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ITT Europe Smaller Companies

ITT

Throughout Europe, the 'smaller' ITT companies are making major contributions in the fields of telecommunication and electronics. These companies are not only important industries in their own countries, but have international reputations for high quality, state-of-the-art products. Through their own development programmes and cooperation with ITT laboratories worldwide they are able to build future progress on the latest technologies.

Overview

ITT is widely known as a multinational corporation that includes among its subsidiaries a number of large and successful companies which produce an advanced range of telecommunication, consumer, and related products. In addition these companies undertake extensive advanced research and development into the technologies relevant to current and future telecommunication products. What is less well known is that ITT also incorporates a number of very active companies of more modest size. Many of these companies are located in Europe, and this issue of *Electrical Communication* is devoted to nine of them: ITT Austria (Austria), Standard Electric Kirk (Denmark), Standard Electric Puhelinteollisuus (Finland), Standard Electrik Hellas (Greece), Nederlandsche Standard Electric Mij (Netherlands), Standard Telefon og Kabelfabrik (Norway), Marconi Española (Spain), Standard Radio & Telefon (Sweden), and Standard Telephon und Radio (Switzerland).

These companies, in common with ITT companies throughout the world, are managed and staffed by nationals. While in many ways the companies are very different from one another, reflecting the individual needs of the countries within which they are located, at the same time they share several common features. Management in these subsidiaries enjoys a degree of autonomy unknown in any other major corporation. This freedom of action, while it does not make coordination within ITT an easy task, gives the companies a flexibility seldom found in subsidiaries, and has resulted in increased inventiveness.

A second common feature is that all the companies have made it a key policy to be highly responsive to the needs of their respective countries. Much of their R & D and manufacturing is carried out for their local telecommunication administration or other national agencies. Product engineering is largely tailored to local needs. These units are well established and integrated into their national industrial and economic environments. Nevertheless, it should not be thought they are in any way inward looking. Most have extensive export markets, frequently specializing in particular countries where they can become expert in the market requirements. In addition, these 'smaller' ITT companies make considerable use of the products and technologies of ITT's larger European subsidiaries and their laboratories. They can also consult ITT's microelectronics and software centers, particularly in the United States. Expert groups throughout ITT are able to provide assistance in such specialist areas as digital network planning, programming, VLSI application, and intelligent products where the size of the units precludes them from undertaking major research programmes. Through these contacts within the ITT network, through training programmes, and via rotational assignment in ITT's leading laboratories, staff in smaller units are able to develop experience and expertise in sophisticated technologies.

ITT has been described as one large consulting organization – for its own benefit and that of its clients. The Corporation has long practiced conventional technology transfer – transferring new techniques and know-how to outside companies, mostly in developing countries – as well as technology transfer within ITT, which is highly beneficial to the smaller ITT unit and the country where it is located. The result is a high degree of creativity and a quick response on the part of these smaller companies, as mentioned earlier. A prime example of this technology transfer is the support being given by Bell Telephone

Manufacturing Company in Belgium, to STK who are producing the ITT 1240 Digital Exchange for the Norwegian telecommunication network.

It is worth emphasizing that product and technology transfer is not limited to the direction 'large' to 'small', as is easily demonstrated by a few examples. ITT Austria has produced the operator position for the ITT 1240 Digital Exchange, which is being introduced in many countries throughout the world, and the versatile ITT 5200 business communication system – a new generation digital PABX. In Denmark, SEK has developed and is producing an advanced telephone subset, the Digitel 2000, which has proved an international bestseller and is made under licence in several countries. Still in Scandinavia, STK in Norway has become a world leader in underwater power transmission and has installed nearly 1 100 submarine power cable systems to date.

In Sweden, SRT is a major supplier of sophisticated data modems to a number of ITT units both in Europe and the United States. STR in Switzerland is a leading supplier of telecommunications service systems, including test and maintenance equipment, and has developed the Ovid optical fiber television transmission system which is being sold in Germany and to British Telecom in the United Kingdom.

These few examples illustrate that small size does not preclude success. Indeed, in 1982 an independent jury voted STK 'Company of the Year 1982' in Norway. All units have expanded their R & D activities, and are planning to increase investment to keep pace with changes in the field of telecommunication caused by rapid advances in digital microprocessor technology and other key technologies. This will ensure that all next generation products use state-of-the-art technology for the benefit of the equipment users. As a result the 'smaller' ITT companies in Europe should remain successful, bringing continuing benefits to their national economies and the ITT Corporation.

D. P. Weadock
President
ITT Europe Inc, Brussels

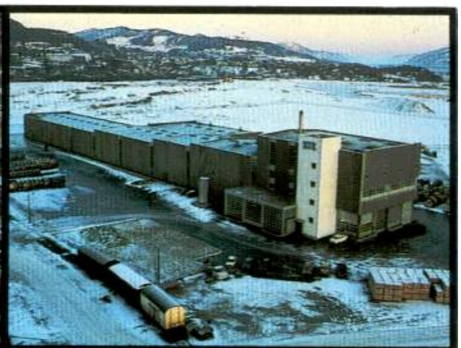
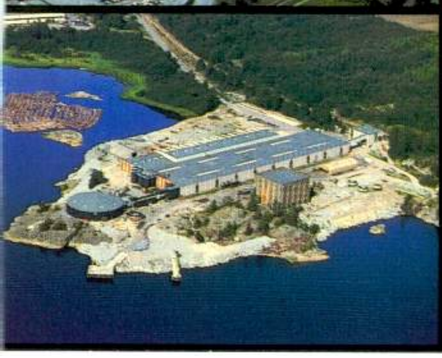
STK Wins Switching Contract for Norway

When the Norwegian Telecommunications Administration decided to modernize the country's telephone system, it looked for a system that would meet present and future needs. The administration foresaw that any new system must be able to incorporate new features and technologies as they become available without affecting the system architecture.

In July 1982, the NTA invited seven telecommunication equipment manufacturers to tender for the supply of digital public telephone exchanges. NTA's enquiry incorporated draft proposals for the supply agreement, including commercial and legal requirements, project specifications for about 200 exchanges totaling 520 000 lines, and detailed specifications relating to technical functions, traffic handling capacity, reliability, and so on. The NTA also specified requirements for the engineering and manufacturing capability of the selected supplier in Norway. Bids were evaluated by a highly qualified team of experts.

NTA's board of directors concluded that the tender from Standard Telefon og Kabelfabrik based on the ITT 1240 Digital Exchange offered the greatest benefits to the NTA and subscribers in Norway. The criteria on which this decision was based included both technology and economics. With regard to economics, the total system cost was considered, including hardware, installation, technical services, and operations and maintenance, as well as the costs of exchange buildings, power, and air conditioning. Other advantages of the ITT 1240 are the ease with which new features, services and technologies can be introduced, and the flexibility of the operations and maintenance facilities.

The first exchanges are scheduled for delivery in mid 1985 when they will undergo thorough acceptance testing by the NTA.


Standard Telefon og Kabelfabrik A/S

Standard Telefon og Kabelfabrik A/S (STK) was established in 1915 and joined ITT in 1930. Cables and telecommunication have been and still are the main product lines. During the almost 70 years since the company was established, STK has become one of the major industrial enterprises in Norway. Development of new, important products has been strengthened through close cooperation with key customers like the Norwegian Telecommunication Administration, the Norwegian Water Resources and Electricity Board, and the Ministry of Defense. Success in the home market has been a good basis on which to build up export orders in the face of strong competition from both domestic and foreign companies.

The Norwegian market and its industrial environment are relatively small, and limited resources restrict what it is economic to develop and produce. However, STK has always been open to technology transfer from outside, and thus managed to establish a fruitful balance between received know-how and own developments. Being an ITT company, STK benefits from the advanced technology and product information available within the ITT System.

As a result of the company's technological achievements and the resulting contracts, STK was voted the title 'Company of the Year 1982' by an independent Norwegian jury. Contributing to the nomination was the good internal relationship between the company's management and its employees.

