects. The relatively high Q value of the oscillator and its insensitivity to disturbances ensure low output jitter. The signal is regenerated with the aid of the recovered timing. The binary regenerated 41 MBaud signal and the clock signal are fed to the decoder. These signals are also accessible for an optional error detector, ED. The voltage across the APD can be measured at a test point on the front of the unit. For reasons of safety the voltage is divided down to a hundredth of the actual value.

The binary input bit stream to the decoder is series/parallel converted and code converted in a 5B6B decoder. The data signal and timing signal are then fed to the previously described interface unit.

The redundant words that are not used in the normal encoding process cause error pulses. These pulses are fed to a resynchronization circuit and also to an alarm unit. The resynchronization logic ensures that the system is resynchronized if the error rate of the incoming bit stream is too high. The logic is fitted with error burst blocking in order to prevent resynchronization for high error rates of very short duration. The information transmitted over the data channel is also accessible in the decoder.

The timing conversion required for the code conversion is carried out with the aid of a crystal-controlled PLL. The loop bandwidth has been chosen so that the line signal jitter is reduced. The jitter out towards the multiplex equipment is thereby minimized. The serial data stream from the line decoder is HDB3 coded in the send side of the interface unit.

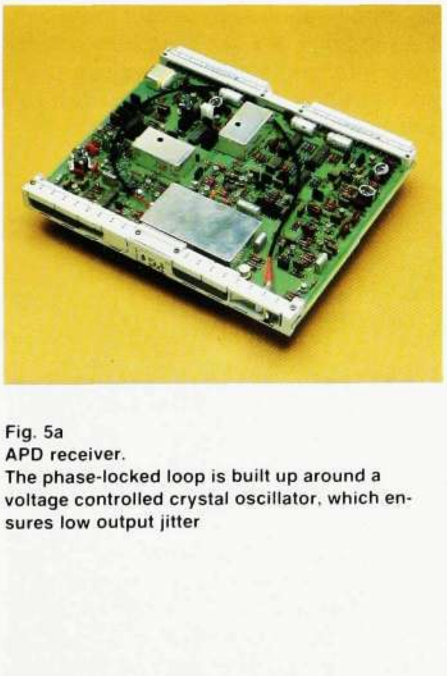


Fig. 5a
APD receiver.
The phase-locked loop is built up around a voltage controlled crystal oscillator, which ensures low output jitter

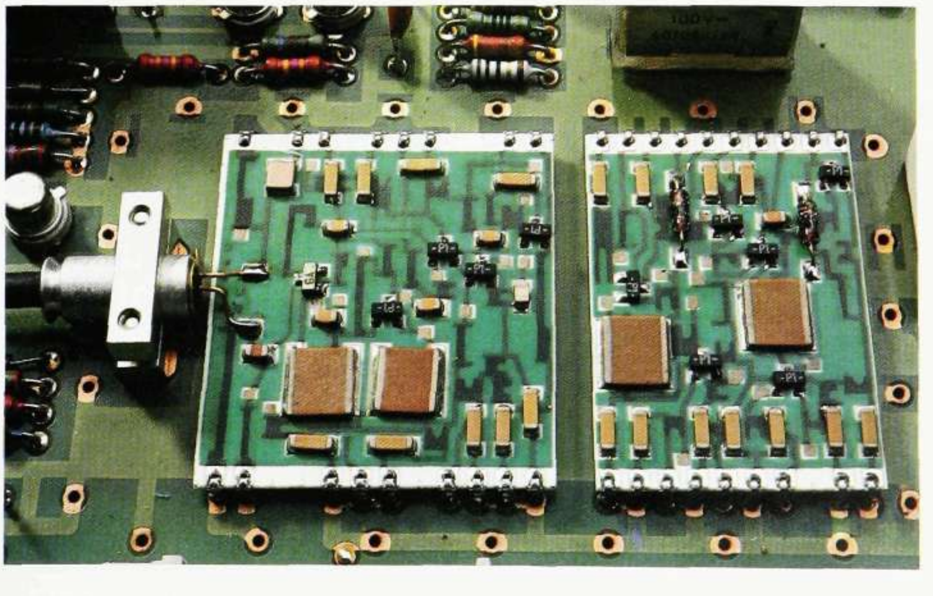


Fig. 5b
A part of the APD receiver with the lid removed from the unit containing the hybrid circuit with the avalanche photo diode. The APD is mounted direct on a hybrid circuit in order to ensure the best possible receiver sensitivity. A wide dynamic range is obtained through regulation of the APD voltage combined with an AGC amplifier

The primary alarms are those recommended by CCITT. They are concentrated in the alarm unit of the magazine. Alarm concentration and alarm transmission are carried out in accordance with the same principles that apply for other equipment in the BYB construction practice. An alarm indication signal (AIS) is inserted in the send or receive direction if there is a loss of input signal at the D3 interface or too high error rate on the line.

The equipment is mounted in a BYB magazine, fig. 7, and is powered from a battery via the d.c./d.c. converter in the magazine.

It has been possible to use standard wire wrapping methods for the wiring of the equipment, since the internal transmission symbol rate between the printed board assemblies is limited to 41 MBaud.

Intermediate repeaters

The 34 Mbit/s line system is intended primarily for urban networks. This means that the repeater spacings are such that point-to-point circuits or circuits containing only a few intermediate repeaters are most likely to be used. In those cases where intermediate repeaters have to be used it is therefore also possible to use local power supply.

The two-way intermediate repeater with local power supply is mounted in a BYB magazine which is an equipment version of the line terminating magazine. The intermediate repeater conists of
- two laser transmitters
- two APD receivers
- one d.c./d.c. converter.

Identical types of units are used in the intermediate repeater and the line terminating magazine.

Intermediate repeaters which can be power fed via a separate copper pair in the fibre cable will be available later. The only modification required for these will be to change the d.c./d.c. converters. These intermediate repeaters can then be installed in buried containers if required.

Fig. 7
The line terminating magazine is built up in a BYB magazine, with the connectors at the front of the units

The line terminals and intermediate repeaters can be equipped with a unit for supervising bit errors in the transmission. This error detector unit, ED, contains two bit error detectors, one for each direction of transmission, which monitor the variations in the running digital sum, and which send error pulses to the fault detector in the fault detection system if bit errors are detected. ED also contains a function for monitoring laser alarms from the laser transmitters in the intermediate repeaters and line terminals. Any laser alarm causes short circuiting of an extra copper pair, which is connected to ED. The short circuit is detected by the ED in the line terminal of the supervising station which is connected to the terminal alarm unit via connectors on the fronts of both units. The extra copper pair is power fed from the supervising line terminal via a separate remote power feeding unit.

Bay

A bay holds terminal equipment for a maximum of ten systems. The capacity is reduced to six systems if a fault detection shelf and rectifier are also mounted in the bay. A board holder unit is mounted at the bottom of the bay. It can contain an alarm concentrator for bay alarms, a service telephone connector, and also a d.c./d.c. converter for powering the fault detection shelf, if provided.

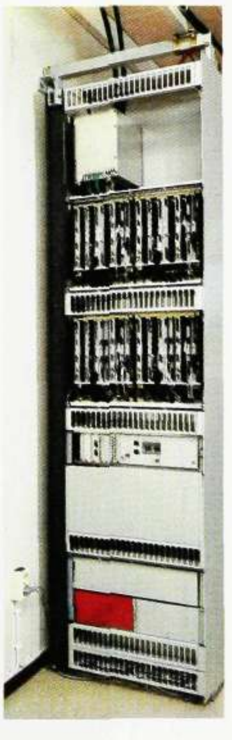


Fig. 8
A line terminal bay with six line terminals, cable terminating box, fault detection shelf and mains rectifier

Fault location

Repeater fault location

Faulty line repeaters are located with the aid of a fault detection system which is common for several line systems. The system consists of a fault detection shelf (FDS), placed in the station from which fault location is to be carried out, and a fault detector unit (FDU), placed together with the intermediate repeaters. The fault detection shelf is identical to the one used for Ericsson's 2, 8 and 140 Mbit/s line systems, which means that it can be connected to the Transmission Maintenance System ZAN 101.

The fault detection system makes it possible to detect bit errors in the line signal during traffic.

Each repeater contains an ED, which is connected to a common FDU. The latter, in its turn, is connected to the fault detector shelf via a separate metallic wire pair.

The fault detector unit is identical to that used for the 140 Mbit/s coaxial line system ZAY 140-1[6].

Information concerning the bit error rate on a certain repeater output can be transmitted to the fault detection shelf for analysis by means of a simple addressing procedure. It is possible to locate a fault by addressing consecutive intermediate repeaters and comparing the bit error rates. The system can be tested by injecting code errors in the line terminating magazine.

Service circuit

A two-wire or four-wire service telephone circuit is available in system ZAM 34-2, for example at fault location. The fault detector units in repeater stations have access to a service telephone. At the ends of the route the two or four-wire service circuit equipment is connected to the ordinary two-wire service telephone in the M5/BYB bay.

Technical data

Digital interface (D3)	
Bit rate	34.368 Mbit/s
Line code	HDB3
Impedance	75 ohms, unbalanced
Pulse amplitude	1 V
Permissible cable attenuation at 17 MHz	12 dB
Optical interface (F3)	LASER LED
Symbol rate	41.2416 MBaud
Line code	5B6B
Output power	−2 dBm −24 dBm
Wavelength (typical)	830 nm 880 nm
Spectral bandwidth (3 dB)	≤3 nm ≤50 nm
Input sensitivity	−51 dB −47 dB
Dynamic range	30 dB
Transmission medium	
Fibre type	Low loss, graded index
Permissible dispersion	9 ns 18 ns
Core diameter	50 μm
Cladding diameter	125 μm
Numerical aperture	0.21 ± 0.02
Power supply	
Nominal battery voltage	36, 48, 60 V
Mains voltage	110, 127, 220 V
Tolerance	10%
Mains frequency	45−65 Hz
Power consumption	
Line terminal	45 W
Intermediate repeater (two-way)	16 W
Fault location	
Locating faulty repeaters	Via a metallic pair
Maximum number of repeater stations	32
Service telephone circuit	Two-wire or four-wire
Ambient temperature	
Line terminal	0 to +45°
Intermediate repeater	−20 to +55°C
Dimensions	H, W, D
Line terminal and intermediate repeater	244×244×225 mm

The permissible attenuation over the loaded service circuit pair is 25 dB for two-wire and 40 dB for four-wire circuits.

In order to avoid extra copper pairs in the fibre cable, a new fault detection system is being developed which utilizes the possiblitity of transmitting the service channel and fault detection signals over the fibre used for the normal transmission.

Summary

Line system ZAM 34-2 for optical fibre is a product with good technical performance and high reliability. It has a modular structure, is robust and easy to handle and install, features which are all of great importance to the user. These properties have been achieved by

- drawing on the experience gained from the design and installation of other digital line systems, such as ZAM 34-1 and ZAY 140-1
- applying standardized construction practice, alarm and supervision methods
- using plug-in optical fibre connectors
- simulating and optimizing the system performance and the functions of the circuits with the aid of computer programs
- using active cooling and continuous alarm monitoring of the laser
- using LED for distances of less than 5 km.

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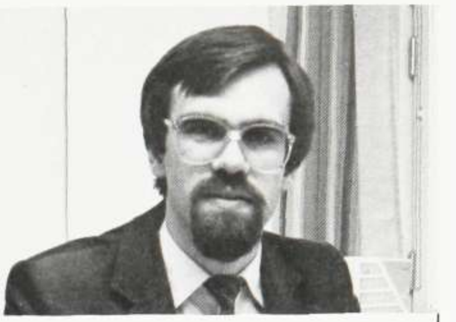
Transmultiplexers



Sixten Ekelund

Analog transmission systems are distinguished by their high capacity, which makes for good economy in the long-distance network. Since digital systems are being introduced in various parts of the network, it is likely that both FDM and TDM signals will be used in the telecommunication network in the foreseeable future. The transmultiplexer carries out the conversion between the two multiplex structures.

Ericsson has developed transmultiplexers for different purposes, all of which are in accordance with applicable CCITT recommendations. In this article the transmultiplexer applications, properties and structures are described. The equipment is designed using conventional analog/digital conversion per speech channel in the baseband, which has technical and economical advantages.

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UDC 621.376
621.395.43

During recent years there has been an increasing demand for analog multiplex equipment (FDM). FDM technology is superior in the long-distance network as regards capacity and economy.

The transmission media used are traditional coaxial cable systems having a capacity of up to 10800 channels, as well as radio relay link and satellite systems. Recently developed radio systems, which permit single sideband transmission, have a capacity of up to 6000 channels.

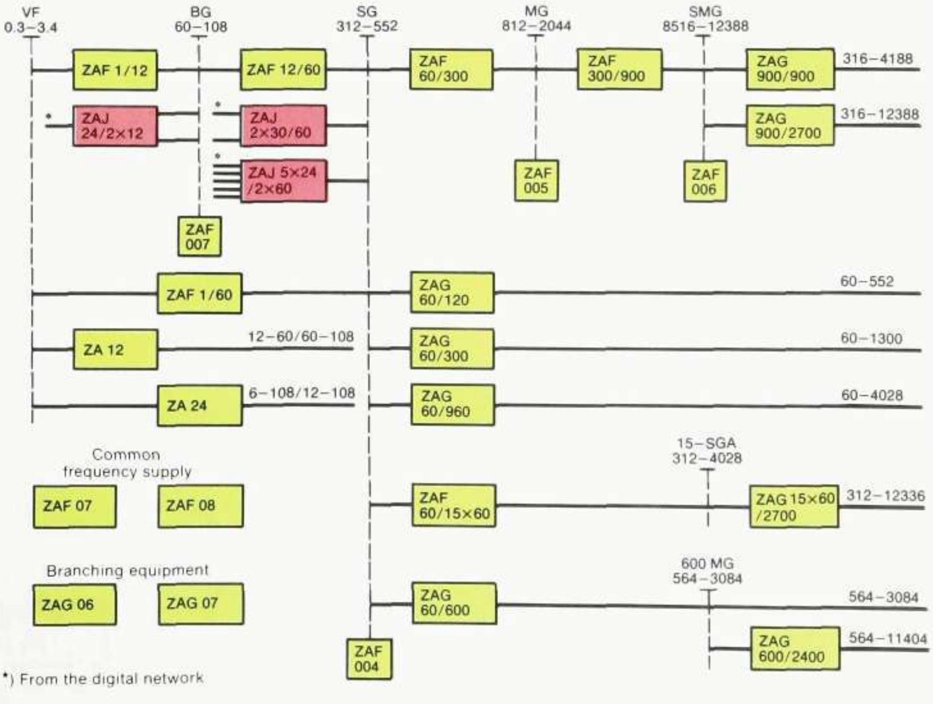
Fig. 1 shows the current hierarchy of analog multiplex systems and the positions of the transmultiplexers in it.

The hierarchy comprises not only the systems standardized by CCITT but also systems in accordance with the American L-plan, ZAG 60/600 and ZAG 600/2400, which have recently been developed by Ericsson.

Digital multiplex equipment (TDM) is being introduced on a large scale at the lower levels of the telecommunication network. This means that the number of circuits in existing cables can be increased and at the same time the transmission characteristics can be improved.

Demands for new services, which are most easily realized by means of digital technology, expedite the introduction of digital switching and transmission equipment at different levels in the network. The digital networks that are being built up in many countries are often countrywide but initially they have small routes and an open structure.

The above-mentioned factors mean that there is a need for through-connection between analog and digital networks. The through-connection can be carried out by means of transmultiplexers, with the task of converting an FDM signal to a



*) From the digital network

Fig. 1
The analog multiplex hierarchy (FDM), and the positions of the transmultiplexers, ZAJ 2×30/60, ZAJ 24/2×12 and ZAJ 5×24/2×60, in it

Fig. 2, left
The transmultiplexer as through-connection equipment between the digital and the analog network

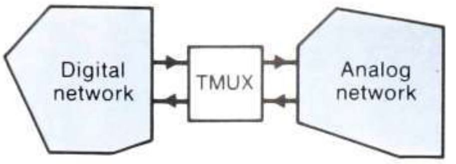
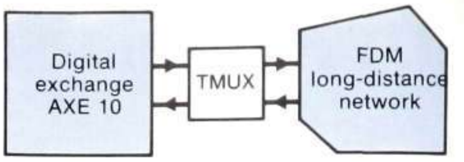


Fig. 3, right
A digital exchange connected to an analog long-distance network via a transmultiplexer



TDM signal and vice versa. Transmultiplexers have very well defined interfaces and can therefore be utilized in a simple and flexible manner, regardless of the degree of digitalization.

Transmultiplexers will be required for international traffic for a very long time, since there is an efficient analog network in existence, which will be retained for many years in parallel with the digital networks that are now being built up in many countries.

System applications

Different strategies are used when extending and modernizing telecommunication networks, and the need for transmultiplexers can therefore vary. However, there are always problems when two networks with different structures have to interwork, and these problems are easily solved with the aid of transmultiplexers, which are primarily used for:

- through-connection between the analog and the digital network, fig. 2
- connecting digital switching systems to the analog long-distance network, fig. 3.