Cables and Telecommunication

STK's first 15 years of operation were dedicated to manufacturing wires and cables. When STK joined ITT in 1930, the product area was extended to cover telecommunication equipment for a rapidly growing automatic telephone network. Since then, these two main fields have been the platform for all STK activities.

The main products today are power cables, telecommunication cables, telecommunication equipment, and military electronics. More recently data products have been added to this list. Characteristics of all these products are rapid technological evolution and strong competition both at home and abroad.

A key export market in which STK has been very successful is submarine power cables. Successful completion of many submarine power cable projects in Scandinavia, and in particular the cable between Norway and Denmark under the Skagerrak, provided the foundations for STK to undertake the largest power cable project in the world – the power cable between British Columbia mainland and Vancouver Island in Canada. This 525 kV, 1200 MVA cable project ensured STK a leading position in submarine power transmission.

The Skagerrak installation was based on experience gained from more than 1100 submarine cable installations along the coast of Norway.

Another traditional STK key area is main telephone exchanges. The first 7A rotary exchange was installed at Frogner in Oslo in 1921. (This exchange remained in service until 1980, when it was replaced by a METACONTA* 10C exchange.)

The company's outstanding technical staff has experience of the development of main telephone exchanges (8B Crossbar, Metaconta 11B) and transmission equipment (digital equipment, modems, military equipment).

This experience combined with new technology led to the development of a small digital switch (SDS) which forms the

* A trademark of ITT System

basis for complete communication networks, the largest being an entire digital network for the Norwegian Defense Ministry. This network comprises switches, multiplexers, service terminals, and nodes for packet switching. Moreover, the SDS is the key element of the ITT 5500 business communication system – a digital PABX with inherent possibilities for ISDN enhancement.

A third area in which experience and advanced technology have created new products is fiber optic systems. A modern manufacturing plant has been installed at STK's Rognan factory, and continuous efforts are made by the laboratory to create new applications for fiber optic systems. Examples of this are the development of a power cable with an optical fiber core, and the introduction of fiber optics to control petrol pumps.

Research and Development

STK has always given priority to research and development in order to remain competitive. Two different approaches are adopted to ensure this. First is the company's links with ITT and, in particular, the ITTE laboratories.

The second approach is to seek cooperation with key customers at the earliest possible development stage. This provides important market information and makes it possible to undertake full-scale tests of new products at an early stage.

Manufacturing Units

In addition to the main plant in Oslo, there are six specialized manufacturing units spread throughout Norway: Kongsvinger (telecommunication equipment), Halden (submarine cables), Rognan (telecommunication cables), Namsos (power cables), Karmøy (aluminum wires), and Gjøvik (heating foils). These units are key industries in their local environments, and continuous efforts are made to ensure that each of them utilizes the most advanced technologies.

F. Thoresen
Managing Director
Standard Telefon og Kabelfabrik A/S
Oslo, Norway

Submarine Power Cables

In Norway, power is widely distributed using submarine power cables across the many fjords and to the numerous islands around the extensive coastline. Both paper insulated cables and cables with extruded insulation have been used.

K. Bjørløw-Larsen
Standard Telefon og Kabelfabrik A/S,
Oslo, Norway

Introduction

Over the past 40 years, the generation of electric power in Norway has expanded considerably. As the country has a long coastline with a vast number of fjords and islands, there has been a great need for submarine cables to distribute power throughout the country. Standard Telefon og Kabelfabrik (STK) has been the main supplier of submarine power cables in Norway, providing nearly all such cables. By 1979, about 1100 submarine power cables had been manufactured and installed by STK in Norway; several of these cables are lying at considerable depths, the maximum being 670 m – to the author's knowledge, the greatest depth for a power cable. This extensive use of submarine power cables is unique even on a world scale.

Major submarine power cables manufactured by STK in Norway.

Most such cables are of the paper-insulated type ranging in voltage from 1 to 420 kV; the first was laid more than 50 years ago. However, some of the more recent installations have used PVC (polyvinylchloride) and XLPE (crosslinked polyethylene) insulated three-core and single-core cables; XLPE cables are used for voltages up to 145 kV.

Experience

Before 1940, 20 kV AC was the highest system voltage at which submarine power cables had been used. Since then submarine cables have been employed in grids with much higher voltages; by 1979 submarine power cables for up to 420 kV AC had been laid in Norwegian waters. The total length of these cables is more than 2000 km. In addition, two 130 km high voltage DC cables have been laid between Norway and Denmark on the bed of the Skagerrak.

About 85 cable installations laid during the past 30 years use pressurized oil-filled cables with rated voltages from 36 up to 420 kV. STK's first oil-filled cable, supplied in 1948, was a 52 kV three-core cable crossing a lake. The first deep-sea oil-filled cables were laid in 1949 (four single-core 72 kV cables, each 3100 m long, laid at a depth of 180 m).

A number of submarine power cables with extruded insulation have also been supplied. Most of them are PVC insulated, for the voltage range up to 12 kV. STK's first submarine cable with XLPE insulation, a 12 kV three-core cable without a metal sheath, was installed in 1971 and has been in continuous operation since.

Up to the end of 1979, 17 submarine XLPE cable installations for up to 72 kV had been commissioned; the majority were for 12 and 24 kV, but three installations were for 72 kV. The first 145 kV submarine cable with XLPE insulation was installed in 1980. The total route length of STK submarine XLPE cables in Norway exceeds 800 km, most of them being three-core cables. The longest individual length is 14 km with a maximum laying depth of 340 m.

The lengths of submarine cables in Norwegian waters range from a few hundred meters up to 35.5 km (apart from the two Skagerrak cables).

Oil-Filled Cables

For rated voltages up to 145 kV, three-core or single-core cables are used. Above 145 kV all the cables are of the single-core type. In some single-core cable installations, a spare cable has been laid as an insurance in case a fault should occur in service.

Conductor
The conductor of a submarine power cable normally consists of concentric layers of helically wound copper or aluminum wires. This ensures excellent mechanical stability and a smooth surface, which is important for application of the insulation.

Insulation
The insulation in oil-filled cables consists of cellulose paper impregnated with a low viscosity mineral oil. During the past few years a very fluid mineral oil (viscosity of 5 centistokes at 20°C) has been developed which is particularly suitable for long submarine cables.

Metal Sheath
To date the only sheath metals used for preventing the ingress of water into paper-insulation have been lead alloys. Among these, the arsenic lead alloy F-3 (0.15% arsenic, 0.1% tin, 0.1% bismuth, 99.65% lead) has proved the most suitable as a result of its favorable mechanical properties (resistance to vibration, low creep, etc) and it is now used for all submarine cables made in Norway unless the user requests another composition for some specific reason.

Armoring
A submarine cable frequently has to withstand very high tensions during laying and recovery. When in place on the seabed it may be exposed to hazards such as anchors, fishing gear, wear and tear due to waves and tides causing movement, etc.

When single-core cables are used, the armoring carries a relatively large proportion of the induced currents since the conductivity of lead is rather low. Thus the conductivity of the armoring material must be good in order to reduce ohmic losses.

STK single-core oil-filled cable.

Hot-galvanized mild steel wires are the most common armoring material. In situations where mechanical impact is expected to be especially severe, a double layer of steel wires may be used. In a few cases the two layers have been wound helically in opposite directions to avoid twisting during laying. Experience has shown that this gives better mechanical protection than applying the two layers in the same direction.

Outer Corrosion Protection
Normally the outer corrosion protection for a submarine cable consists of layers of bitumen combined with two layers of impregnated polypropylene yarn.

Flexible Joints
Submarine cables are, as a rule, shipped in continuous lengths covering the whole crossing. When factory joints are needed, they must be flexible and preferably should be installed prior to armoring the cable. Flexible joints also simplify repair.

In only three cases have flexible factory joints been necessary for cables in Norwegian waters as a result of long cable lengths. Two of them were three-core 24 kV cables, 26 and 35.5 km long, laid in rather shallow waters. The third case was the two high voltage DC Skagerrak cables which have a total of 20 flexible factory joints laid at depths up to 530 m. All these cables are of the solid paper-insulated type.

However, even though flexible factory joints have not been necessary so far for oil-filled cables in Norwegian waters, such joints have been developed as a precaution in the event of damage occurring during manufacture or while the cable is in service.

Accessories
Generally the accessories for all oil-filled submarine cables do not differ markedly

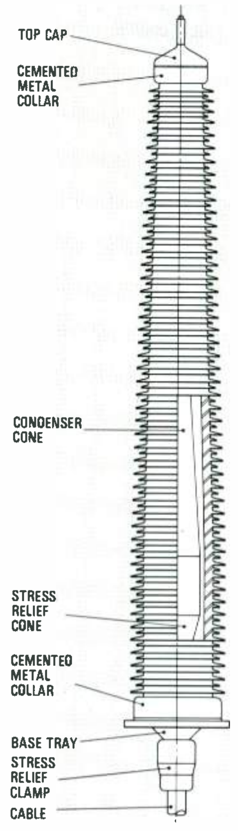


Figure 1
Design of sealing end with condenser cone.

from those for underground cables. However, for long cables and those in deep water, high pressure equipment is necessary to maintain the oil pressure above that of the surrounding water pressure under all operating conditions. For pressures up to 1.5 MPa (about 210 lb inch^{-2}), high pressure feeding tanks can be used for moderate cable lengths. When higher pressures are required, pumping stations become necessary.

Normally the most significant difference between sealing ends for land cables and submarine cables is the length of the porcelain insulator. Since sealing ends for submarine cables are usually exposed to tougher climatic conditions than those installed on land cables, it is often necessary to increase the flashover distance and the tracking length.

An important part of the sealing end is the condenser cone which controls the longitudinal electrical stress. This cone consists of concentric cylinders made of a special type of carbon black paper, separated by paper insulation. A lapped stress relief cone is applied to obtain a smooth transition between the outer shield of the cable and the ground electrode of the condenser cone (see Figure 1). This design has been used in all oil-filled cable installations in Norway for the past 20 years with excellent results. An advantage of this design is that it requires only a very slender porcelain insulator, which again is an advantage when the sealing end has to withstand high pressures.

XLPE Cables

A number of aspects of oil-filled cable design also apply to XLPE insulated cables. However, one outstanding question is whether XLPE insulated submarine cables should be designed with or without an impervious metal sheath.

Water tree growth in XLPE insulation in a wet environment is now reasonably well understood. Present discussion is more about the types of water trees that occur and how they affect the ability of a cable to withstand AC and pulse voltages, and their effect on cable life. A crucial question is the influence of voltage stress on water tree growth and on the acceleration of aging caused by increased stress in wet insulation.

STK has had more than 10 years practical experience with underground XLPE insulated cables, many of which are laid in wet environments such as marshland. During this period there have been virtually no insulation failures. The same is true for 12 and 24 kV submarine cables, which have been in operation since 1971. Because of the high cost of applying a lead sheath to cables, the good service record of XLPE insulated cables in Norway has resulted in many submarine power cables being designed without such a sheath. However, the lower cost has to be evaluated against a possible reduction in the service life of unsheathed cables as a result of water tree growth, and against the vastly more expensive repair of submarine cables compared with underground cables.

Although the service record of XLPE insulated cables for underground and submarine use is very good, the experience in terms of time is relatively short compared to the expected cable life of 30 years or more. Thus STK takes a somewhat cautious approach to the design of XLPE insulated submarine cables for voltages above 24 kV. At present, for these voltages STK generally recommends lead-sheathed cables with the cores manufactured in a completely dry curing process to minimize or eliminate water treeing with the consequent possibility of reducing cable life or reliability. For voltages below 24 kV, however, a lead sheath is generally not used based on service experience and because electrical stress in the insulation at these voltages is low.

STK single- and three-core XLPE cables.

Manufacture

Oil-Filled Cables

The manufacture of submarine power cables requires special production

Turntable at STK's submarine cable plant; it can hold up to 6500 tonnes of cable.

processes. One important factor is the problem of handling long lengths of cable. At STK, large turntables are used both for interprocess storage and for the storage of finished cable. The turntable at the new submarine cable factory south of Oslo can accommodate 6500 tonnes of finished cable.

The lengths of oil-filled cable that could be processed satisfactorily using the orthodox method (i.e. predrying, lead sheathing, and impregnation after sheathing) were rather limited. It was therefore necessary to develop new methods for impregnating long lengths of self-contained oil-filled cable. Norway's first deep-sea cables were dried and impregnated in a tank, lead sheathed while passing through a pipe connected to the lead press, and then filled with oil – the so-called mass-impregnation method. However, during sheathing the surface of the oil in the tank was exposed to air, thereby absorbing oxygen and water vapor.

The insulation properties achieved were sufficient for voltages up to 72 kV, but for higher voltages it became necessary to find a better method, and in 1952 the first oil-filled cable was dried, impregnated, and lead sheathed in a completely closed system. This first mass-impregnation system developed by STK ensured that neither the cable nor the oil enclosing the cable in the vessel were exposed to air or moisture after drying and impregnation of the insulation, thereby greatly improving the electrical and mechanical properties of the insulation. Subsequently, further improvements were made to this manufacturing process (Figure 2).

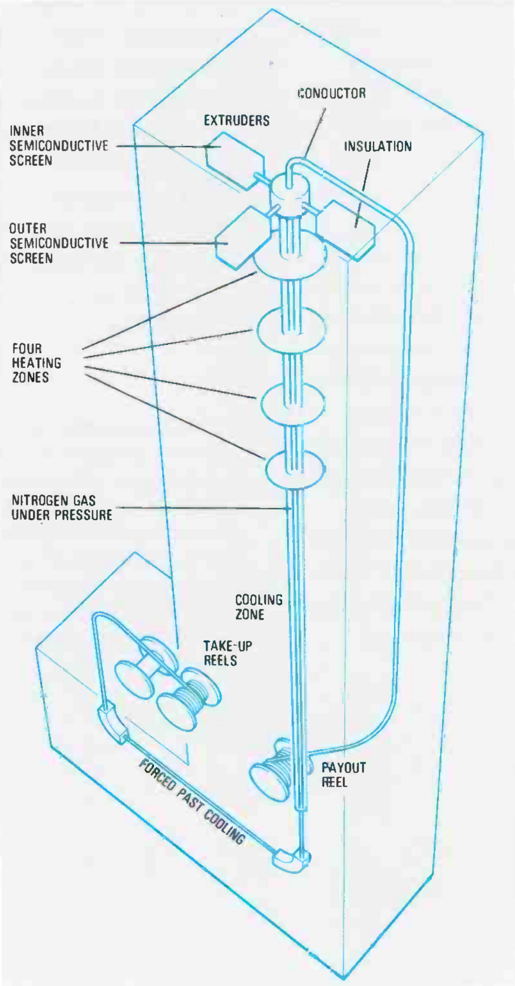


Figure 2
Production of submarine power cable using vertical triple extrusion of XLPE insulation.

XLPE Cables

The insulated conductors for submarine cables with XLPE insulation are manufactured in much the same way as for underground cables. If no metal sheath is required, the armoring is wound directly on the insulation screen, the thickness of which is increased to serve as a bedding for the armor wires. Triple extrusion (i.e. simultaneous extrusion of the conductor screen, insulation, and insulation screen) is normally used to maintain the highest possible standard of cleanliness during the insulation process. A completely dry insulation process has been adopted for cables rated above 24 kV in order to reduce the number and size of voids, and to prevent the ingress of water during manufacture.

Flexible joints between separate core lengths are normally made with XLPE insulation of the same thickness as the core insulation. In other respects the manufacturing processes are much the same as for paper-insulated cables.