Through-connection

The hierarchic structure of both the analog and the digital multiplex equipment makes it possible to inter-connect standardized groups of each type. Transmission networks are built up in such a way that only a proportion of the circuits provided by the line systems are terminated in each terminal, the remainder being connected through to other lines. When the network contains both analog and digital line systems the transmultiplexer offers a means of simple through-connection between the systems, fig. 4.

Similar problems with through-connection in digital satellite transmission (TDMA) can also be solved with the aid of the transmultiplexer. Earth stations usually have analog connections to the national network, fig. 5.

Many networks are built up with standby routes. For example, an analog radio relay link route can have a standby route, although with a reduced capacity, via a digital cable system, fig. 6.

Exchange connection

New exchange systems with digital through-connection are becoming increasingly important in the networks. When digital switching is introduced, particularly at higher levels in the network, connection to the analog long-distance network is necessary. The transmultiplexer allows the long-distance network to be connected direct to the digital standard interface of the exchange. From the point of view of the exchange the digital standard interface is preferable to analog speech and signalling interfaces. The transmultiplexer solution offers economic advantages and requires very little space.

It is often desirable to divide the traffic between two independent routes, fig. 7. This means that the transmultiplexer equipment will be required for a long time, since the introduction of indepen-

Fig. 4, left
Throug-connection of routes between different line systems
Analog line
Digital line

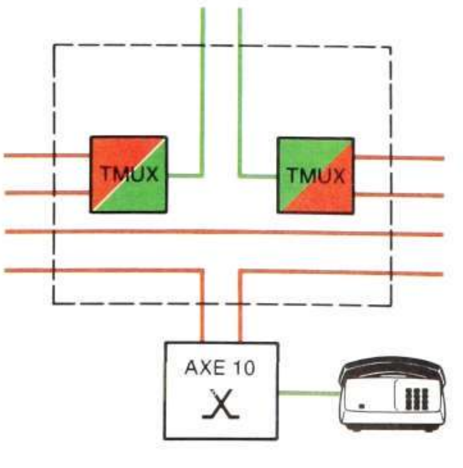


Fig. 5, right
An earth station for satellite transmission connected up via a transmultiplexer

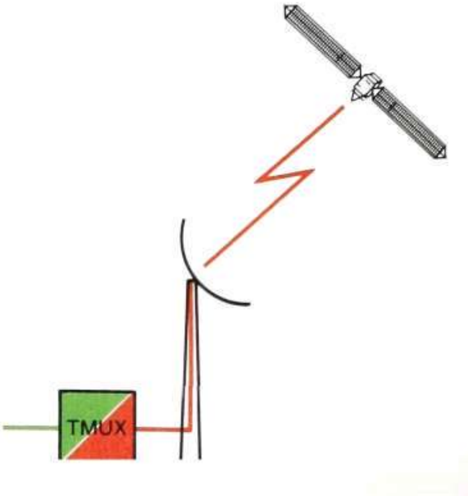


Table 1
Comparison of certain parameters for the transmultiplexer and separate FDM and PCM equipments respectively

Parameter		FDM LME	FDM CCITT G.232 G.233	PCM LME	PCM CCITT G.712	TRANSMUX LME	TRANSMUX CCITT G.792
Group delay	ms	1	–	0.45	0.6	1.5	3
Delay distortion, 1000–2600 Hz	ms	0.25	0.525	0.17	0.25	0.42	0.5
Attenuation distortion, 600–2400 Hz	dB	±0.5	±0.9	±0.3	±0.5	±0.4	±0.6

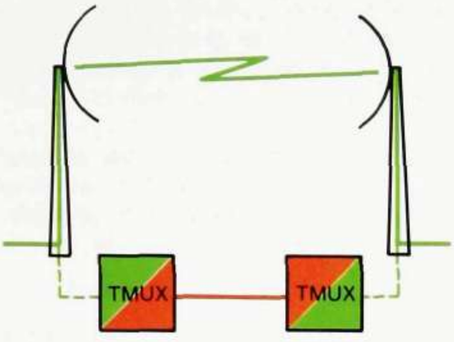


Fig. 6
A digital line used as the standby route for an analog radio relay link route

dent digital circuits to each digital exchange at this level will not take place in the foreseeable future.

System characteristics

CCITT has standardized a transmultiplexer for 60 channels[1]. Standards for a 24-channel transmultiplexer are being prepared. Ericsson's transmultiplexers ZAJ 2×30/60, ZAJ 24/2×12 and ZAJ 5×24/2×60 are intended for networks built up with 30 and 24 channel PCM systems respectively. The transmultiplexer described here is the first-mentioned one, which converts 60 channels.

ZAJ 2×30/60 converts two 30 channel 2048 kbit/s PCM bit streams to a standardized supergroup in the frequency band 312–552 kHz and vice versa. The transmultiplexer is a mechanical and fuctional unit with a performance in accordance with relevant CCITT recommendations, which ensures safe and rational operation.

In order to illustrate the performance requirements, a comparison has been made between the requirements for the transmultiplexer and the appropriate FDM and PCM equipment, with regard to certain essential parameters. In table 1 the values recommended by CCITT are compared with the values for the Ericsson equipment. Note the very small addition in group delay caused by the transmultiplexer.

The transmultiplexers described in this article work in accordance with the conventional method, i.e. analog/digital conversion takes place at speech frequencies. An alternative method based on digital filtering and fast Fourier transformation has been studied in detail. However, the conventional method offers many technical and handling advantages, for example greater flexibility and higher reliability. Moreover, the subsystems that form part of the transmultiplexers have been developed and optimized through several generations of FDM and PCM systems, resulting in, for example, low power consumption.

Installation

Analog/digital conversion with individual equipment for each speech channel requires extensive installation work, including the wiring up of a large number of connections in the four-wire and signalling interfaces. The installation of transmultiplexers only necessitates the connection of a few cables to easily accessible connectors at the front of the equipment. Fig. 8 shows the required connections.

The setting of various supergroup levels is carried out by means of plug-in U-links. On the analog side the signalling tone levels can be set in a similar way.

Synchronization

In Ericsson's transmultiplexers the carriers and bit rates of the outgoing PCM streams are derived from separate

Fig. 7, left
Independent routes between digital exchanges in a mesh-shaped network containing both FDM and TDM lines

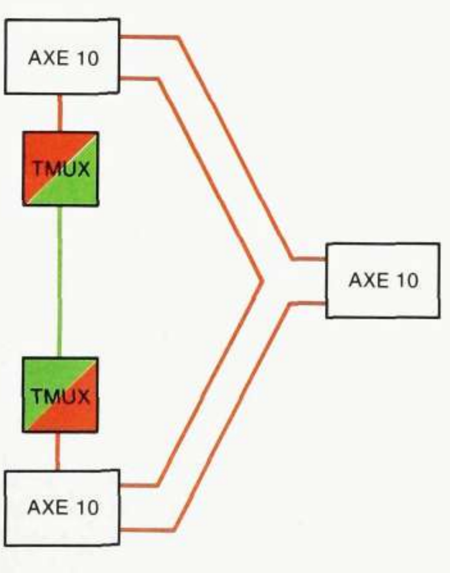
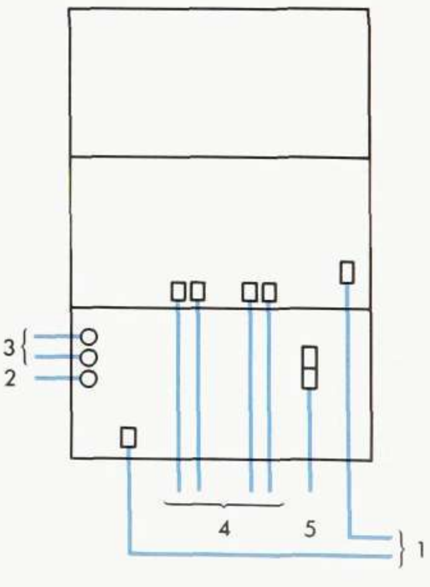


Fig. 8, right
External connections to ZAJ 2×30/60

1 Battery voltage −30 to −72 V
2 Basic frequency, 12 or 124 kHz
3 Supergroup input and output
4 2×2048 kbit/s input and output
5 Alarm interface



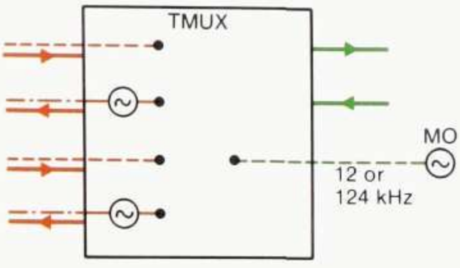
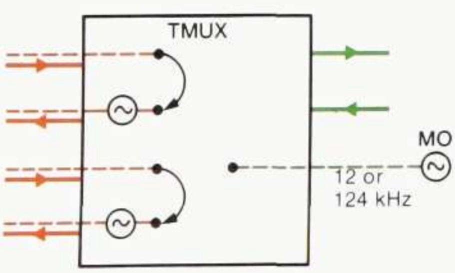


Fig. 9
Synchronization of the transmultiplexer

MO	Master oscillator
– – – –	Incoming timing
—·—·	Outgoing timing
—— (orange)	TDM signals
—— (green)	FDM signals

Fig. 10
Synchronization of the transmultiplexer when working towards a synchronous digital network

Synchronization path



clocks. This means that problems are avoided that would otherwise arise, for example because of the different frequency accuracy requirements that apply for FDM and TDM signals respectively. This method also eliminates the risk of slip, and data signals can therefore be transmitted over the speech channels without excessive error rates.

The timing signal for the outgoing PCM flow is generated by a built-in oscillator in the PCM part concerned, fig. 9. The carriers in the FDM part are derived from an external basic frequency, 12 or 124 kHz, which in accordance with normal FDM practice is available in the central frequency generating equipment of the terminal.

The transmultiplexer can work towards a synchronous digital network. In this case the recovered timing signal from the incoming PCM streams controls the internal clocks so that the outgoing signals have the same bit rate, fig. 10. The control signal is obtained by making a simple loop connection via two external cables on the front of the equipment.

Signalling

Signalling information on carrier circuits is usually transmitted by means of channel-associated outband signalling at 3825 Hz. In digital circuits one or more signalling channels in the common signalling timeslot, T16, of the PCM system are used for each speech channel. In the transmultiplexer the signalling information in the outband signal is converted, for each channel, to the corresponding information in timeslot 16 and vice versa. This method makes the transmultiplexer wholly transparent to different signalling diagrams as long as only one bit in T16 is used for each speech channel.

On the FDM side the equipment can easily be matched to outband signalling with either high or low level. Only low level is suitable for continuous signalling diagrams. The equipment can also be adapted to either of the two possible signalling diagrams where "tone" corresponds to "1" or "0".

Signalling system R2, in its original or modified form, is widely used for international and national traffic. The system is designed to transmit the line signalling in the signalling channel, whereas the register signalling is carried out by means of MFC in the speech channel.

The equipment meets the relevant CCITT recommendations for signalling system R2. For example, it is equipped with pilot receivers for interruption control.

In the analog version of system R2 a link is established for each channel between bit a and the signalling frequency, whereas bit b in the FDM-PCM direction is used to transmit alarm information from the pilot receiver.

In FDM systems there are only two signalling conditions in each direction per speech channel, and the analog version of system R2 therefore contains certain timing conditions which supplement the two signalling conditions. In the digital version of system R2, however, two bits (a and b) per speech channel are used to transmit the corresponding information, which amounts to four signalling conditions. This means that simpler terminal circuits can be used in the switching equipment.

Using the digital version of R2 means that extensive recoding of the signalling information has to be carried out in the transmultiplexer. However, this recoding can be arranged for all 60 channels by adding just two control units containing microprocessors. The general nature of processor control means that adaptation can also be made to other signal conversion requirements.

The transmultiplexer is transparent to channel-associated inband signalling. A variant of the transmultiplexer without signalling equipment is available for networks with inband signalling.

Common channel signalling with a capacity of 64 kbit/s is normally not possible on circuits containing transmultiplexers, since the transmission capacity of a transmultiplexer channel is limited to what the band 300–3400 Hz permits. This applies for all transmultiplexers, regardless of design approach. However, common channel signalling at a reduced rate can be used on a speech channel via modems.

The transmultiplexer with no signalling equipment offers economical advantages in networks where common channel signalling is used and the signalling is transmitted via a separate circuit.

Reliability

Transmission equipments require a high degree of reliability. Breakdowns in the long-distance network, with the consequent loss of traffic, can be very expensive. Poor reliability in the other parts of the network can lead to unacceptably high maintenance costs.

Ericsson has developed several generations of FDM and PCM systems. One of the objectives in this development work has been to obtain high reliability. This aim has been met by adhering to strict design rules and by careful choice of components. At the same time the number of components has been kept low. The power dissipation has also been kept low, and since this gives a low operating temperature it is another factor contributing to high reliability. The reliability of the transmultiplexer is expected to be very high since most components are well-proven types that have been used in the individual FDM and PCM systems.

The structure of a system has an effect on its function. For example, it determines the consequences of different faults. The structure of Ericsson's transmultiplexers is such that very few items are common to many channels. The probability that a fault will affect several channels is low, and the probability of a total breakdown is extremely low. Since the transmultiplexer is thus built up of system components with very high reliability and has been given the most adequate structure, the prerequisites have been created for very good operational reliability of the equipment. This is confirmed by the calculations of the MTBF (Mean Time Between Failures) that have been carried out.

Flexibility

Many administrations use the supergroup as the smallest extension unit, which gives many advantages as regards handling and administration. However, the group may be a more suitable basic unit in certain network configurations and for special temporary requirements. In Ericsson's transmultiplexer the groups are therefore accessible both towards the TDM and the FDM side. A program channel, high-speed data modem, through-connection equipment etc. can be connected towards the FDM side.

The equipment can be made transparent to a channel having a transmission speed of 64 kbit/s. This means that common channel signalling can be used over a transmultiplexer circuit. For example, remote subscriber stages for AXE 10 can be connected over a carrier circuit via transmultiplexers supplemented by external modems, fig. 11. This can solve certain network problems, for example in large-mesh rural networks. The transmission capacity required for the remote subscriber stage is often no more than 30 channels, and thus the loss of the 12 channels that are needed for the 64 kbit/s transmission is quite acceptable. The necessary synchronization information is transmitted via the data modem.

Individual speech channels are also accessible and can be used, for example, for connecting data modems in either direction.

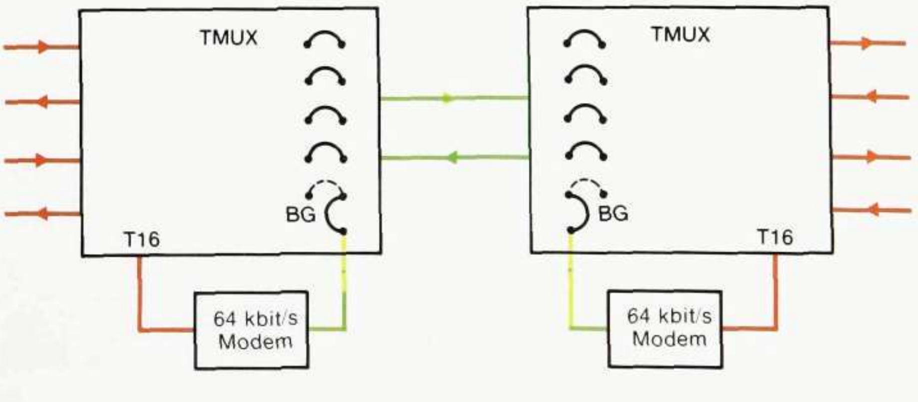


Fig. 11
Transmission of a synchronous 64 kbit/s channel over a carrier system via transmultiplexers

T16 64 kbit/s interface
BG Basic group interface

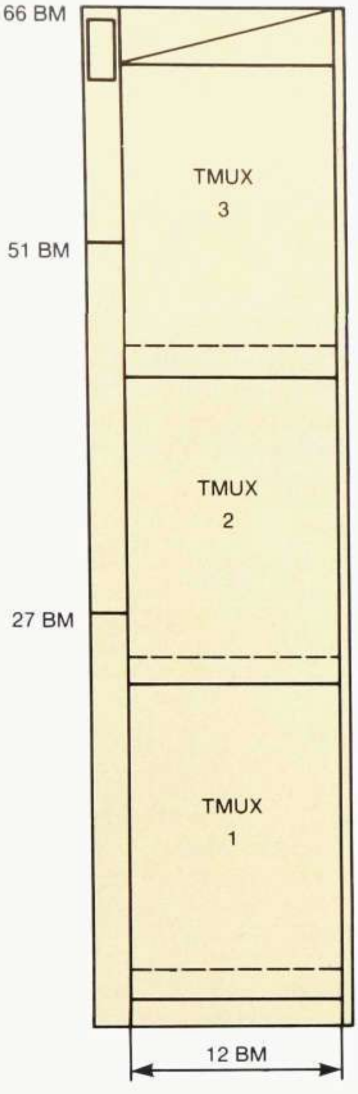


Fig. 13
The M5/BYB bay equipped with three transmultiplexers for 60 channels
BM Building module

For traffic *towards* the TDM side units can be provided which permit the connection of eight 64 kbit/s data channels, in place of eight of the speech channels.

The data channels, which give an extremely large data transmission capacity, can for example be used to transmit data bit streams from a digital data multiplexer. Such a unit can be connected in during operation without disturbing other traffic. The available bits in timeslot T0 are also accessible.

The transmultiplexer can be partially equipped to suit lower capacity requirements, resulting in lower costs.

For AXE 10 it is possible to connect a variant of the transmultiplexer direct to the digital group selector. In this case the ETC (Exchange Terminal Circuit) is not required in AXE 10, and the transmultiplexer can be integrated into the structure of the system, which is advantageous from the point of view of handling and maintenance.

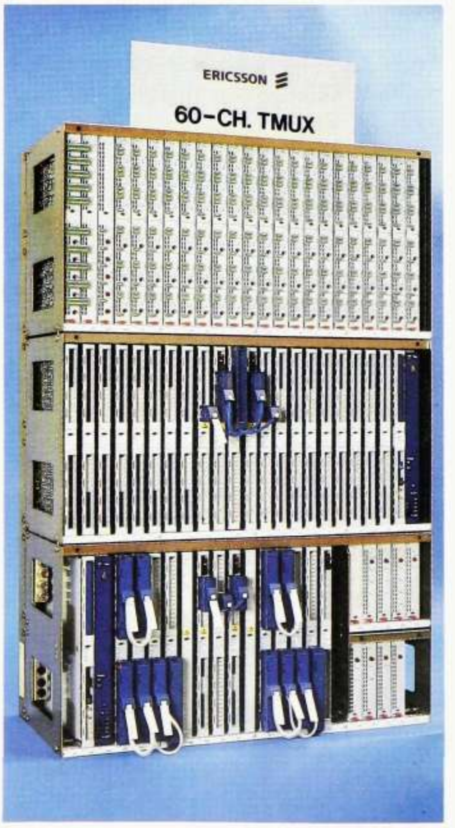


Fig. 12
ZAJ 2×30/60, transmultiplexer for 60 channels

Operation and maintenance

Ericsson's transmultiplexers do not require any preventive maintenance. The system parts are designed so that very high stability is obtained throughout the life of the equipment.

The equipment contains alarm monitoring and alarm transmission in accordance with CCITT recommendations. All alarm information is available in a special alarm unit. This unit also transfers some alarms between the FDM and TDM sides. All transmultiplexer alarms can be allocated to the urgent or non-urgent category by means of straps in the alarm unit.

Reminder indication can be initiated by means of a push-button. When an alarm occurs, a light emitting diode on the front of the unit lights. Several units in the subsystems are also equipped with light emitting diodes, which simplify fault tracing. The alarm unit also contains service alarm outputs, which provide alarm information for the whole equipment as well as individual subsystems at carefully chosen levels. The alarm outputs can be connected to alarm equipment of the traditional type or to Ericsson's computer-controlled alarm system, ZAN 101.

Detailed measurements can be carried out from various test points. The FDM part contains short-circuit proof test points at important system points and also access points for connection to the speech frequency and signalling interfaces. The PCM part contains short-circuit proof test points for, for example, the transmitted and received bit stream and the transmitter and receiver timing. On the FDM receive side manual level regulation can be carried out individually for each channel as well as for each basic group. This means that the residual level deviations, such as those caused by a complex network configuration, can be compensated. Incoming group or supergroup pilot frequencies can be monitored for interruptions by a pilot receiver, which can be included in the equipment. The frequency of the send clocks can be adjusted. Incoming TDM bit streams are supervised for loss of timing and bit error rate. All test and measuring points are readily accessible at the front of the equipment.

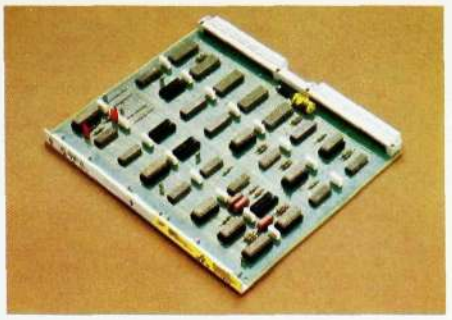


Fig. 15
Control unit with microprocessor

The basic principle of the transmultiplexer as regards alarms is that it should behave as a PCM multiplexer towards the TDM side and as a carrier system towards the FDM side. There are certain extra features, however, such as the transmission of the AIS (Alarm Indication Signal).

Fault-clearing maintenance consists of changing faulty printed board assemblies, and a suitably dimensioned stock of spare parts must therefore be available. The conventional structure of Ericsson's transmultiplexers means that many of the boards are also used in other Ericsson multiplex equipments, so the stock of spares can be shared, with a consequent reduction in maintenance costs.

System structure

The description above applies in the relevant parts for all Ericsson transmultiplexers. The mechanical construction and functional design of three different equipments are described separately below.

ZAJ 2×30/60

The transmultiplexer for 60 channels, ZAJ 2×30/60, fig. 12, converts two 2048 kbit/s PCM streams to a standardized basic supergroup in the frequency band 312–552 kHz and vice versa.

The equipment is built up using Ericsson's BYB construction practice. All equipment for the transmultiplexer 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 two d.c./d.c. converters. The magazine can be mounted in M5/BYB bays[2], in BYB rows or in a BYB cabinets. All cabling is accessible from the front. An M5/BYB bay can be equipped with three transmultiplexers, fig. 13.

As far as possible the transmultiplexer has been built up of the same parts as Ericsson's other multiplexer systems. The functional structure is shown in fig. 14. The TDM part contains two identical PCM units. These subsystems contain well-tried system components which also form part of AXE 10 as well as separate PCM equipments[3]. The FDM part contains units from Ericsson's FDM systems[4]. The continuous technical development also affects the FDM field. For example, hybrid type thick film circuits have been replaced by monolithic circuits in the signalling circuits, which means that it has been possible to reduce the number of integrated circuits by a third on the send side and by half on the receive side. The necessary carrier and signalling frequencies are generated internally with the aid of phase locked oscillators, which are controlled by an incoming basic frequence of 12 or 124 kHz. The group pilot frequencies, 84.08 or 104.08 kHz, can either be generated internally or obtained from an external source. The supergroup pilot frequency, 411.92 or 547.92 kHz, is obtained from an external source. FDM terminals normally contain central fre-

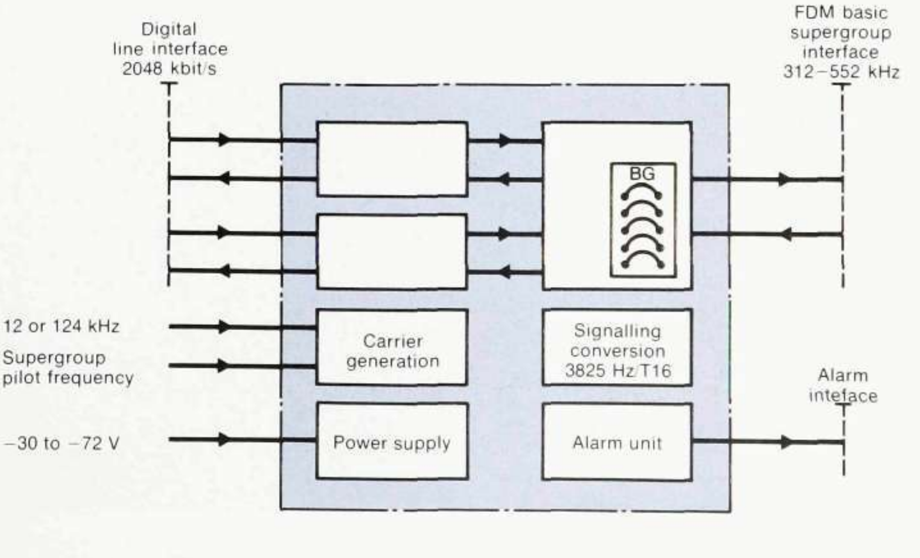


Fig. 14
Block diagram of the transmultiplexer ZAJ 2×30/60

BG Basic group interface

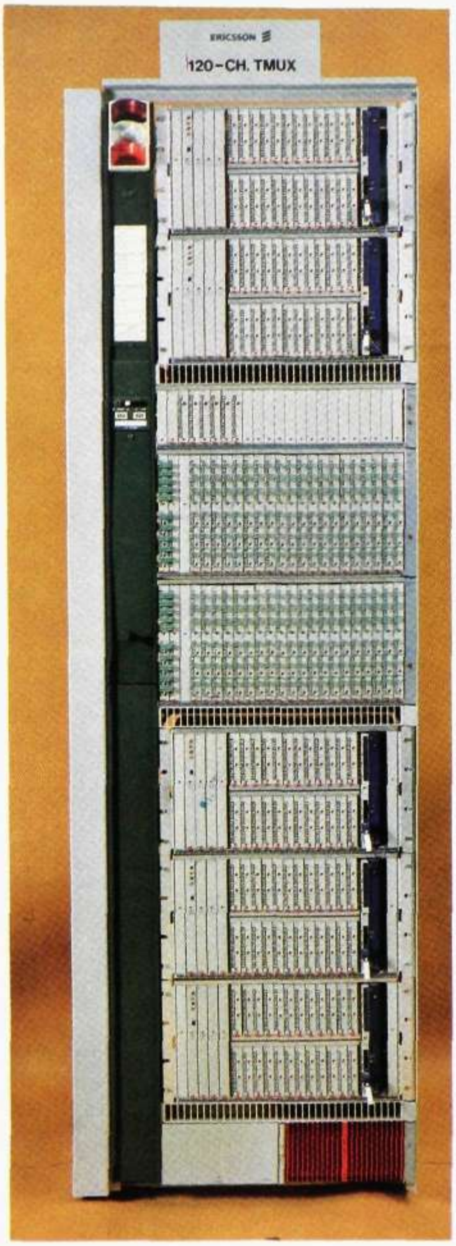


Fig. 17
ZAJ 5×24/2×60, transmultiplexer for 120 channels

quency generating equipment for basic and pilot frequencies.

The conversion between the outband signals of the FDM side and the bits in time-slot 16 of the TDM side takes place partly in the channel units in the FDM part and partly in units from Ericsson's latest equipment for E&M signalling, ZAK 01-3[5]. The signalling part can be supplemented by a control unit, fig. 15, which carries out the necessary recoding between the analog and the digital version of the CCITT signalling system R2.

The transmultiplexer is powered by two d.c./d.c. converters which are fed from a battery voltage of between −30 and −72 V.

The equipment contains an alarm unit which collects all alarms and generates service alarms etc. at different system levels. The unit also handles the transfer of alarms between the FDM and TDM sides. The alarm unit is provided with the standard alarm interface for transmission equipments in th BYB construction practice.

ZAJ 24/2×12
The transmultiplexer for 24 channels, ZAJ 24/2×12, fig. 16, converts a 1544 kbit/s PCM stream to two standardized basic groups in the frequency band 60–108 kHz and vice versa.

The equipment is built up using Ericsson's BYB construction practice but is intended for rack mounting in the standard 19″ construction practice. All equipment for the transmultiplexer is mounted in a pre-wired double magazine having a height of 12 building modules (488 mm) and a width of 19″ (483 mm). The magazine occupies 11 mounting spaces in a standard 19″ rack. The magazine is an independent unit and contains two d.c./d.c. converters.

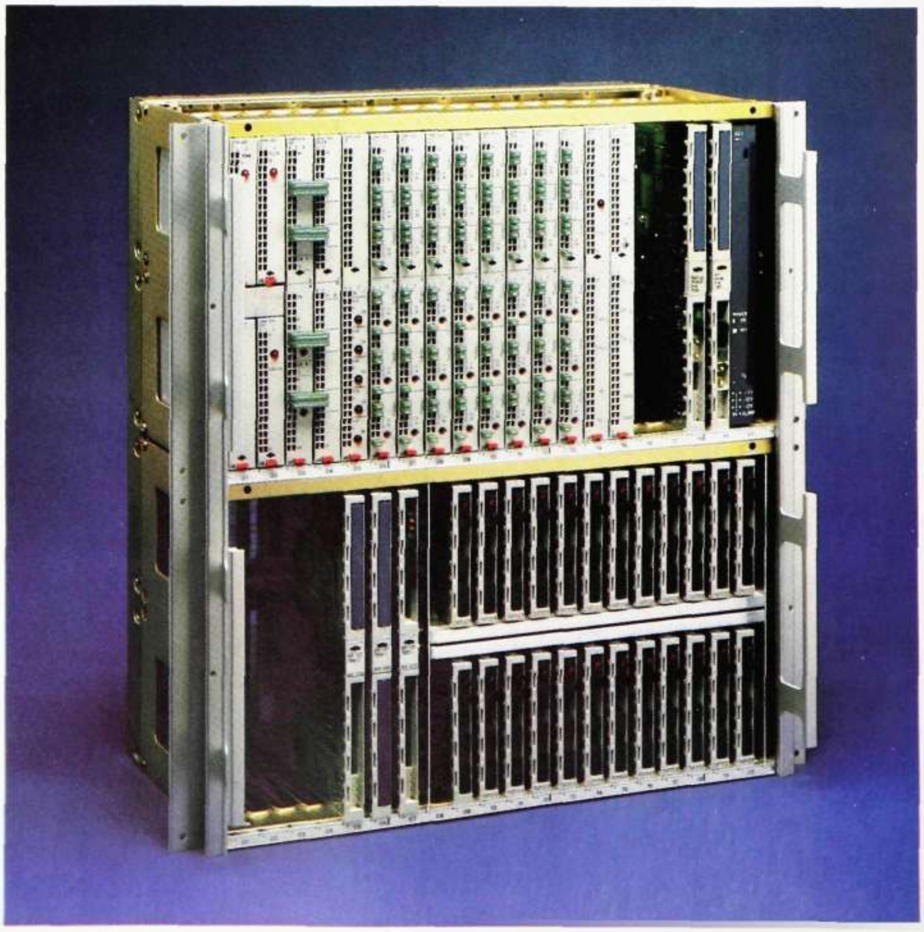


Fig. 16
ZAJ 24/2×12, transmultiplexer for 24 channels

Fact panel

ZAJ 2×30/60

Capacity	60 channels
Digital interface	2048 kbit/s, HDB3, A-law
Analog interface	Basic supergroup, 312–552 kHz

ZAJ 24/2×12

Capacity	24 channels
Digital interface	1544 kbit/s, Bipolar, μ-law
Analog interface	Basic group, 60–108 kHz

ZAJ 5×24/2×60

Capacity	120 channels
Digital interface	1544 kbit/s, Bipolar, μ-law
Analog interface	Basic supergroups, 312–552 kHz

The equipment is built up of the same system components as Ericsson's FDM and D4 systems. These components are characterized by a high degree of reliability and low power consumption. The ordinary transmultiplexer contains equipment for conversion between PCM signalling and FDM outband signalling at 3825 Hz, but a version without signalling equipment is also available. All necessary pilot, signalling and carrier frequencies are generated internally, with the aid of a built-in quartz oscillator. Alternatively it is possible to control the equipment by means of an externally generated basic frequency, 12 or 124 kHz, injected via an auxiliary unit using the phase locking technique.

The transmultiplexer is powered by two d.c./d.c. converters, which are fed with a battery voltage of between −30 and −72 V. The equipment is provided with several alarm circuits which monitor both internal and external functions. Alarms are indicated by light emitting diodes on the front of the unit, which simpifies fault tracing.

ZAJ 5×24/2×60

The transmultiplexer for 120 channels, ZAJ 5×24/2×60, fig. 17, converts five 1544 kbit/s PCM streams to two standardized basic supergroups in the frequency band 312–552 kHz and vice versa. The group interface is also accessible.

The equipment is built up in Ericsson's M5 construction practice and is mounted in a bay having a height of 51 building modules (2134 mm). The bay is constructed for use also on data floors, and it is thus possible to run the cabling to the bay via the base plate.

The equipment is built up of Ericsson's well-proven system components. All wiring between the various subsystems in the bay is completed during the manufacture, and hence it is simply a question of plugging in towards the FDM part when installing. The transmultiplexer is an independent unit, with d.c./d.c. converters and carrier frequency generation. The basic and pilot frequencies are obtained from the central frequency generating equipment of the terminal. The subsystems contain equipment that handles the signal conversion from outband signalling at 3825 Hz to PCM signalling and vice versa. The transmultiplexer is equipped with the normal alarm and supervision functions.

The transmultiplexer can be equipped with pilot receivers for interruption control working on the group or supergroup pilots.

The transmultiplexer for 120 channels offers economic and handling advantages, among other reasons because the MDF equipment at both the channel and group level can be dispensed with. This transmultiplexer has been developed at the request of customers who use the supergroup as the smallest extension unit.

Conclusion

The transmultiplexer will be in use during the foreseeable future for various types of through-connection between FDM and TDM networks. The efficient transmission and well defined interfaces of the transmultiplexers make them simple and flexible enough to be used in many different stages of the current digitalization process.