Laying and Installation

A good route survey combined with a very accurate navigation system is a prerequisite

Floating a cable end ashore.

for successful submarine cable laying operations. The prelaying survey may have to be performed in two stages to arrive at the optimum route. In addition, a further survey may be necessary after laying.

Transportation
To minimize cable handling, STK prefers to transport submarine power cables in the ship that is to be used for laying.

Regardless of the type of armoring, the best way of storing a cable is on a turntable. For cables equipped with two layers of armoring applied counter-helically, a turntable must be used because such a cable cannot be twisted.

Laying
Cable laying is done in three phases: the first end is brought ashore, then the cable is laid on the seabed, and finally the other end is pulled ashore. It is no longer necessary for cable laying ships to go close in to the shore as suitable methods have been developed for floating the cable end ashore.

The main cable laying is carried out using the cable ship equipped with the same navigation system as was used during the route surveys. The cable laying ship must be capable of maintaining the correct course under the prevailing current and wind conditions, either using its own power or with the aid of tugboats. As both the size and length of submarine cables have increased, and they are laid at greater depths, it has become necessary to develop more sophisticated cable laying equipment. The trend has therefore been towards the use of ships that are specially designed and equipped for cable laying. A specialist cable laying ship, the *C/S Skagerrak*, has been available in Norway since 1975.

Conclusions

The experience with submarine cables laid in Norwegian waters has, in general, been good. Failures have been caused mainly by mechanical damage or corrosion. In installations where cables have been well protected both against mechanical damage and corrosion, or where such stresses are relatively moderate, the failure rate has been virtually nil.

Thorough planning of a submarine cable installation is essential. Planning covers cable design and manufacture, as well as various aspects of installation – route surveying, navigation, laying, and trenching. Methods are available for the thorough evaluation of all relevant aspects of a submarine cable project so that reliable installation is possible, even if conditions are difficult. Many problems associated with the transmission of electrical energy, which were considered insurmountable a few decades ago, can now be solved by installing suitable submarine power cables. In most cases it is merely a question of economics.

K. Bjørløw-Larsen was born in Drammen, Norway, in 1939. He graduated from the Norwegian Institute of Technology in 1965, and joined STK's Power Cable Division in 1967. Since then he has held various positions within the technical department, from cable design and engineering to management of the quality and laboratory departments. In 1979 he was entrusted with the STK management of the 525 kV submarine cable project in Vancouver, British Columbia, Canada.

ITT 5500 Business Communication System

The ITT 5500 BCS is a flexible, modular digital business communication system which uses CEPT standard transmission interfaces between functional units. By configuring modules in different ways, it is possible to build communication systems that meet a wide variety of user needs ranging from digital PABXs to integrated services networks.

E. Sletten
Standard Telefon og Kabelfabrik A/S,
Oslo, Norway

Introduction

The ITT 5500 business communication system is based on digital switching and CEPT (Conference of European Posts and Telecommunication Administrations) standard digital transmission. The system concept was originally developed for special-purpose private network communication; a trial network of seven exchanges commissioned early in 1977 is still in operation. A variety of analog and digital transmission links were used between the exchanges and system modules, including metallic cables, microwave links, and optical fibers. The present ITT 5500 BCS is a development of the original system concept, involving repackaging of hardware, restructured software, an extended range of user facilities, and improved system control facilities.

System modules can be combined to build both traditional centralized PABXs and less conventional arrangements such as distributed PABXs, as well as private communication networks. Non-voice traffic can be handled by the system, thus offering users integrated services communication.

The ITT 5500 system is based on a modular concept which allows it to be configured to meet a wide range of users' needs. Here the DT 80 digital subset is shown connected to an ITT 3290 microcomputer via a V.24/V.28 interface.

System Concept

Modularity is the main feature of the ITT 5500 business communication system concept – a feature which opens up new approaches to business communication.

At the core of the ITT 5500 BCS is a small digital switch (autonomous switch unit or ASU) capable of switching the equivalent of eight CEPT standard 1st order PCM (pulse code modulation) systems, or 256 nonblocking channels. The switch is controlled by a 16-bit microprocessor with optional RAM (random access memory) or PROM (programmable read-only memory). The ASU is divided into three parts (see Figure 1): an interface part, an auxiliary control part, and a central control part. The auxiliary control consists of devices such as generators and special preprocessing units designed to meet the requirements of the particular application. The interface facilitates connection of other system modules to the ASU, such as:

Line terminating group for the local connection of up to 90 users or lines to the ASU; it may occupy 30, 60, or 90 channels of the ASU capacity, depending on the application. The line terminating group to ASU interface, which consists of the BUT (buffer and timeslot interchanger) and LGP (line group preprocessor) units, is specific to the ITT 5500 BCS.

Subscriber (or exchange) line multiplexer for the connection of a remote group of up to 30 users (lines) to the ASU via a CEPT standard digital transmission interface unit.

Other system modules are, in general, connected to the ASU as individual users via either the line terminating group or the multiplexer. Modules connected in this way include:

Operator desktop console, which may be connected via a system dependent digital extension line interface.

System control unit, which is connected via a CCITT X.25 interface. The system control unit is used for operation, maintenance, and administration of the ITT 5500 business communication system.

Data packet switch PS 2000, which is connected to the system via a CCITT X.25 interface.

Physical implementation of the ITT 5500 business communication system is such that the system modules are also physical units (i.e. equipment subracks). These units may be flexibly combined in lightweight cabinets constructed from extruded and cast aluminum parts.

The system programs have been developed according to the principles of ITT's SAFP 2 method, a powerful program development technique that defines a systematic series of development steps. Each step is well documented and test procedures for error checking are specified when relevant. The method ensures the development of well structured programs, and facilitates extensive debugging before programs are integrated with the hardware.

The ITT 5500 BCS programs have a modular architecture; they are coded in CHILL, the CCITT high-level language for telephone switching, using the compiler of the computing division of the University in Trondheim.

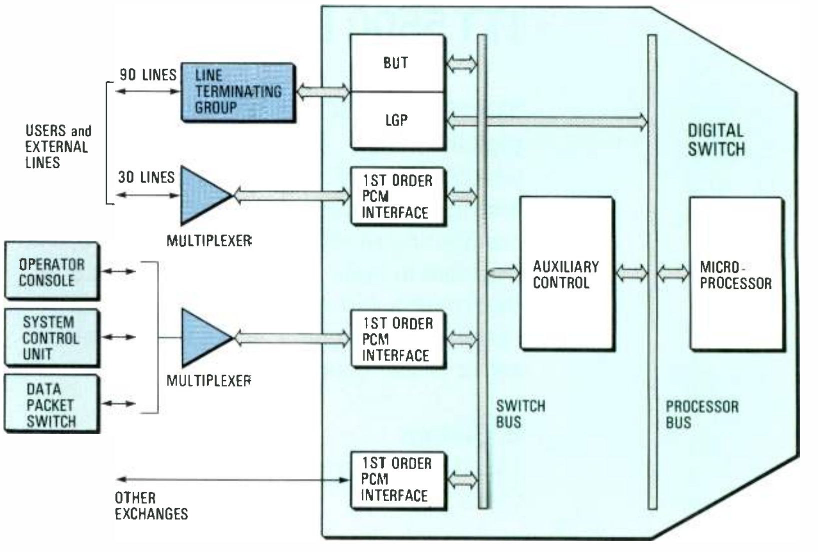
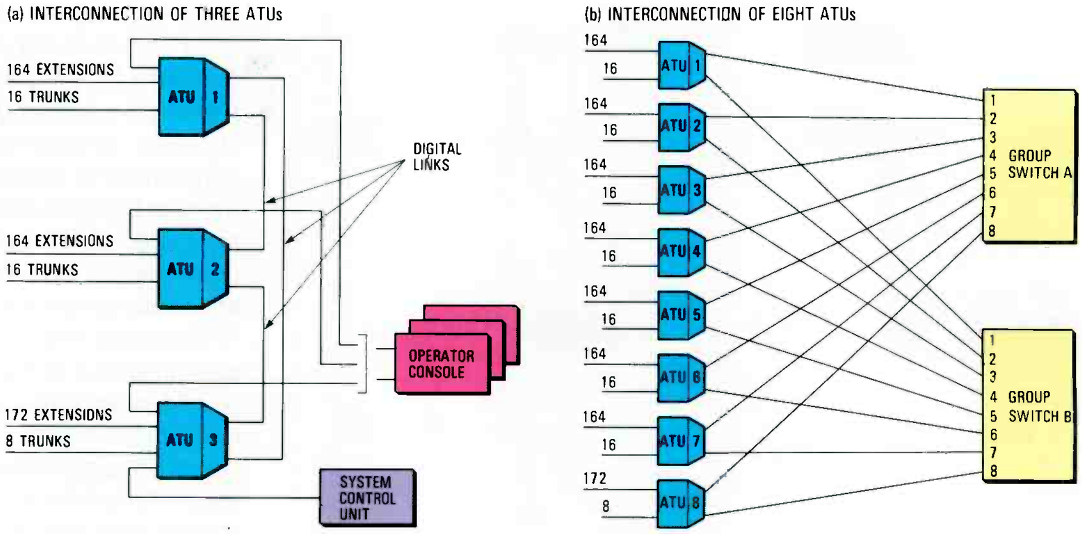