Ericsson's transmultiplexers, which are built up of system components that have been developed and tested through several generations of equipments, are all flexible and have a high degree of reliability. These characteristics, together with their low maintenance costs and good performance, enable the transmultiplexers to become functional and economic components in telecommunication networks.

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ERITEX 10 for Teletex and Word Processing

Thomas Augustinsson and Björn Sjöstrand

Ericsson Information Systems AB has developed an electronic word processor with facilities for teletex communication, ERITEX 10.
The authors discuss the need for and purpose of this type of machine and describe the system structure and the design of the various units that make up the system.

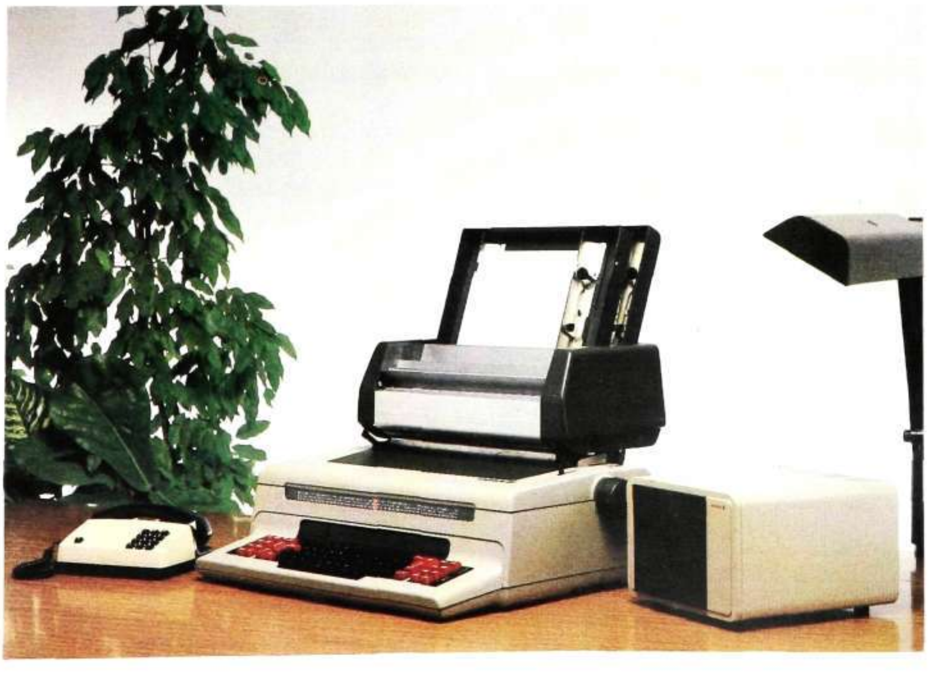
UDC 654.145
651.9:681.3

Teletex is a new international text communication service which has recently been standardized by CCITT. Teletex makes it possible to transfer documents between data memories in terminals that are connected via telecommunication or data networks.

All teletex communication is carried out memory to memory, which means that the transmission of documents takes place very quickly. Local work at the terminals, such as input and printout, is not disturbed when sending and receiving takes place, fig. 1.

The introduction of special interface equipment between teletex and telex networks makes it possible to offer communication with all national and international telex terminals as a part of the teletex service. In several respects teletex offers improved facilities for text communication compared with telex. Not only is the transmission speed approximately 30 times higher than for telex, but teletex also makes it possible to transmit most national alphabets, and this gives the received printout a high quality.

Fig. 2
ERITEX 10, electronic typewriter for teletex communication and word-processing



ERITEX 10 is the first member of a new family of office equipment. It can be used as a word processor and as a terminal for office communication, fig. 2 and 3. ERITEX 10 consists of an ergonomically designed keyboard, a line display with 40 characters, a daisy-wheel printer with 105 characters and circuits for connection to an external DCE (Data Circuit terminating Equipment). The electronic typewriter ERITEX 10 is

- a teletex terminal for external communication
- a terminal for internal communication
- a word processor with an electronic memory
- an electronic archive.

The teletex communication in ERITEX 10 is in accordance with CCITT Recommendations F.200, S.60, S.61, S.62 and S.70. ERITEX 10 can be connected to a teletex service which uses a data network for circuit switching (CSDN), a data network for packet switching (PSDN) or the public telephone network (PSTN). Teletex communication can take place either via public telecommunication networks or via private networks. It makes no difference to the user whether ERITEX 10 is connected to a data network or a telephone network.

By internal communication is meant traffic with other ERITEX units or with other terminals within an organization. The communication can take place via a PBX or a local data network.

ERITEX 10 used as a word processor offers a *large number of editing functions* for altering, adding, removing and transferring text. The word-processing functions are supported by an extensive internal text memory for documents and messages, together with memories for different formats, for frequently used phrases and for the last line entered. Some of the usual word-processing functions in ERITEX 10 are:

- indented left-hand margin
- straight right-hand margin
- automatic carriage return, wrap-

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Ericsson Information Systems AB

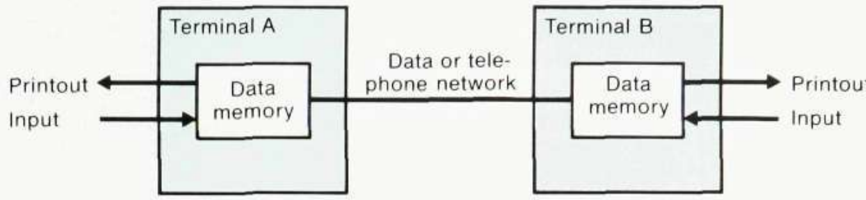


Fig. 1
The basic principle for teletex communication

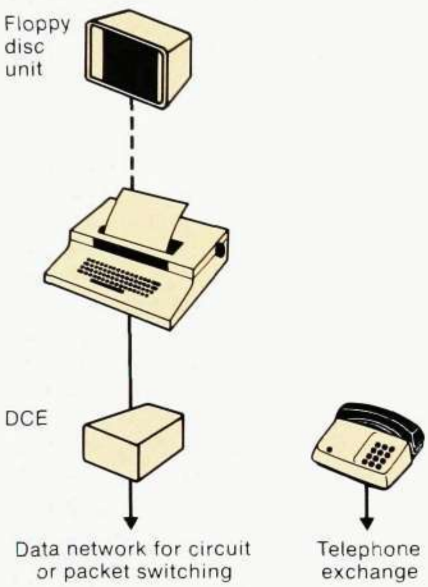


Fig. 3
The ERITEX 10 terminal with a floppy disc unit containing a memory for 1.2 million characters, a data modem DCE for teletex communication and a telephone set

- page riffling
- division into pages
- joining two pages into one
- tabulation
- centering
- underlining
- bold type
- stop codes
- memory function
- deletion
- choice of document type.

An electronic archive in the form of a separate floppy disc unit can be connected to ERITEX 10 when large amounts of text have to be stored. The unit holds two 5 1/4" floppy discs, each with a storage capacity of approximately 300 pages. The floppy disc unit also includes a file management system, and the operator can therefore easily open, define, store and read files. Documents can also easily be copied from one floppy disc to another in the floppy disc unit.

System description

Microprocessors are used to perform the various functions in ERITEX 10. The system is described with reference to the block diagram in fig. 4.

The four motors included in the printing and striking mechanism are controlled by servo circuits and power amplifiers. The necessary sensors are controlled by their own microprocessor (motor control unit), which is connected to the main system bus via a parallel/series converter (USART).

The central processing unit (CPU) consists of a microprocessor with the associated memories, fig. 5. It handles interrupt signals, keyboard functions and store functions. This CPU controls the flow of information on the main system bus and the execution of certain pro-

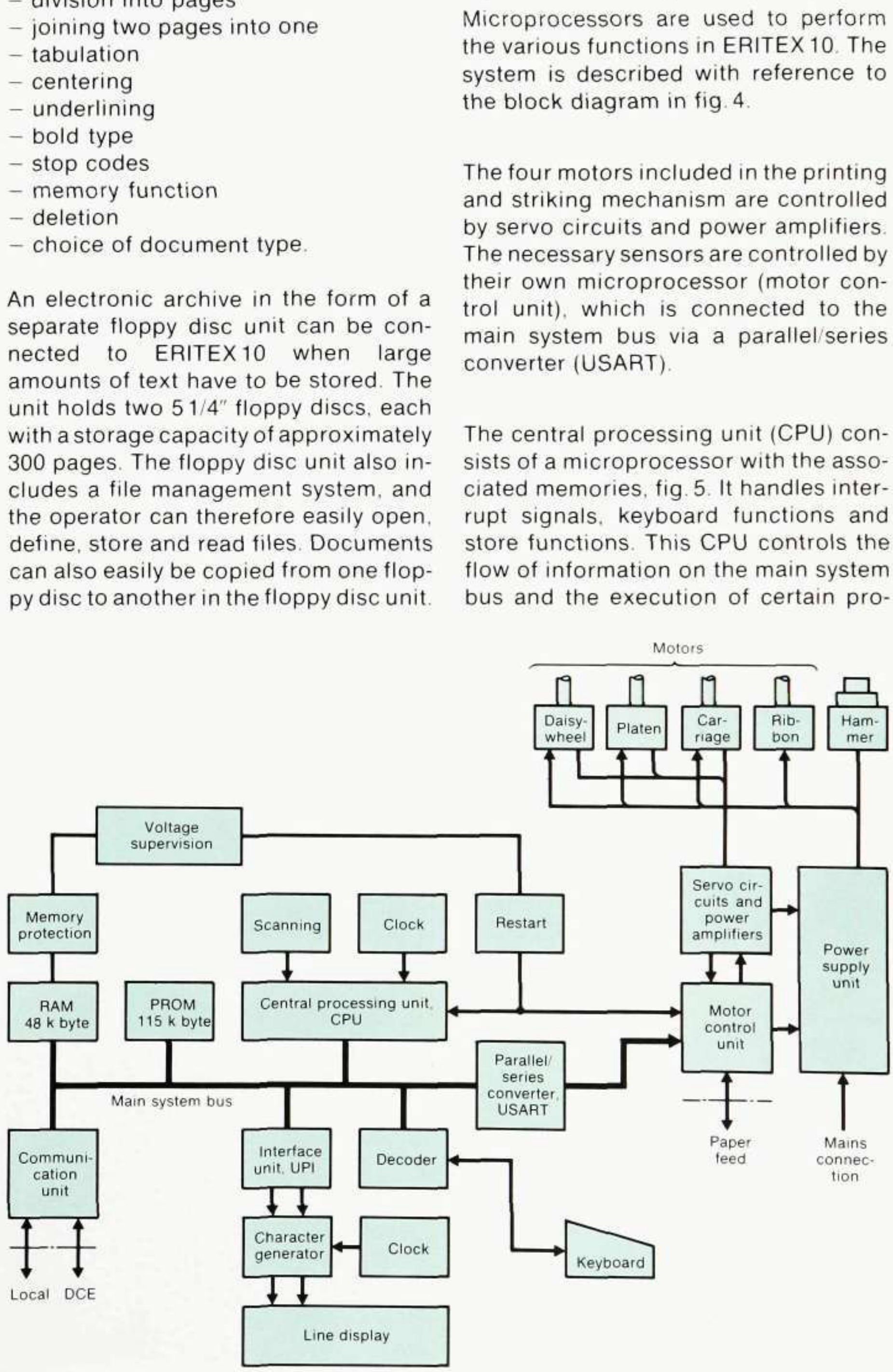
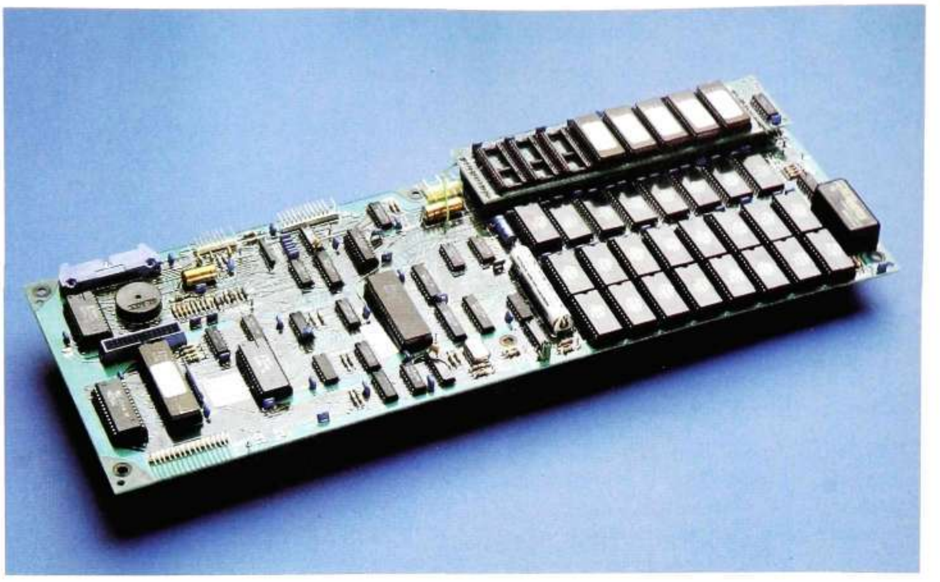


Fig. 4

Fig. 5
The central processing unit in ERITEX 10. The printed board contains a microprocessor and RAMs and PROMs



grams that are stored in the programmable read-only memory (PROM). The random access memory (RAM), which has standby battery power, is divided into a line memory, a format memory, a phrase memory and a text memory.

When the terminal is switched on, the line memory is automatically activated. This memory is emptied of its contents when automatic or manual carriage return is initiated. The information is transferred to the text memory. The line memory holds 350 characters.

The format memory holds information concerning margins, tabulations, first line of writing etc.

The phrase memory is used to store frequently used expressions, which can then be retrieved and inserted in the text. The storage capacity, together with format information is 1000 characters.

The fourth and largest primary memory, the text memory, is used for temporary storage of received and input documents. Its storage capacity is 42 k byte, which corresponds to 20–25 normal pages.

Communication unit

The communication unit is connected to the main system bus. It comprises all necessary circuits for connection to a data circuit terminating equipment, DCE, and the circuits required for connection to a secondary memory. It also contains a two-wire interface for communication with other ERITEX units.

Line display

The front of the machine includes a line display unit, fig. 6, above the keyboard, where a limited part of the stored text can be shown. The display makes it possible to check the input text. It is also a help in correcting and adding to printed text, as well as in making up pages.

Five of the 40 characters in the line display are used for state information, i.e. information regarding the operating

Fig. 6
The line display holds 40 characters, of which 34 are used for text and 5 for state information. The keyboard is of a low profile type and ergonomically designed. Approximately 300 characters can be obtained by means of different key combinations

mode of the machine and which functions are activated. Of the remainder, 34 positions are used for text, and one for separating the line of text from the state information. It is possible to display the name of the document and its contents, as well as riffling the pages. All texts defined in accordance with CCITT Recommendation S.61 can be shown on the line display.

The line display has its own character generator and is connected to the bus via an interface, UPI (Universal Peripheral Interface).

Keyboard

The keyboard has been designed in accordance with modern ergonomic principles, fig. 6. It is of a low profile type, with an average height of only 59 mm from the table to the keys. This enables the operator to work in the correct sitting position.

The keyboard comprises all keys and light emitting diodes required to control and use ERITEX 10. It also contains a buzzer with two different signals. These signals are used for acknowledgement or fault indication. ERITEX 10 has alphanumerical keys and function keys, with a layout in accordance with the international standard for an ordinary typewriter.

The function keys are grouped outside the alphanumeric keys in such a way that they are easy to find and combine with other keys. The top row of alphanumeric keys is also used for functions in combination with a command key. The alphanumeric keys give direct access to 105 characters, but up to 308 characters can be obtained by combining keys.

At the ends of the keyboard there are four controls with associated light emitting diodes for setting
- operating mode
- line spacing
- key pressure
- character spacing.

The keyboard is connected to the main system bus via a special processor (keyboard CPU).

Printing unit

The printing unit is of daisy-wheel type and gives a printing speed of 16 characters per second. The printer can be equipped with different wheels. Each wheel has a set of 105 characters. An even greater range of characters can be obtained by combining these. A special teletex wheel is used for international teletex traffic. Most characters in the languages originating from Latin can be written with the teletex wheel.

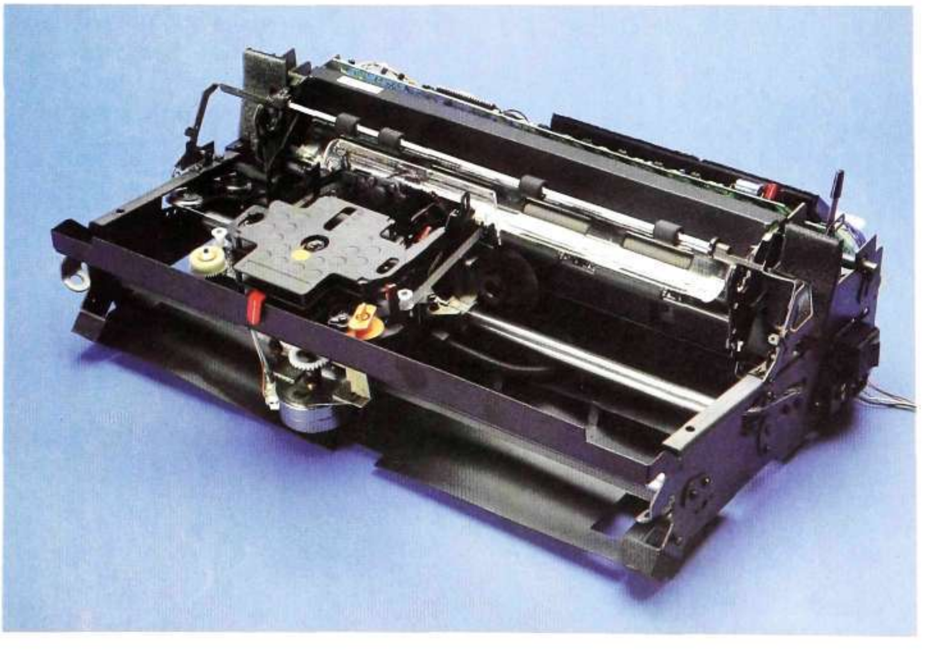


Fig. 7
The printing unit can be equipped with different daisy-wheels, including a teletex wheel for international traffic. The printing speed is 16 characters per second

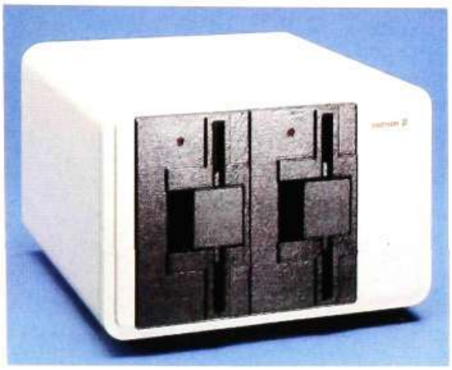


Fig. 8
The floppy disc unit with two 5 1/4″ floppy discs. It has a total capacity of 1.2 million characters

Secondary memory
The secondary memory is a separate floppy disc unit for two 5 1/4″ floppy discs, fig. 8. Each disc holds some 600 000 characters, which corresponds to approximately 300 pages of size A4. The reading and writing in the secondary memory is controlled by a file management system and a processor in the floppy disc unit.

Paper feeder
Two types of paper feeder are available for the ERITEX 10 terminals, a sprocket drive feeder and a cut sheet feeder, figs. 9 and 10. The maximum printing width is 165 characters, allowing the paper to be used for lengthwise or transverse formats. The appropriate paper feeder is simply placed on top of the terminal without needing any special fixing arrangements.

Software

The software in ERITEX 10 is divided into a number of process-orientated programs, which interwork with the aid of signals. The exchange of signals between the various programs is controlled by an operating system. The most important parts of the software are word-processing, file management and teletex communication. Fig. 11 shows a block diagram of the software structure.

OPERATING SYSTEM
The main program in the operating system supervises and controls the program handling by means of signals from different buffers. Signals to these buffers are generated by an interrupt routine in the operating system, which scans different units and circuits in the hardware.

WORD PROCESSING SOFTWARE
The programs in the word-processing software scan the keyboard, feed the information to the line display and printer, interpret the commands given by the operator, fetch, edit, store and erase text in the internal memories and check the codes and characters. In addition to certain basic programs there are programs for
- editing
- interpretation of commands
- code checking
- store management
- printout.

Basic programs
The line display, keyboard and printer are all controlled by their own programs for code conversion, interpretation of input characters and output of characters via the buffers of the units.

Editing
The main editing program works one page at a time in the indicated document and is responsible, with the aid of the store management program, for setting up, storing and erasing pages and documents. A page can also be divided into two, and two pages can be joined into one. Text pages for editing are transferred from the main memory to the editing memory, where the editing is carried out.

The editing program also ensures that the relevant command is shown on the line display.

Fig. 9, left
Cut sheet feeder

Fig. 10, right
Paper feed with sprocket drive

Command interpretation
Each command received by the editing program from the basic program for keyboard scanning must be interpreted. This is done by a special program, which is initiated by the key COMMAND.

Code checking
A program checks that every input character conforms to the format and code rules that apply for the type of document in question. This program can also be used to change a document or a page to another type of document.

In the case of teletex and telex, the program also checks that the set of characters and codes are correct for the specified format. In addition the program carries out the necessary checks when two document pages are joined into one.

Store management
ERITEX 10 has an internal text memory for storing and editing. This is divided into an editing memory and a communication memory. The editing memory is used by the basic programs as well as the editing program for line and page editing, and also by the printing program for the printing of pages. The communication memory is used by the store management program for storing, retrieving and erasing documents and individual pages, and by the teletex program for the storing and handling of transmitted and received documents. These two memories share a common storage area. The memory sizes and the boundary between them can vary.

The store management program contains functions for the administration of these memories, such as

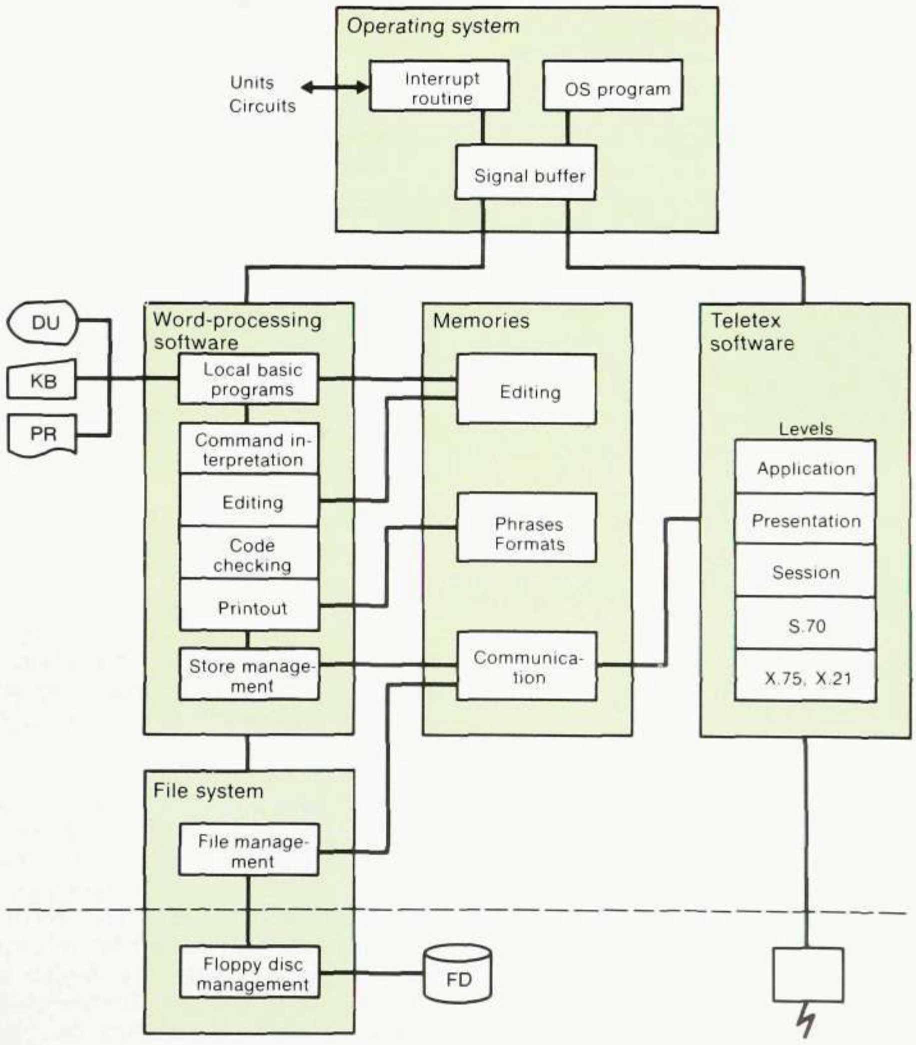


Fig. 11
The software structure in ERITEX 10

- finding space for a document or a page
- releasing areas when a document or a page is erased
- searching for documents by their name
- checking that document names are not duplicated
- packing and reorganizing the information in the memory.

The phrase and format memories are used by the basic programs and the editing program for the storing and handling of phrases and formats.

Printout

The printing program prepares and checks the printout of the document page or the document selected by the operator. The document page to be printed is transferred to the editing memory if it is not already stored there for editing.

Printout of documents that are stored in the floppy disc memory takes place only after each page concerned has been transferred to the editing memory via the communication memory.

If the communication memory is not sufficient for the number of received documents, the information that cannot be accommodated is automatically printed. Each printed document page is normally retained in the memory concerned, with the exception of any received pages that are automatically output when the whole storage area has been used up.

FILE MANAGEMENT SOFTWARE

The file management software, which carries out the storing, retrieving and other handling of documents, is divided into two parts. One part is placed in the terminal and is controlled by its operating system, and the other part is placed in the microprocessor of the floppy disc unit.

Terminal part

The transmission of data and interrupt signals takes place over the bus between the terminal and the floppy disc unit. The interrupt routine of the operating system checks when the information can be transmitted via the bus system to the file management software, and gives priority to teletex communication.

The file management software in the terminal part compiles and forwards pages and documents from received teletex messages, and transfers them between the communication memory and the secondary memory where they are automatically stored on floppy discs.

Floppy disc part

The software in the floppy disc part consists of a number of programs that carry out and check such functions as
- creating, opening, reading, erasing, changing the name of and copying documents
- reading or printing blocks of data.

A program controls the transmission of information at the interface to the floppy disc hardware. Another program controls and supervises the signal buffers.

TELETEX SOFTWARE

The teletex communication software carries out the procedures defined by CCITT for the teletex service.

The teletex software is divided into levels in the way defined in the OSI (Open Systems Interconnection) model and conforms to CCITT recommendations for the international teletex service (F.200, Basic Teletex Service). It is divided into programs for the following levels:
- application
- presentation
- session
- transport (S.70)
- link and network (X.75 and X.21).

Application level

The application program checks the given commands for sending or receiving, and initiates the actions to be taken by the store management program and other programs. When the document is stored on a floppy disc the file management program is also initiated when necessary.

The information to the operator concerning the transmission and reception in progress, any faults etc. is controlled by this program, which also handles logging into an internal log file, where information is stored regarding transmitted and received teletex messages. The log file can only be erased when output has been correctly effected.

Presentation level
When a document is transmitted, the presentation program divides the text into suitable parts in accordance with CCITT Recommendation S.62, and handles signals and acknowledgements from lower levels. Retransmission can then be requested. Information is given to the higher application level concerning correct or faulty transmission.

When text is received, the presentation program compiles the different portions of text, page by page, to form a coherent document. For each page the sending terminal is informed as to whether the received text is correct or calls for retransmission. Information regarding the start and end of the document is also sent to the application level.

In addition to the functions recommended by CCITT the presentation program includes functions for
- the administration of data blocks in the buffers
- changing special control characters and codes in the text to formats in accordance with CCITT recommendations
- separating the format row from each document page.

Session level
At the session level the calling and called terminal exchange information regarding their identity, and which of the standardized auxiliary functions the sender wants to use and the receiver can handle. The ERITEX 10 session program includes the standard functions for Basic Teletex Service in accordance with CCITT Recommendation S.62.

Transport level (S.70)
At the transport level ERITEX 10 takes care of various functions connected with the data transmission that are not dependent on the type of network, for example
- establishing and identifying the circuit
- compiling data into blocks
- segmenting or recompiling blocks of optional length received from the session level
- detecting and reporting procedural faults.

Link and network level (X.75 and X.21)
The software at the link and network level handles the setting up, supervision and disconnection of connections with an external DCE, and ensures that all information obtained from higher levels is transmitted faultlessly and in the right sequence.

The software is made up of programs in accordance with CCITT Recommendations X.75 and X.21. One program controls the connection to circuit switched data networks which work to these recommendations. If this program is changed for another, the terminal can be connected to the telephone network or packet switched data network. No other programs have to be changed. Fig. 12 shows the relation of the functions to different recommendations.

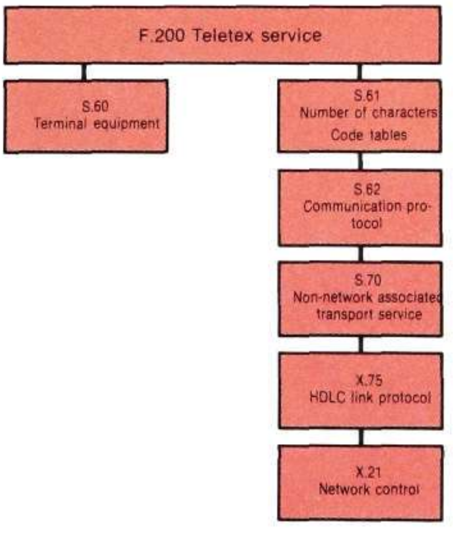


Fig. 12
The structure of the teletex communication software conforms to CCITT recommendations for teletex

Service and maintenance

The only regular maintenance required for ERITEX 10 is cleaning and change of ribbon.

When the terminal is switched on, the control unit carries out a memory test. It also checks the connection to the keyboard and to the printer. An alarm is given if any fault is detected.

The power supply is monitored continuously. If the voltage falls below a certain level, the control unit prepares for changeover, and if the level falls below a lower threshold, the equipment is automatically switched over to battery operation.

The ERITEX 10 software includes special test programs for testing the operation of both the mechanical and the electronic units. If a fault occurs, these programs can be initiated by service technicians in order to locate the faulty unit. The various parts of the terminal are easy to replace.

Summary

The introduction of teletex constitutes an important advance towards the office of the future. In teletex an international method has been created for the transmission of documents between terminals of different manufacture. Teletex is therefore a basic service in Ericsson's integrated systems for office automation and office communication. ERITEX 10 is the first terminal from Ericsson that is equipped for teletex communication. It meets stringent demands as regards function, ergonomic design, quiet operation, integration with other products from Ericsson Information Systems AB and convenient servicing.

Fig. 13
ERITEX 10 terminal with teletex communication

Local Area Radio System —LARS

Hans Lindblad

SRA Communications AB, a member of the Ericsson Group, has developed a radio system for certain military applications. The system is designated LARS (Local Area Radio System) and is intended for radio communication within geographically limited areas, such as military bases, training premises and large store areas.

The author describes the functions offered by the system for stationary, portable and mobile stations and the technology used. The maintenance aspects, operational reliability and environmental factors are also discussed.

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UDC 621.396.7

Radio system LARS (Local Area Radio System) is used for communication between the commanding officers at the operational centre of an establishment and the staff, for internal communication between members of the staff and for connection to PBXs. The operational centre has permanently installed equipment, the other equipments are portable or designed for installation in vehicles or tents, fig. 1. The system also allows the stationary radio equipment at the command centre, with its efficient aerial system, to be used as a relay station between units placed at a distance from each other within the base area. The system also contains facilities for establishing links with adjacent base areas.

The system is specially designed to meet the requirements of military organizations. It offers full communication facilities between all users. A commanding officer who wants to use a certain radio channel can use his priority and enter a call between subordinates.

LARS enables staff to establish communication with each other and with commanding officers and thus use vehicles and personnel efficiently. Transport can quickly be directed to the correct place. Messages can be delivered to staff or units without delay. LARS therefore contributes to the smooth and efficient operation of an establishment.

Speech privacy can be ensured by means of digital ciphering equipment. The radio equipment has been prepared for the transmission of data (data rate 9.6 or 12 kbit/s). Transvertex, a subsidiary of SRA, has developed special encryption equipment which can be connected to the various radio units. This equipment is so light that it can be used together with the portable radio version.

LARS is flexible and easy to adapt to suit different applications and the varying requirements and needs of customers.