Figure 1
System module overview.
BUT - buffer and timeslot interchanger
LGP - line group preprocessor.

The modular nature of the programs simplifies adaption to meet specific requirements. For example, a specific signaling program package can be built from program modules taken from a library, with the possible addition of modules designed to the user's specification.

ITT 5500 business communication system cabinet with the doors open to show the equipped subracks.

Figure 2
Possible combinations of ITT 5500 modules (a) interconnection of three ATUs (a combination of line terminating groups and a digital switch), and (b) interconnection of eight ATUs.



Examples of Module Combinations

Digital PABXs

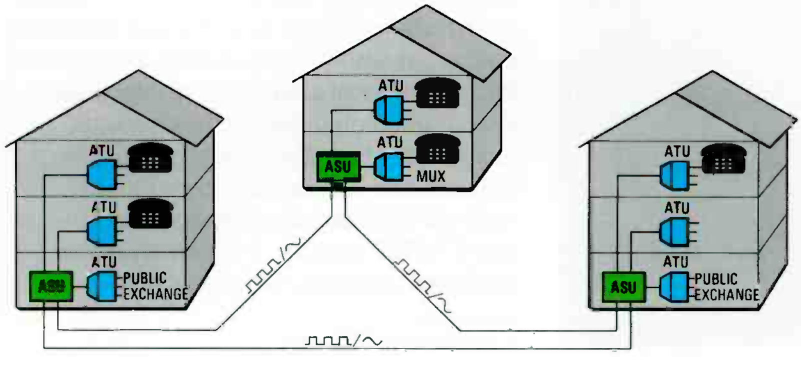
The modularity of the ITT 5500 business communication system opens up various approaches to solving communication problems within businesses and similar organizations. The traditional approach is to provide a centralized PABX. In contrast, Figure 2 shows how ITT 5500 BCS modules can provide such a solution. User terminals (telephone subsets) are connected to the switching equipment via housewiring and a main distribution frame arrangement. Exchange capacities ranging from about 100 to several thousand lines can be constructed. However, as housewiring is becoming an increasingly expensive part of business communication systems, it may prove cost-effective to exploit the modularity of the ITT 5500 to build a distributed communication system. Figure 3 shows a possible module configuration for the distributed equipment approach; this illustrates a distributed PABX with multiplexers and digital switches. Using CEPT standard digital intermodule transmission on either four or eight wires makes it possible to limit housewiring to very short distances, thereby reducing wiring costs. A by-product of the distributed equipment approach is that optical fiber intermodule transmission becomes an economical alternative in environments subject to electromagnetic noise, or where maximum communication security is required.

Private Networks

Geographical separation of various divisions of a company has become more commonplace in recent years. Capacity and facility inadequacies of public telecommunication services have led to the use of efficient, private communication networks over leased lines to provide satisfactory internal communication services between different company locations. The requirement for private network communication can be met efficiently by extending the ITT 5500 BCS distributed equipment approach, as shown in Figure 4. Since CEPT standard PCM transmission is used between system modules, there is virtually no limit to the distance between modules, provided suitable transmission facilities are available. Although digital transmission between switch modules (ASUs) has advantages, analog transmission can also be used.

It is equally easy to build private tandem networks to interconnect existing PABXs. A variety of signaling system program modules is available for interworking between the ITT 5500 BCS and other communication systems.

Figure 3
Application of the ITT 5500 as a distributed PABX with multiplexers and digital switches. For simplicity, the system control unit and operator console are not shown.
ASU - digital switch
MUX - multiplexer.



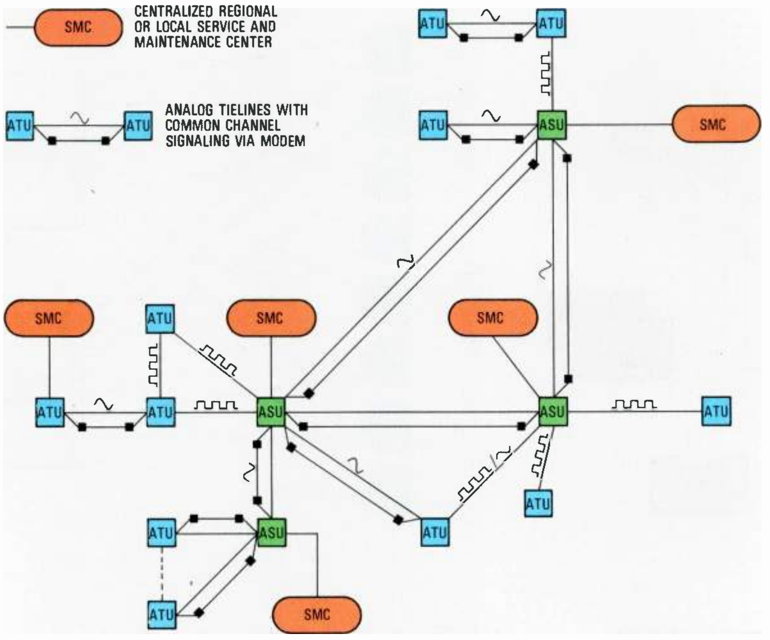
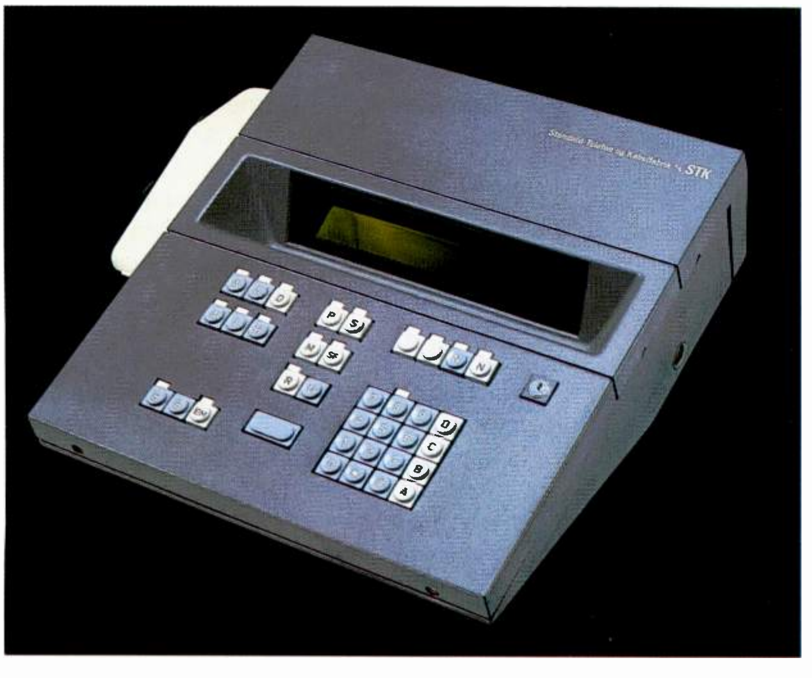


Figure 4
Network example using the ITT 5500 equipment. Where there are multiple SMCs in a network they are generally organized into a hierarchical structure.
∿ - analog transmission
⎍⎍ - digital transmission.