Fig. 1
Simplified model of LARS

- Direct traffic to and from the command centre (HQ)
- Other direct traffic
- Relay traffic

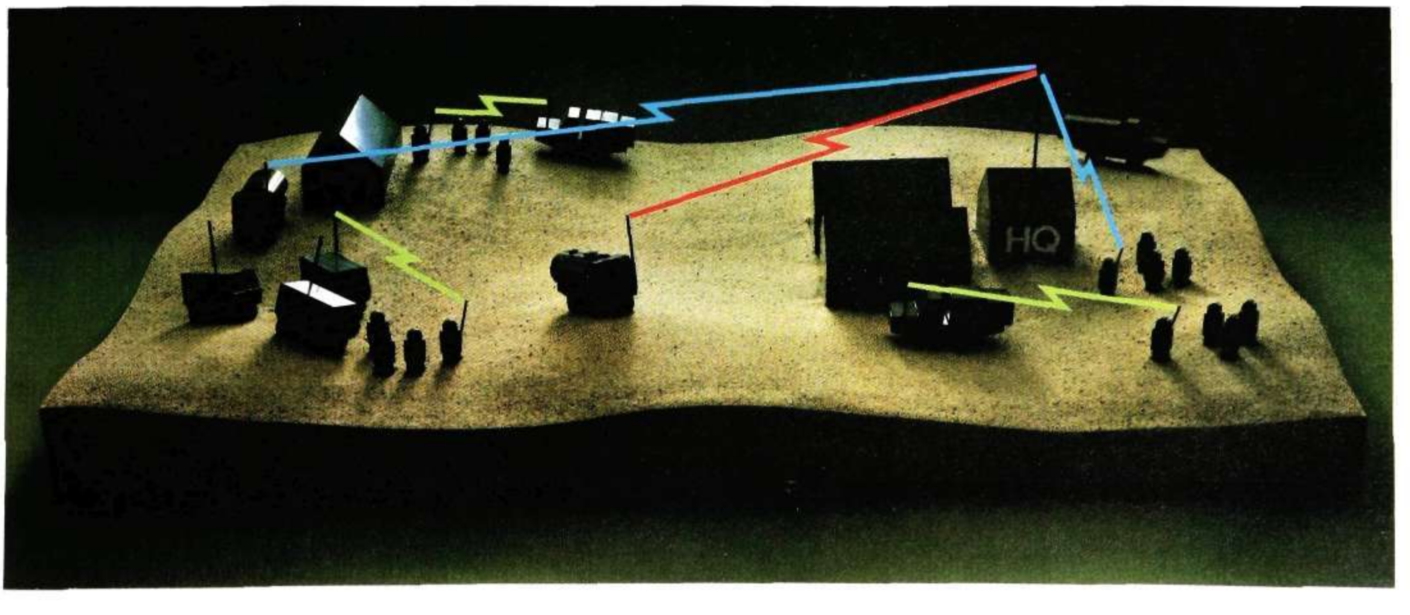


Fig. 3
A soldier equipped with the portable station

Stationary radio equipment

The stationary equipment of the command centre constitutes the central part of the system, fig. 2. It consists of control units, a central unit, filter unit, transmitter and receiver. All units are constructed for mounting in a 19" rack, which gives a high degree of flexibility and makes it easy to adapt the equipment to suit the relevant communication needs.

The control unit can be connected up to five stationary radio stations, which can be used simultaneously. Open or selective calling is chosen individually for each channel. Open calling (squelch control) means that the receiver is connected in for listening immediately there is a carrier on the channel. Selective calling means that the channel is opened for listening by means of the DTMFC (Dual Tone Multi Frequency Code) signal, and the operator need not listen to irrelevant traffic on the channel. Several control units can be connected to the stationary radio equipment. Each unit is served by two operators, one of which has priority.

The central unit in LARS is controlled by a microcomputer. It administers and supervises selective calling, the setting up of relay traffic, connection to PBXs, priority calls and other traffic functions.

Fig. 2
Control set for a command centre with parts of the stationary radio equipment mounted in a rack in the background

The central unit also controls automatic routines which test the system during operation and give an alarm if a fault is detected.

The stationary transmitter has an output power of approximately 35 W. A special power amplifier can be connected, which raises the output power to approximately 250 W.

All units in the stationary radio system are powered by 24 V d.c. from a battery-fed rectifier system. This makes the radio system immune to variations and temporary failure of the local mains supply and improves the system reliability.

Portable radio station

The portable radio station is simple to use and light enough to be easily carried, fig. 2. The radio unit is small and flat, and the microphone, the sound source and most of the operating devices are placed in a separate unit which is designed for one-hand operation. This makes the station ergonomically well suited to the special usages that apply for portable military equipment.

The station is powered by rechargeable NiCd cells which are mounted in a plug-in cassette.

The aerial consists of a quarter-wave blade aerial mounted on a flexible joint. This type of aerial gives a good compromise between manageability and aerial performance in the relevant frequency band.

Special carrying devices have been developed for different combinations with other equipment carried by the user (combat harness, hand weapons etc.).

This portable radio station is unique in its combination of small dimensions, robust structure, multitude of channels and system functions and large battery capacity.

Mobile radio station

The mobile radio staiton is a small, compact unit which is plugged into a cassette in the vehicle, fig. 4. The main advantage of this installation method is that the station can easily be moved

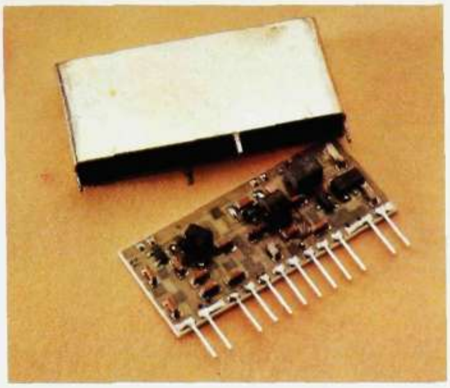


Fig. 5
One of the thick film hybrids that form part of the portable equipment

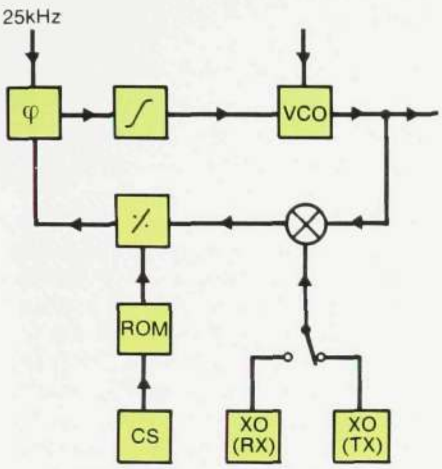


Fig. 6
Block diagram of the frequency generation

φ	Phase detector
∫	Integrator
VCO	Voltage controlled oscillator
÷	Frequency divider
⊗	Mixer
XO	Crystal oscillator
ROM	Read only memory
CS	Channel selector

Fig. 4
The cassette station (in the top right-hand corner)

from one vehicle to another. The power supply and aerial are connected up via connectors in the cassette. The installation accessories make it possible to change the position of the station without restriction.

Operating devices, indicators and terminals for connecting the hand-held microphone and ciphering equipment are accessible on the front of the unit.

The transmitter power of the mobile station is 15 W. The efficiency of aerials mounted on vehicles is normally very high, and a large coverage is therefore obtained.

The basic version of the station is designed for connection to 12 V d.c. (car battery). If necessary it can be supplemented for feeding from 24 V.

The cassette station can also be mounted in a magazine suitable for placing on a table. This type of mounting is particulary suitable for semi-permanent installations (i.e. in concrete dugouts and maintenance tents) where the same system functions are required as in a vehicle installation. The table magazine is equipped with a power unit, so that the station can be powered from the mains. A 12 V battery can be connected as a standby. If a mains failure should occur, the station is then automatically switched over to battey operation. The station can also be installed in a special waterproof carrying case containing relatively powerful accumulators.

If a portable radio station is needed, the portable equipment described above would of course normally be used. It has the same system functions as the cassette station. However, the cassette station in the carrying case has higher output power (15 W as against 1 W).

Modern circuit technology

Thick film hybrid circuits are used in the radio units, fig. 5. This technology has proved efficient for analog high-frequency equipment, particulary as regards miniaturization and structural modularity. All radio units, including the portable station, can be equipped for up to 100 different radio channels. The stationary radio equipment also has 20 channels intended for link traffic in another frequency band. The internal frequency generation is carried out by means of frequency synthesis. This means that a voltage-controlled oscillator (VCO) is phase-locked to a fixed crystal oscillator in such a way that its output signal, which is the working signal, is mixed with a signal from a fixed displacement oscillator. The resultant signal is divided down by means of programmable dividers and compared with a 25 kHz signal, fig. 6. The phase error is integrated and the resultant signal is used to bring the VCO frequency to the desired value in a 25 kHz raster. The actual frequency is decided by the programmable divider. The loop bandwidth of the servo system has been set so that frequency modulation can be carried out in the VCO circuit.

Different channel frequencies are obtained by different encoding of the divider. The group and channel settings of the stations are converted in a ROM (Read Only Memory) circuit to the code that defines the frequency. This circuit is unique for the frequency plan used. The ROM circuit must be changed if the channel spacing or individual frequencies have to be altered.

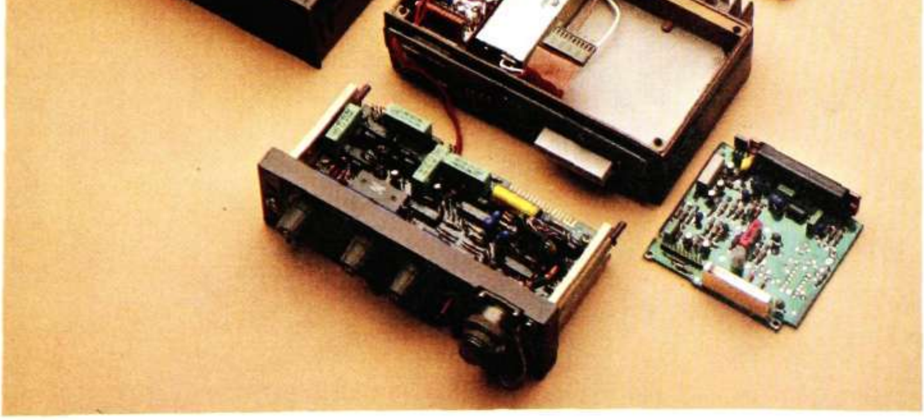
This type of frequency generation makes great demands on the design of the VCO circuit in the portable equipment, whose current consumption and volume must be small. Such parameters as transmitter noise, two-signal selectivity and microphony, i.e. unwanted modulation due to mechanical effects, can be directly related to the VCO.

The control computer in the central unit, which administers all traffic functions, is programmed to carry out advanced automatic fault testing. It performs plausibility analyses of, for example, the responses given by the system to external signals and control data. It also provides an outgoing combined alarm and a local indication of the type of fault.

The two-way transmission of state information between the control unit and the central unit takes place over a four-wire circuit, with the data transmitted in serial form. In this way approximately 100 wires are replaced by four. The control unit is equipped with a microcomputer which converts the state information to the appropriate form.

The standard DTMFC signalling is used for selective calling. The radio operator can thereby call staff at the command centre selectively, or activate functions in the system, such as setting up relay connections or connections to PBXs. The reasons for choosing this signalling system are its high sensitivity and high (crystal-controlled) frequency accuracy. Moreover, the complex digital signal processing circuits required for the system are available in the form of small, low-current LSI circuits.

Fig. 7
The cassette station partly dismantled



Maintenance aspects

Special attention has been paid to the maintenance aspects of the complete system as well as the components.

The radio equipment is designed for automatic testing. Frequency selection, keying, squelch disconnection etc. can be carried out electrically via a special connector on the unit, where the low frequency signals, internally generated voltages and squelch state can also be measured. The modulation in the stationary transmitter can also be determined by means of deviation measurements.

All the equipment has been given a modular structure with a view to simplifying service and maintenance, fig. 7. The electronic equipment is logically grouped on printed boards, which are connected via internal cables and connectors. Hybrid circuits constitute special function blocks in the miniaturized circuits. For maintenance purposes these blocks are considered as components.

The possibility of efficient maintenance of military radio equipment is extremely important and the customer often requires that the contract includes guarantees regarding the mean time to repair (MTTR).

Operational reliability

The operational reliability of the equipment has been assured by using only well-tried component types of good quality.

Certain stages in the production process include tests and measurements in accordance with a special quality control program. In addition, before the equipment undergoes its final factory testing it is subjected to a bump test and burning-in in accordance with a special program. The result of this is equipment with such high and even quality that the customers can be given guarantees of operational reliability in the form of MTBF values.

Technical data for LARS

General

Selective calling system	DTMFC signalling
LF interface between stationary units	600 ohms, balanced
Nominal level	−3.5 dBm
Transmission of states between control unit and central equipment	600 bauds serial data in a current loop
Temperature range	
Stationary equipment	+5 to +55°C
Mobile equipment	−40 to +55°C
Vibrations	
Stationary equipment	3 *g*
Mobile equipment	5 *g*
Bumps	
Stationary equipment	25 *g*, 1200 bumps
Mobile equipment	40 *g*, 6000 bumps
Power supply	
Stationary equipment	24 V d.c.
Cassette station	12 V car battery
Portable station	10.8 V NiCd battery

Radio units	Stationary radio	Portable radio	Cassette radio
Bandwidth	2.5 MHz	2.5 MHz	2.5 MHz
Frequency band	150 MHz	150 MHz	150 MHz
Bandwidth	0.5 MHz		
Frequency band	140 MHz		
Number of channels	100+20	100	100
Channel spacing	25 kHz	25 kHz	25 kHz
Traffic mode	one- or two-frequency simplex		
Modulation method	FM (6 dB/octave LF characteristic)		
Output power	35 W	>1 W	15 W
With a separate power stage	250 W		
Sensitivity (EMK) (12 dB SINAD)	0.6 µV	0.8 µV	0.6 µV
Battery life with normal operation	–	~10 h	~8 h (in carrying case)

Environment

The equipment is designed to the environmental requirements set by the Swedish Armed Forces for mobile and stationary radio equipment.

The specification of environmental conditions comprises temperature limits for operation, data for bump tests, vibration tests, static humidity tests and cyclic humidity tests. This applies for all units, but the requirements are different for stationary and mobile equipment. Moreover, the portable radio station and the cassette station installed in the carrying case also have to be waterproof.

In order to be able to meet the bump and vibration requirements the radio equipment, particulary the portable and mobile units, have been given a construction that makes them especially sturdy, robust and durable. The user is assured of a working radio even during the most difficult operating conditions.

Summary

LARS (Local Area Radio System) has been developed for stationary and locally delimited military applications. For such applications the system provides an almost optimum solution for the communication requirements.

However, the mobile units can also interwork regardless of the other units, and they can therefore also be used to advantage for tactical communication. The sturdy construction and high operational reliability makes them suitable for this, perhaps the most demanding of all military applications.

In addition to its ability to withstand adverse environmental conditions and its high operational reliability, LARS has features which enable it to meet the stringent demands made by the military user, and thus put it in a different category from conventional, civil land-mobile systems. Some such features are:

- Channel allocation. All users must have access to all radio channels. Initially each local system is allocated a specific group of channels, and each staff category within the system is allocated a specific channel.
- Traffic mode. The two traffic modes available are single-frequency or two-frequency simplex. The latter mode is used for setting up connections towards relay stations and PBXs. The basic principle is that all users must be able to follow the traffic on the channel they have been allocated. Consequently the traffic is normally open, i.e. squelch controlled. Selective calling is used towards staff at the command centre. They themselves can choose between selective and open monitoring of the radio channel in question. Selective calling is also used for requesting services from the system, for example the setting up of relay circuits.
- Priority. The basic principle is that all members of the staff at the operating centre shall have the same degree of priority. This means, for example, that several operators can connect in to one and the same radio channel simultaneously, in which case the speech is interleaved. This is essential so that all operators are able to send out important messages quickly.

The new base radio system of the Swedish Air Force is a current LARS project. It is intended to meet the communication needs inside air bases. The order was placed in the spring of 1980 and the equipment is now being delivered. The development work was carried out in collaboration with the Swedish Air Force.

Load Study of the AXE 10 Control System

Berth Eklundh and David Rapp

The Department of Telecommunication Systems at the Lund Institute of Technology in Sweden has, with the support of Ericsson and the Swedish Telecommunications Administration, carried out a study of the traffic capacity of the AXE 10 control system with the APZ 210 central processor.
The authors describe the various methods used and the result of the study.

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In the analog version of AXE 10 the switching system consists of link connected units. The proper dimensioning of link connected systems has long been known, and in this respect system AXE 10 presents no new problems. In the digital version the dimensioning is also so ample that only routine checking is necessary.

In the AXE 10 system the equivalent of the registers and markers in crossbar systems is a complex of computers, the system being stored program controlled. The function of the computers is more intricate than that of the markers, and the division between different interworking functional units is complex. The available mathematical methods have not been sufficient. It has been necessary to develop new methods in order to be able to solve the problem. These methods will also be useful for other, similar problems.

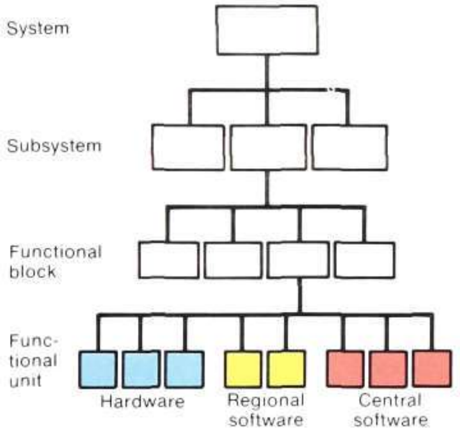

Fig. 1
The functional levels of system AXE 10

- Hardware
- Regional software
- Central software

The structure of AXE 10

The design and function of AXE 10 has been described previously in several articles in Ericsson Review[1,2,3]. A brief summary of the function of the control system is given here as a background to the description of the study that follows.

AXE 10 is built up in a hierarchic structure with systems, subsystems, functional blocks and functional units, fig. 1. The interworking between different functional blocks is by means of signals. When a subfunction is completed in one functional block, the block sends signals to one or more other functional blocks. This initiates other subfunctions in these blocks, which in their turn send out new signals.

Fig. 2 shows a diagram of the structure of system AXE 10. The parts that are of importance to the capacity analysis are the processor system with the central processor, CP, and the regional processors, RP. CP and RP are connected via a bus system with separate buses for groups of RPs. The main work of CP consists of complex, infrequent tasks, such as number analysis, while RP handles routine work, e.g. scanning the line circuits.

Other units of importance in this connection are RPH-I and RPH-O, which

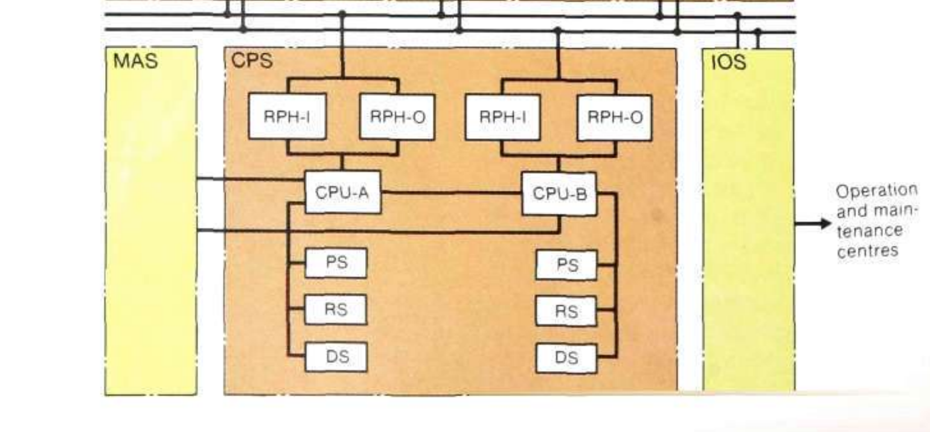

Fig. 2
The structure of system AXE 10

SSS	Subscriber switching subsystem
GSS	Group switching subsystem
TSS	Trunk and signalling subsystem
MAS	Maintenance subsystem
IOS	Input/output subsystem
RPS	Regional processor subsystem
RP	Regional processor
CPS	Central processor subsystem
CPU-A	Central processing unit A
CPU-B	Central processing unit B
PS	Program store
RS	Reference store
DS	Data store
RPH-I	Unit for sensing incoming calls from RP
RPH-O	Unit for distributing outgoing signals to RP

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handle the interworking with the regional processors. RPH-I scans all RP buses cyclically using a polling process, and the signals stored in RP are transmitted to CP. RPH-O transmits signals to different RPs in a corresponding way.

The processors in stored program controlled systems have different priority levels for the jobs they have to perform. Thus certain jobs are allowed to pass others. This method of working complicates the calculations of processor capacity.

In CP the job management is mainly carried out by micro programs. An interrupt system is used which has four program levels and which includes job buffers for queue administration, fig. 3.

The malfunction level, MFL, has top priority. An interrupt signal is generated for MFL when a fault detecting circuit has found a hardware fault which must be cleared quickly.

The normal work of CP is carried out at the traffic handling level, THL, and at the basic level, BAL, with sublevels in accordance with fig. 3. BAL has the lowest priority. Job management at levels THL and BAL uses the job buffers JBA-JBD for storing different signal messages. Such a message contains the receiving block number and the signal number together with the data being transmitted in the signal. An interrupt signal is automatically generated when a signal message is placed in a buffer. Such a signal sent to THL then interrupts work in progress at level BAL. If other work is already being carried out at THL, it must always be completed before the next job can be started.

All signals from RP, and most signals between the CP program blocks are temporarily stored in the job buffers. In addition level THL receives a clock interrupt signal every 10 ms so that signals in accordance with a job table can be sent to program blocks that measure time.

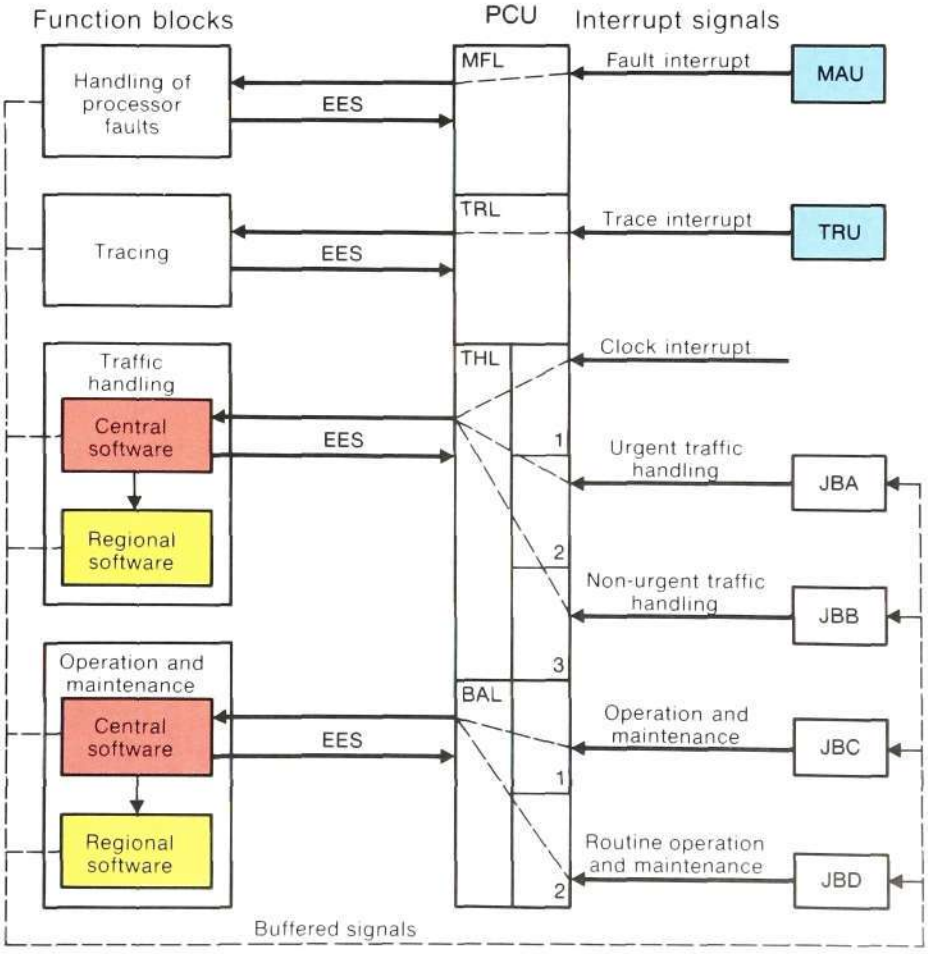


Fig. 3
Job management in CP

PCU	Priority control unit
MFL	Malfunction level
TRL	Tracing level
THL	Traffic handling level with sublevels 1–3
BAL	Basic level with sublevels 1–2
MAU	Maintenance unit
TRU	Tracing unit
JBA	Job buffer A
JBB	Job buffer B
JBC	Job buffer C
JBD	Job buffer D
EES	Execution stopped

Models

In order to carry out a capacity analysis of the system it is necessary to create a model that describes the internal "mechanism" of the system. A model is always a simplification of reality. However, in order for a model to be of any use, the essential functional characteristics must be adequately represented. The practicability of the model as a working tool is of course also dependent on how suitable it is for mathematical treatment or computer simulation.

The following models have been used in the project of analysing the capacity of the AXE 10 processor:
- the Queue Flow Model
- the simulation model
- analytical models
  - queueing network models
  - models of the central processor.

During the work on the models certain assumptions were made regarding the behaviour of the subscribers. On the basis of the results of many measurements the call arrival process has been assumed to be a Poisson process. The operating times of different functional blocks in CP and RP have been determined by means of measurements, or simply by counting the number of instructions in each block.

THE QUEUE FLOW MODEL

A Queue Flow Model shows how the various equipments (with their programs) interwork when carrying out their jobs. This model forms the basis for further work on the project.

Fig. 4 shows a basic Queue Flow Model of the whole AXE 10 processor system. This model takes into consideration CP, RP, RPH-I and RPH-O, all of which are depicted as individual Queue Flow Models in themselves. For example, CP is divided into a server, the processor itself, and several queues, JBA-JBD.

The connections between the different units in the model represent the flow paths, i.e. the paths of the signals in the system. Physically they consist of the bus system. Most signal sequences at the various priority levels use these paths, but to a varying extent. A complete Queue Flow Model would therefore be extremely complicated. More-

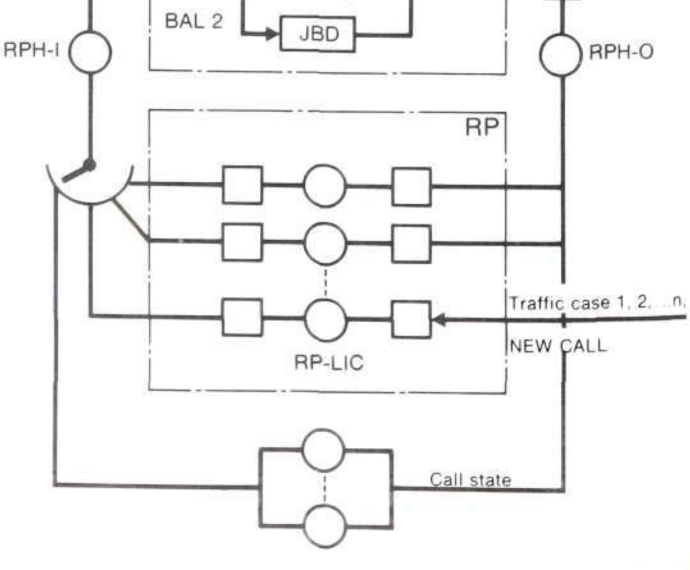


Fig. 4
Basic Queue Flow Model for the AXE 10 processor system

CP	Central processor
JBA-JBD	Job queues
MAL	Malfunction level
THL	Traffic handling levels
BAL	Basic levels
RP	Regional processors
RP-LIC	Regional processor for line circuits
RPH-I	Unit for sensing incoming calls from RP
RPH-O	Unit for distributing output signals to RP

over, the type of traffic handled by an exchange differs depending on its size and application. In order to obtain a manageable model a configuration has therefore been studied for a traffic case where the traffic handling signals mainly use a single priority level. This case is used in fig. 5 to give a somewhat simplified description of the conversion to a Queue Flow Model. The process of events when detecting a call attempt is as follows: A signal for the detected call is created in RP, NEW CALL, and is transmitted to CP. This means that the signal comes in over the RP buses and RPH-I to JBB. If there is a queue, the signal has to wait there, after which a functional block in CP is activated, which in its turn initiates one or more signals, in this case two. One of these immediately activates a new functional block in CP, which in its turn generates a new signal, this time to an RP. The other signal generates a shorter sequence for certain secondary functions, and this sequence is eventually completed. This generation of several output signals leads to activities being performed in parallel, a fact which is of importance for the model.

The setting-up sequence can thus be described as a job, NEW CALL, arriving at RP, where it receives a certain service, and then continuing to CP, where it receives service several times. Sometimes the job receives new service immediately, at other times it has to queue in JBB. Sometimes the job leaves CP directly, and other times it returns to an RP. The basic principle is the same regardless of traffic case or level.

The Queue Flow Model contains no information regarding, for example, the arrival process and service time in different subsystems. Such factors are specified in connection with the further choice of model. In this respect simulation models offer better possibilities than the analytical models.

SIMULATION MODEL

A simulation model can be made "as accurate as one likes". It is possible to model the internal behaviour of a system in the most minute detail, and the simulation program will then behave exactly like the reality it depicts. However, the model development should not be taken so far in practice. There must be a balance between the effort put into the model and the results that can be expected.

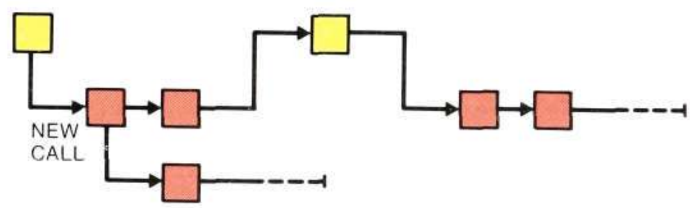


Fig. 5a
Signal flow diagram for level THL 3
RP functional block
CP functional block

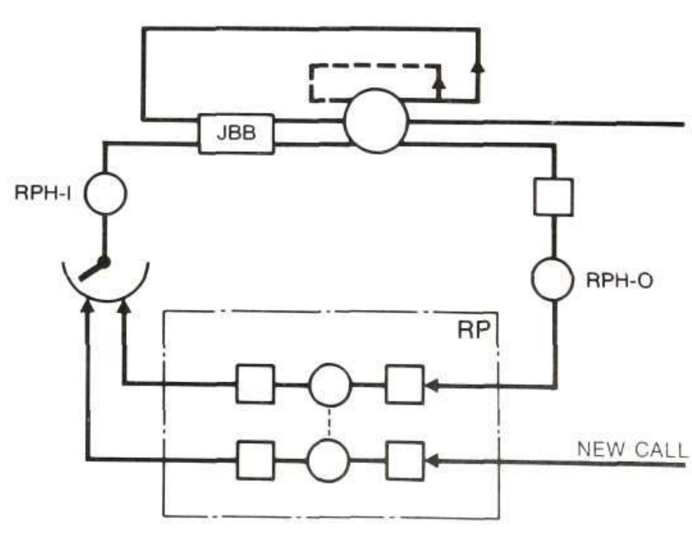


Fig. 5b
Queue Flow Model for level THL 3

Queue theory, queueing network theory

Queues occur because of a temporary shortage of resources. This applies in many every-day occurrences, in the community as well as in technology. In the telephony field queueing problems aroused great interest almost since the first telephone exchanges were built. The development of telecommunication technology has given rise to a multitude of problems, which have often required sophisticated mathematical treatment. The article deals with a difficult queueing problem in modern telephony.

Some basic concepts
The purpose of queueing theory is to provide an answer to the question of how long the waiting time will be under different conditions. The limit for what is acceptable must be decided individually in each case, depending on the overall function of the telecommunication system and the expected reactions of the subscribers.

Configurations. The queueing systems are built up of servers. A simple queueing system consists of a number of servers (one or several) working in parallel and carrying out the same type of service, for example the devices in telephone exchanges that work in delay systems. This type of system is treated with the aid of *queueing theory* (in a restricted sense).

More complicated systems contain different types of servers that must interwork in order to be able to carry out their tasks. Such systems work in several stages, e.g. the processors in system AXE 10. The crossbar systems of types ARM and ARE (transit exchanges) also contain such service systems. Such systems are treated with the aid of *queueing network theory.*

The distribution function for the interarrival times to the queue. Most common (and most realistic) is to assume an exponential distribution. If there is only one server the mean interarrival time must be longer than the mean service time, otherwise the queue will eventually become infinite. *The service time* is either constant or varies with different call categories.

There are a great number of *queue handling methods.* The most common one is FIFO (First In, First Out). The calls can also be fetched from the queue at random. Other alternatives can mean different priority categories for different types of calls.

The result of a theoretical or empirical solution to a queue problem is often given as a series of curves showing the distribution of the waiting time for the calls.

There were several reasons for starting a simulation study of AXE 10. The simulation result was needed for comparisons with analytical results, and the construction of a simulation programme provides further insight into the behaviour of the system. Simulation also has an intrinsic value as a study of methods.

Rather extensive compromises had to be made when preparing the program. The central and regional processors and the communication between them are described in a fairly detailed way, whereas the hardware connected to the regional processors has been left out of the model. The functional blocks for the processing of incoming calls are described in detail for the sequences of events considered to be of the greatest interest.

Jobs that are generated internally within the exchange are usually handled at a lower priority level than jobs which concern the traffic handling. However, certain internal jobs have a degree of priority such that they compete with the traffic handling, and hence these have been included in the model.

The simulation program comprises approximately 8000 SIMULA statements. It was developed using a UNIVAC 100/80, and both the compiler capacity and the available storage space were fully utilized.

ANALYTICAL MODELS

It is not possible to create a mathematical analytical model of the whole AXE 10 processor system containing all the elements described by the Queue Flow Model, the signal operating times in different processors, the behaviour of the subscribers etc. The reason for this is that there are no general analytical methods for networks of queues. It is therefore necessary to create different models, depending on what features are to be studied. For overall analyses the most suitable methods are those in the branch of queueing theory known as queueing network theory. For detailed studies, on the other hand, many useful methods are to be found in the branch of queueing theory that deals with queueing systems consisting of only one stage[5,6].

Queueing network models

A so-called Markovian queueing network model requires simplification of the central processor structure. Thus it is necessary to refrain from giving certain types of signals priority. Moreover, no consideration can be paid to the fact that certain functional blocks generate more than one signal. Finally an approximation has to be made of the distribution of operating time.

The Markovian queueing network model can provide:

1 Information regarding the mean flows in different parts of the system, and the mean load on different parts of the system.
2 Information regarding the length of queues, the waiting time in different parts of the system and sometimes the total time through the whole system.