Service Integration

The rapid growth in the use of electronic data processing, and in particular distributed electronic data processing, has given rise to an equally rapid growth in non-voice telecommunication traffic carried by dedicated wire-pair or coaxial cables, preprocessing devices, and even special-purpose data switches. An expanding requirement for the interconnection of such non-voice devices together with a terminal user requirement for access to multiple electronic data processing facilities has led to the development of local area networks and integrated services digital PABXs. The latter offer the user excellent flexibility and economy for bit rates up to 64 kbit s^{-1}, since existing housewiring may be exploited. However, considerably higher bit rates may sometimes be required, such as for file transfer between host data processors or slow-scan video. In such cases a combination of an integrated services PABX and a local area network may be the answer.

Operator console for the ITT 5500 BCS.



The digital nature of the ITT 5500 BCS system suits it to the construction of such private integrated service communication networks. An 80 kbit s^{-1} digital extension line and a system dependent digital telephone subset, such as the DIGITEL* 2000, are the most important instruments for connecting non-voice terminals. Digital extension line interfaces are available both in the line terminating group and the multiplexer. The Digitel 2000 subset, which is equipped with a 2 line by 20 character liquid crystal display, is connected to the ITT 5500 BCS on a 2-wire basis. It may be equipped with programmable keys for special features, and with a standard CCITT V.24/V.28 interface for the connection of V-series data terminals. Thus, simultaneous voice and up to 9600 bit s^{-1} asynchronous or 4800 bit s^{-1} synchronous data traffic is possible on an individual user basis, which is more than adequate for most interactive data processing applications.

Voice and data channels can be directed independently to any desired system address using standard telephone signaling protocols. Alternatively, the data channel can be connected on a per call basis using the system hotline or delayed hotline features. Permanent data channel connection can be established through the system control unit.

Various interface units are available for data transmission rates up to 48, 56, and 64 kbit s^{-1}, and for X-type terminals.

Although the ITT 5500 BCS readily lends itself to flexible line switching of data between user terminals and various data processing hosts, and vice versa, modules such as PS 2000 and others may be added to provide for CCITT X.25 packet switching, packet assembly/disassembly, and protocol emulation; local area network interfaces; and connection to public data network facilities such as the Nordic public data network and Teletex services. Figure 5 outlines the integrated services features of the ITT 5500 system. It shows also how multiplex modules, with appropriate

* A trademark of ITT System

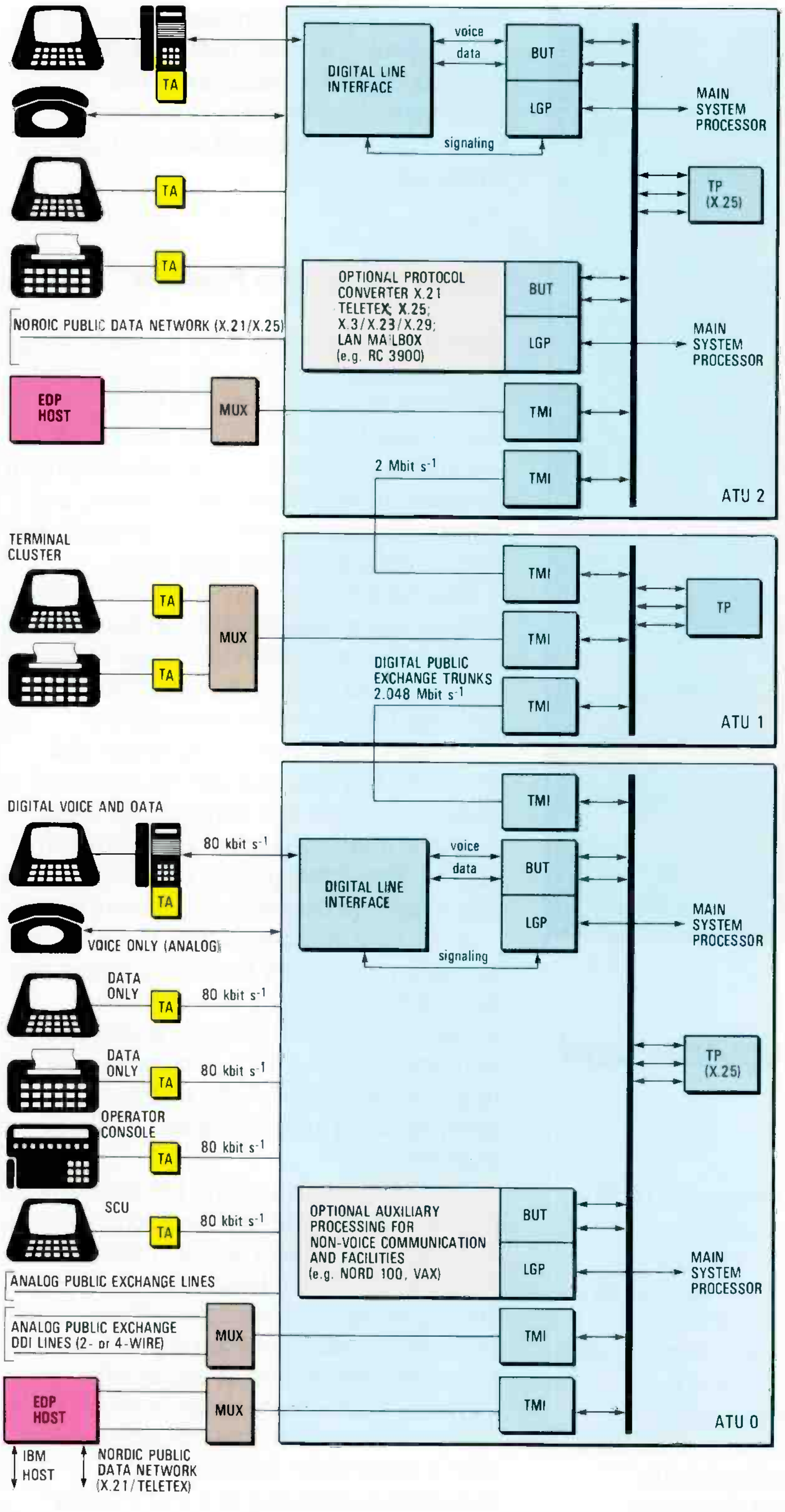


Figure 5
ITT 5500 integrated services distributed equipment used to realize either a PABX or a private network.

EDP - **electronic data processing**
LAN - **local area network**
TA - **terminal access unit**
SCU - **system control unit**
TMI - **1st order PCM interface**
TP - **trunk signaling processor.**

interfaces, colocated with a data processor provide an efficient means of connecting terminal users in the network to data processing facilities. A further possible development is the introduction of electronic data processors that interface directly with CEPT standard 1st order PCM lines, making it unnecessary to colocate a multiplexer with the host machine.

Conclusions

The modularity of the ITT 5500 business communication system provides unparalleled flexibility. Traditional, centralized digital PABXs ranging from 100 lines to several thousand lines, distributed PABXs, and private networks can be easily built using suitable combinations of system modules. The system also offers considerable potential for cost effective housewiring solutions in the case of service integration. The traditional multiwire DTE (data terminal equipment)/DCE (data circuit terminating equipment) may be limited to only a few meters by colocating a multiplexer with the DTE or the host data processor, or by using the digital extension line with Digitel 2000 subsets.

Finally, since the system modules are also physical equipment units with internationally standardized interfaces for connection to other modules, it is easy to redesign modules to take advantage of new technology; also new modules can be developed without affecting the system architecture.