Much of the structure in the Queue Flow Model for traffic handling, THL, can be retained in a Markovian queueing network model of system AXE 10, but parallel activities cannot be described in the latter model. This is one of the major disadvantages of this type of model.

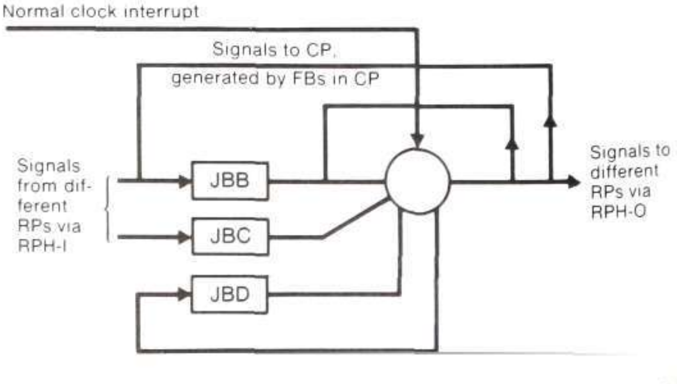


Fig. 6
Model of the central processor (CP)

Models of CP

A comparison between the results obtained from the Markovian queueing network model and the simulation results showed large differences. This meant that certain system parts had to be studied in more detail. It was found suitable to study CP on its own, removed from the system. If the quite reasonable assumption is also made that the call arrival process is a Poisson process, there are a number of methods available. This results in a so-called M/G/1 queueing system in accordance with the usual queueing theory designations[5,6].

This means that previously omitted factors can be included in the model, such as priorities, feedback to CP and parallel signals. In addition a larger number of quantities can be calculated, including the distribution of the total time in CP. The final CP model is shown in fig. 6.

All traffic handling at levels THL 1–3 has higher priority than the maintenance functions at levels BAL 1–2. The priority is of the preemptive-resume category, i.e. no BAL job can be started while a THL job is waiting to be serviced, and a BAL job being serviced is interrupted if a THL job arrives (preemptive), and is resumed when all THL jobs are finished (resume). The waiting times for THL jobs are therefore not affected by the situation at the BAL levels. This means that the THL levels can be studied separately. Further simplification can be made when studying JBB, since the regular interrupts from levels MAL-THL 2 generate jobs of approximately constant length, which normally take up no more than 5–10% of a primary interval.

It was found that good approximate results could be obtained by adding a constant load to the JBB load.

Parallel activities are introduced into the model in the followig way. As soon as any signal has activated a functional block in CP, i.e. when CP is busy, new signals to CP are generated at a constant intensity. This gives an obvious improvement of the model, in spite of the fact that the intensity is low, since a functional block in CP only generates more than one signal in less than 10% of all cases. The resultant model of JBB is shown in fig. 7.

This Queue Flow Model of the CP by itself, together with the previous assumptions regarding the calling process and the distribution of operating times, makes it possible to determine queue lengths and waiting times.

The final queue model is called a feedback queue model, and methods for this type of model have been published[7]. The method can be modified to include the parallel activities discussed above.

However, a final model of, for example, the waiting time to dialling tone means that the whole processor system must be considered, since the signalling sequence up to this stage in the call handling includes processing in different RPs, RPH and CP several times. Because of the internal relations in the system, the delays in different processors will be dependent on each other. This dependence is difficult to include in an analytical model. The waiting time to dialling tone must therefore be described in the model as the sum of a number of independent delays. This is an approximation, the accuracy of which is revealed by a comparison with the simulation results. Classic models for queues with priority can be used for calculating various quantities at levels BAL (JBC and JBD), on condition that the arrival processes are Poisson processes. If feedback and parallel activities are also introduced into the model at these levels, very complicated problems will be obtained, particularly if it is the distributions of the delays of certain signal sequences that are sought.

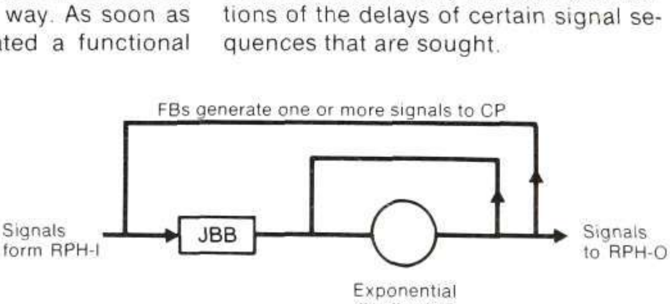


Fig. 7
Working model of JBB

Load ϱ	Number of calls/s	Waiting time (s) is less than for 90 % of the calls	for 99 % of the calls
0.75	47	0.380	0.409
0.90	57	0.437	0.487
0.95	61	0.552	0.652

Table 1
Maximum waiting time to dialling tone for 90 % and 99 % of the calls, at different loads

Results and comparisons

To conclude it can be stated that the available analytical methods for the system as a whole gives a summary picture of its behaviour. Using appropriate assumptions, different parts of the queueing system can be treated separately, for more detailed structural studies. However, it is impossible to form any general conclusions of the effects of such approximations. Comparisons must therefore always be made with measurements on real systems and with simulations.

The numerical results given in this section are based on certain simplifications. Only one traffic case has been considered, and the total load from levels MAL-THL 2 towards JBA has been assumed to be 10 %.

The operating time distribution at different priority levels is an important system parameter. The distribution for jobs at level THL 3 towards JBB is shown in the form of a histogram in fig. 8, but its exact shape varies with the composition of the traffic.

In order to simplify the analysis the operating time distribution was approximated to continuous distributions. The original distribution was replaced by a hyperexponential distribution, well fitted to the original, the blue curve in fig. 8. There was also great similarity to an exponential distribution, and such a distribution was therefore tried, the red curve in fig. 8.

The distribution of the RP delay was assumed to be uniform (t_i, $t_i + \Delta t$). The constant part refers to the action time in the hardware equipment for subscribers and corresponds to, for example, the time required to activate a relay. The other part is given by the clock pulse interval (Δt) in an RP.

The work flows to various parts of the processor system can be determined with the aid of the Queue Flow and queueing network models. The work flows and the mean operating times are then used to calculate the offered loads. The curves on the left of fig. 9 show the measured load on CP from THL 3 jobs via JBB, the curve designated Sim, and its confidence interval (97.5 %). Analytically the load on CP has been calculated in three ways. Curve no. 3 shows the mean value of the total operating time per call, multiplied by the calling rate. Curves nos. 1 and 2 are calculated with the aid of the queue network model. Curve no. 2 shows the load when consideration is paid to parallel activities.

The total time, i.e. the waiting time plus the operating time, through CP for THL3 jobs determines the load properties of the system. Since the operating time is already known, only the waiting time in JBB needs to be investigated. The right-hand side of fig. 9 shows the mean waiting time for an arbitrary signal, which arrives from an RP and which passes once through CP. The blue area marks the confidence interval. Since both the load and the mean waiting time are measured, the uncertainty of the result falls

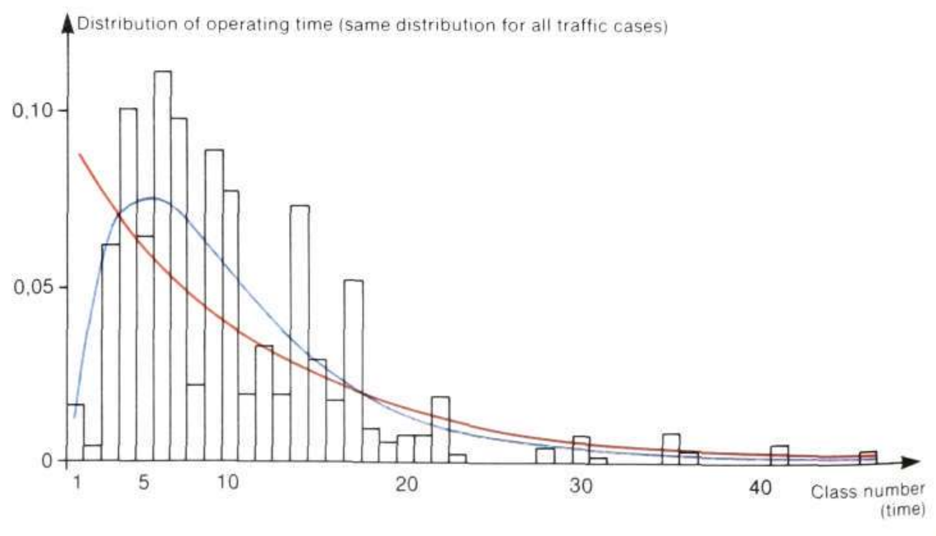


Fig. 8
The distribuition of the operating time in CP for jobs at level THL 3 towards JBB

Measured values, histogram (1)

H_2-E_2, approximation (2)

Approximation with an exponential distibution (3)

Table 2
The table shows the different combinations of operating time distributions for CP and RP, the different models of RPH that have been simulated and how different measured quantities have been affected by alterations. Distributions in accordance with fig. 8

Operating time distribution for CP	Operating time distribution for RP	RPH	Measured quantity	Effect of change
Original (1) H_2-E_2 (2) Exponential (3)	Rectangular Exponential	Original Modified Without RPH	Arrival process to JBB Waiting times, JBB Queue lengths, JBB Busy time, JBB Waiting time to dialling tone	Small Small Small Small Depend. on RP
Exponential (3) with all signals buffered	Exponential	Without RPH	A certain reduction of all quantities	
Without any parallel signals			Large reduction of all quantities	

within a certain area. This is also marked in fig. 9. The results from analytical calculations are also included in fig. 9. The queueing network model was used (curve no. 1, right-hand side) and the results were also adjusted to take the parallel activities into consideration (curve no. 2, right-hand side). The latter curve clearly shows an improvement, but a systematic deviation is still obvious. This behaviour is characteristic of all studied parameters and is probably caused partly by dependency mechanisms which have not yet been fully clarified and partly because the model of the parallel activities is fairly simple. For example, a number of parallel signals to various RPs have been omitted.

Table 1 shows the most important system parameter, waiting time to dialling tone. This time includes not only several passages through CP but also activities in different RPs.

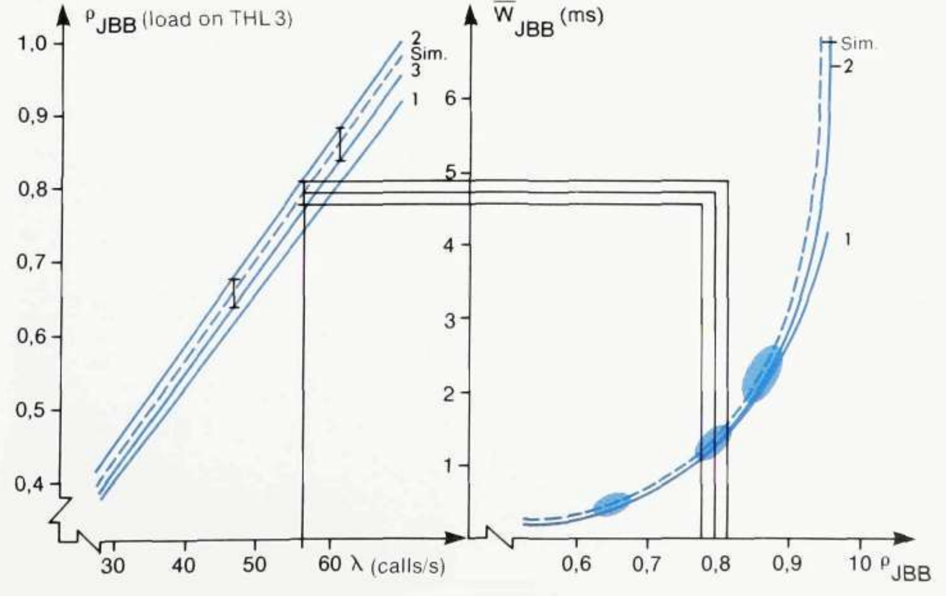
Detailed investigations were carried out with the aid of the simulation program and various analytical calculations, in order to find the causes of the systematic deviations. The exact signal lengths in the simulation were replaced by the distributions in fig. 8. The polling strategy and the cycle time for RPH were also changed, and the RPH function was omitted in one calculation. Finally the uniform distribution in RP was replaced by an exponential distribution. Table 2 shows the possible combinations, the quantities that were studied and the possible effects of the changes. The mean value and variance were compared for all measured quantities.

In the analytical studies related to the model of CP by itself for level THL 3, fig. 7, was studied how the distribution of the operating time affected queue lengths and waiting times in CP. Small differences in the results were found also in this case. The deviations are so small because the operating time distribution for CP differs very little from the exponential distribution. The distribution of the RP operating times was also changed, and the effects of the changes were small.

The most important result of these studies is that priorities and parallel activities have greater effect than differences in the operating time distribution.

Fig. 9
Left: The load on CP from THL 3 jobs via JBB as a function of the number of calls/s
Right: The mean waiting time $\overline{W}_{JBB}$ for an arbitrary signal to JBB

Confidence area
Confidence interval
1 Queueing network model
2 Modified queueing network model
3 $\lambda \times$ mean operating time



Conclusions and summary

Methods

The experience gained from these studies shows that there is no point in trying to build up a very detailed model of the whole system. It appears to be more useful to construct more accurate models of different subsystems, and then to try to combine the results from these part models to give an overall picture of the whole system. This applies particularly in those cases where one part of the system, in this case CP, carries the greatest part of the load. It has also been shown that intuitive ideas about what is essential and not essential are often false, and can easily lead to the wrong line being chosen in the analysis work.

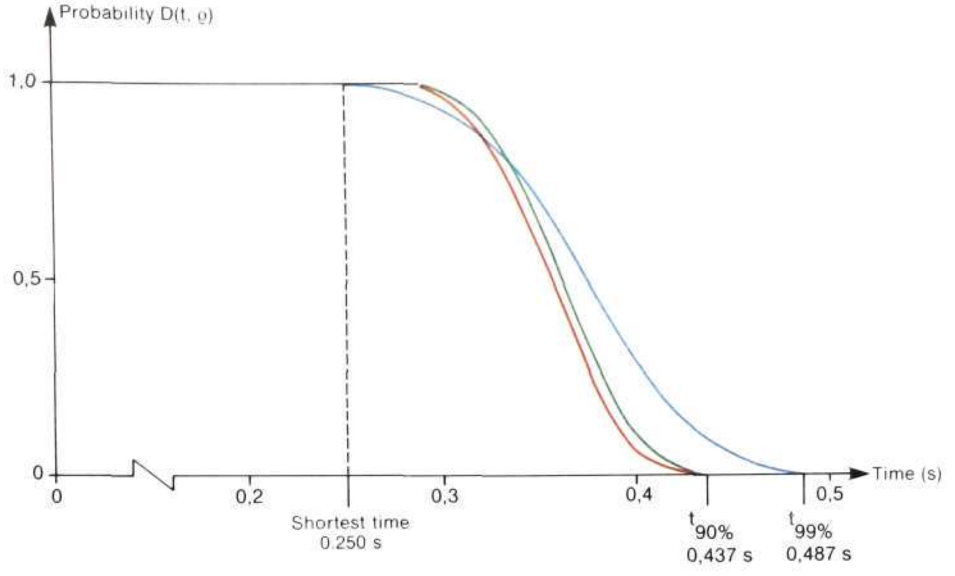


Fig. 10
The distribution of the waiting time to dialling tone (DTD) with a load ρ = 0.9

- Simulations
- Analytical calculations without any correction for parallel activities
- Analytical calculations with correction for parallel activities

Curve	Blue	Red
Mean value for DTD (s)	0.375	0.360
Standard deviation (s)	0.048	0.029

$D(t, \rho) = P(DTD > t, \rho)$
$\rho = LHL + LJBB$
$LHL = 0.1$
$\rho = 0.9$
57 Calls/s

Practical results

Fig. 10 shows an example of the distribution of the waiting time to dialling tone with the load ρ as parameter. A corresponding value for the total number of calls has also been given. Such curves correspond to the practical knowledge that is required concerning the capacity of the control system. Table 1 shows how long the maximum waiting time will be for 90% and 99% of the calls with different loads, based on the results of the simulation.

The table shows that with 47 calls/s the waiting time is less than 0.380 s for 90% and less than 0.409 s for 99% of the calls, i.e. an increase by only 0.029 s. With 61 calls/s the corresponding values are 0.552 s and 0.652 s.

It is a normal requirement that the waiting time to dialling tone must not exceed 1 second for 99% of the calls. The results of these studies show that this requirement is met with good margins even with a very high load on the control system.

The other results of the simulations that have been carried out also confirm the information that has previously been given regarding the traffic capacity of the AXE 10 control system[1].

Comments

The work on improving the models and finding the dependency mechanisms has continued after this article was prepared for publication. Models have been developed which show that the combined feedback (CP–CP, CP–JBB, CP–RP–JBB), together with signal branching, are the predominant structural elements of the system.

The experience gained from the project has also led to a simplification of the simulation model, and it has been possible to make it smaller and faster, while retaining its precision.

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