Erik Sletten was born in Oslo, Norway, in 1936. He studied electrical engineering and automation at Twickenham College of Technology, London, graduating in 1964. He joined STK the same year, and was involved in marketing PABX systems, special private networks, and document handling equipment. From 1969 to 1980 Mr Sletten worked on the development and introduction of the Scandinavian range of Minimat PABXs in Norway. Mr Sletten is now technical manager of the PABX engineering and development department of STK.

Standard Telephon und Radio AG

Standard Telephon und Radio AG (STR) was founded in 1935 as a subsidiary of ITT's Belgian associate, Bell Telephone Manufacturing Company (BTM). During the second world war, STR became independent of BTM and was constituted as a wholly owned subsidiary of ITT.

From the outset, STR has been involved in the fields of telecommunication and components, becoming a main manufacturing supplier of automatic switching equipment and transmission systems to the Swiss PTT. Traditionally, STR has also covered the components business, reselling primarily ITT components in Switzerland, and producing STR designed electromechanical relays for the world market.

Currently STR employs some 2200 people, generating an annual revenue of around $ 130 million. The company's head offices with the administration, engineering departments, and laboratories, and the marketing and sales functions are located in Zurich, while the main manufacturing facilities are in Au-Wadenswil. The installation department is concentrated around the three regional offices in Zurich, Basel, and Geneva.

Telecommunication Products

Over the past decade, STR's product range has increased considerably in response to new market requirements. In the field of public switching, exchanges for special applications such as mobile radiotelephone systems, railway networks, videotex, and operator assistance have been added to the traditional local and toll exchanges.

Special attention has been given to the area of telecommunication service systems (TSS) with the creation of a range of totally new products. This new product family includes both local and universal call simulators, a network quality tester, call diverters, and a call handler for advanced telephone subscriber services. All these products make extensive use of program control. They have proved of interest to many network operators around the world.

In the field of transmission, the product spectrum covers low frequency and carrier frequency equipment, coaxial cable systems, and digital terminal and circuit equipment, as well as a comprehensive range of analog and digital microwave radio systems and associated space diversity modules.

The latest addition to the transmission product range is optical fiber technology. STR has developed a family of devices for the transmission and reception of analog and digital signals over optical fibers. OVID can carry a color television signal and up to four audio channels of studio quality over distances up to 10 km. In contrast, OTEL is designed to carry 600 telephone channels over a single fiber. Digital optical transmitters operating at 2 and 8 Mbit s^{-1} have recently been complemented by long wavelength versions that exploit the advantages of operation at 1300 nm.

Broadcasting Studios

Within ITT, STR specializes in the field of studio engineering for both radio and television broadcasting. Outside broadcast vehicles, on-air transmission control equipment, and master control equipment based on STR's remotely controlled audio distributor are manufactured and supplied by the company, frequently as part of a turnkey contract.

Finally, within Switzerland STR markets a wide range of products and systems manufactured by ITT units in other countries. In addition to the electrical and electronic components already mentioned, these products are oriented towards national defense requirements (radar, HF radios, simulators), civil aviation (communication and control equipment, navigation aids), and end-user equipment (intercom systems, data terminals, data and telex communication controllers, videotex).

The Future

A judicious blend of home manufacturing facilities and development supported by ITT's international research facilities, and resale of sophisticated equipment from other ITT houses will ensure STR's success over the coming decade.

W. Thierstein
Managing Director
Standard Telephon und Radio AG
Zurich, Switzerland

Network Quality Tester

The quality of a telephone service as seen by the subscriber is an important characteristic which should be checked regularly. The network quality tester has been designed to provide rapid and accurate assessments of network quality using microprocessor controlled equipment in exchanges.

R. Dietschi
Ch. Gessler
E. Staber
Standard Telephon und Radio AG, Zurich, Switzerland

Introduction

The quality of the telephone service as seen by the subscriber is adversely affected by various types of fault, including the loss rate (i.e. the number of calls that are not completed because of faulty switching equipment or a lack of trunk circuits). In a well managed telephone network, the loss rate should not exceed 1% and equipment failures should only account for a fraction of this value.

Quality-of-service measurements should not only be made when an exchange is cut over, but throughout the operating life of the exchange within the framework of the entire network. This is precisely the purpose of the NQT (network quality tester) developed by STR (Standard Telephon und Radio), which enables the quality of service of a telephone network to be measured at any time.

Tests are performed by initiating calls at predetermined intervals on fixed routes. These calls are computer controlled to ensure that they do not interfere with subscriber traffic. Mutual interference between test calls is prevented by careful coordination.

The NQT test programs, which are easily customized for use in both stored program control and electromechanical exchanges, supply information about the network on which the tests are to be run. Test reports can be formatted and printed out according to various criteria. A central processing unit stores data on all calls, whether completed or not. Depending on the selected output, the NQT provides information on types of fault, measured parameters, and statistical data. Test reports can help locate and eliminate faults.

The modular design of the NQT makes it possible to configure it optimally for a wide range of requirements, from the smallest to the largest local exchanges. This flexibility has been achieved by using microprocessors to control the central and test units, as well as the dialing equipment.

Testing Concept

Figure 1 shows how the NQT is used to test a network group in a telephone network.

The NQT consists primarily of test units built into exchanges, connection to the exchange being tested via standard subscriber lines. Test calls are established between two such test units or between a test unit and a simple answering set in the case of small local exchanges. One or two answering sets can be connected to each exchange test unit. Test programs are generated by a central unit which controls up to 150 test units. Results are transmitted back to the central unit at predetermined intervals over the same paths.

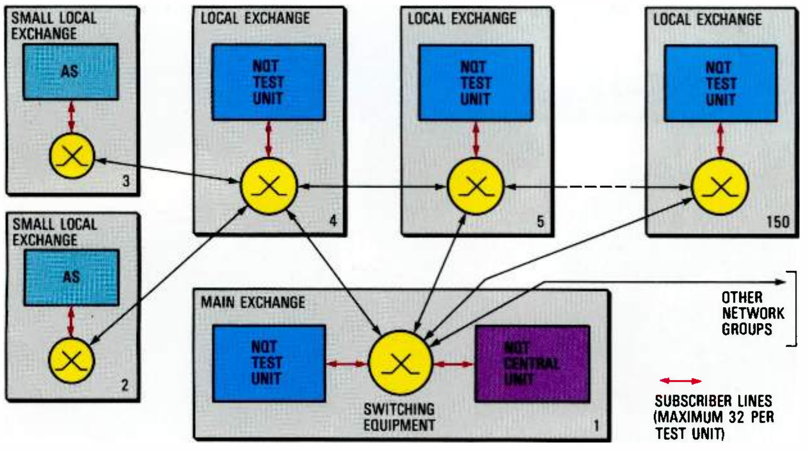


Figure 1
Basic concept of NQT system, showing the configuration of exchange-based test units and answering sets which are controlled by a central unit.

The NQT repeatedly establishes test calls from subscriber line to subscriber line using all trunk groups. To test the quality of service fully, several test calls must be established from each local exchange to all other exchanges. However, as one complete test cycle in a network with 150 test units would require 22350 test calls, the number of participating test units is limited to 75; this still requires 5550 test calls.

The complete network is therefore divided into subgroups, for example, corresponding to a toll exchange section. A test cycle covering all traffic directions within a subgroup can thus be completed within a reasonable time.

Execution of Test Programs

Test programs are initiated by the test units to ensure that a large number of calls can be established simultaneously.

Under the control of the central unit, the test units set up test calls to each other and analyze the results. Each test unit is equipped with calling circuits and answering circuits, enabling it to handle incoming and outgoing calls simultaneously. Additional test line selectors can be installed, providing each test unit with up to 32 connections to the local exchange.

The answering set used in small local exchanges is a simplified version of the test unit; it consists of a single calling circuit and an automatic answering set, as well as a simple operator panel. As it is not equipped with a modem, it can only set up a preprogrammed acknowledgment call when a call is received. An 800 Hz test tone is used to check the speech wire between two exchanges during each call. This tone can also be used to transmit simple messages.

Under no circumstances must the NQT interfere with subscriber traffic, so test calls are limited to a small proportion of the total traffic. The NQT immediately bars any subscriber number if the corresponding test call is not answered by an NQT device; for example, if a number corresponding to an actual subscriber is dialed in error.

Assembling the Test Programs

The test programs required by the test units in various types of exchange are assembled in the central unit with the aid of a minicomputer which is initially loaded with data describing the network configuration. The programs are then transmitted to the corresponding test units via 1200 baud half-duplex modems.

However, the test steps and network data cannot be assembled arbitrarily into a test program. Meaningful results can only be obtained if the traffic distribution in the network and the probability laws of quality assurance are taken into consideration.

Operator panel of the exchange-based test unit.

Prerequisites for Meaningful Test Results

To obtain a representative picture of the quality of service in a telephone network, the test procedures take the following network parameters into consideration:

- size and structure (up to five subgroups are possible)
- number of directions for each exchange
- number of lines per bundle
- number of test units and simple answering sets involved
- number of test circuits per test unit
- network attenuation plan
- permissible error rate per direction
- testing mode (comparison or measurement of actual values)
- test parameters
- test schedule.

Two requirements must be satisfied to obtain realistic test results: there must be no interference with subscriber traffic, and no mutual interference between test calls.

To ensure that subscriber traffic is not affected, test traffic generated on each bundle must not exceed 0.5% of the total traffic. The maximum traffic generated by a test unit is approximately 0.67 erlang. However, at least one call per cycle must be established on each bundle. If an exchange serves several bundles, this test traffic is distributed across all bundles. Depending on the testing mode, calls can either be distributed uniformly or in proportion to the bundle size. Proportional distribution allocates test calls so that each bundle is loaded with the same percentage of its capacity – a prerequisite for comparative evaluation of the test results.

Test Methods

Timeslot Method for Call Distribution
To prevent mutual interference (e.g. caused by busy answering sets), test calls are scheduled on a timeslot basis. The call program in the selected test units is started simultaneously at fixed times. The test program is constructed to preclude coincident test calls, so that an answering set is available for each call. The width of the timeslots depends on the testing mode. All simultaneous calls must be completed within the timeslot, otherwise mutual release of active test connections is forced and a fault recorded.

A test cycle consists of a number of timeslots. In addition to preventing simultaneous seizure of answering sets, the test program must schedule the required number of test calls for each bundle in the network group and allocate these test calls as evenly as possible to the available timeslots.

Proportional Distribution Across Entire Test Network
In this mode, test calls are distributed in proportion to the bundle traffic in the entire test network, making it possible to compare the quality of service in different telephone networks. A proportionality factor specifies how often or in which timeslots a test call must be established on the corresponding bundle. Test calls are distributed across up to 150 timeslots, which make up one test cycle.

Proportional Distribution Across all Bundles of an Exchange
This method is analogous to the previous mode, except that test calls are only distributed over the bundles in a given exchange.

Uniform Distribution of Test Calls
All bundles of an exchange are loaded with the same number of test calls; all exchanges are called one after the other.

Exchange Test
In this test mode, only the exchange connected directly to the test unit is tested. Calls on outgoing test lines are looped back to corresponding answering circuits. This routine can also be initiated as part of the self-testing facilities.

Loss Rate Dependent Test Sequence
The test sequence is influenced by the permissible fault or loss rate specified for each direction. There are two basic possibilities. In the first case, a fault report is only issued for directions in which the admissible fault rate has been exceeded. The test program is subsequently canceled. In the second case, the decision on whether to continue the test program is influenced by the specified loss rate. There are three criteria. If the loss rate is either below the limit or above it, the test program is canceled. However, if the loss rate is within an uncertainty range, the test is continued until a clear result has been obtained or the maximum number of test calls has been completed.

Distribution within a Bundle
A further uncertainty in the interpretation of results occurs if the test load is distributed uniformly across all lines within a bundle. Testing with a relatively small proportion of the traffic load (less than 0.5%) provides only a sample. A method must be found by which each line of a bundle can be reached by the NQT. Two methods are feasible: specific selection of lines, or the setting up of a sufficiently large number of test calls to

Simplified answering set for use in small local exchanges where a full test unit is not required.

ensure a high probability that every line will be reached.

A fully available bundle is a prerequisite in either case. The NQT system uses the second approach since the specific selection of lines is considerably more difficult. Also, it is reasonable to assume that faults noticed by a subscriber will sooner or later occur during a test call.

As faults on frequently seized lines occur with a correspondingly higher frequency, such lines are also seized more frequently by test calls. The probability of detecting a fault is, therefore, the same for the NQT as for any other subscriber. However, the time taken to detect a fault can be reduced if a large number of test calls are established over a given period of time.

Central Unit

The central unit of the NQT has four main tasks: man-machine communication; preparation of test programs for the network to be tested; initiation and monitoring of tests; and preparation and output of test reports.

Figure 2 is a block schematic of the central unit which essentially consists of a minicomputer with 512 kbytes of main memory and up to six dialing units. The dialing units, which are controlled by the minicomputer via the RS 232C interface, are fully equipped to establish a modem connection over dial-up lines.

Performance Characteristics

A test network monitored by a central unit can include up to 150 exchanges, each of which can be equipped with a test unit or the simpler automatic answering set. Tandem exchanges without NQT equipment can also be included in a test network. Up to 32 test lines of a local exchange can be connected to a test unit via the test line selector.

Up to 75 test calls can be initiated within a given timeslot (e.g. 75 test units equipped with one subscriber interface or 37 units equipped with two subscriber interfaces can be selected). Any other combination up to the maximum is also possible.

The maximum configuration with 150 exchanges can be divided into any number of subgroups, but no more than five of them can participate in the tests at one time.

Central Unit Operation

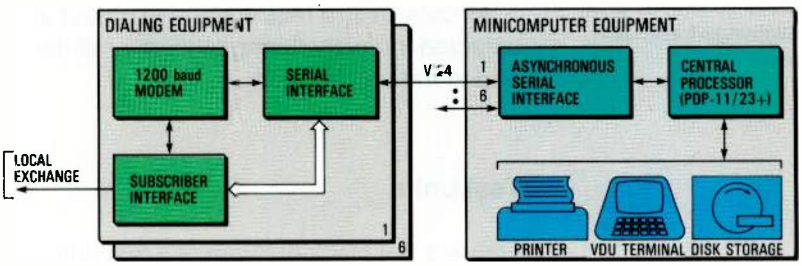
With a few exceptions, the entire NQT system is controlled by the central unit. Man-machine communication takes place via the video terminal. The CCITT man-machine language has been followed as far as possible.

The central unit can perform the following tasks:

- prepare test programs
- prepare and print out test reports
- test program downloading to test units
- accept and temporarily store test results from the test units
- monitor the test sequence
- define the network configuration.

Some of these functions are executed automatically, while others must be initiated manually.

Figure 2
Block diagram of the central unit which is controlled by a minicomputer.



Data Entry

Data is entered using a dialog mode (i.e. the system prompts the user for the required information). Changes to data on the screen can be made by overwriting the corresponding fields. The end of an input sequence is terminated by a special command that generally causes the entered commands to be executed.

All data such as test programs, network configuration, and test parameters can be displayed on the screen or output on the printer.

Output of Test Results

Test results can be output in four basic formats: detailed fault reports, summaries, measuring logs, or statistics.

Detailed fault reports identify all failures with fault type, calling and called exchange, and subscriber number; time and date; and summary, including fault rate.

Measuring logs include all measured values, such as dial tone delay and level, ringing tone delay and level, and speech path attenuation.

Statistics can be prepared and output for all types of fault and measured parameters. They summarize the test results and convey a global picture of the situation.

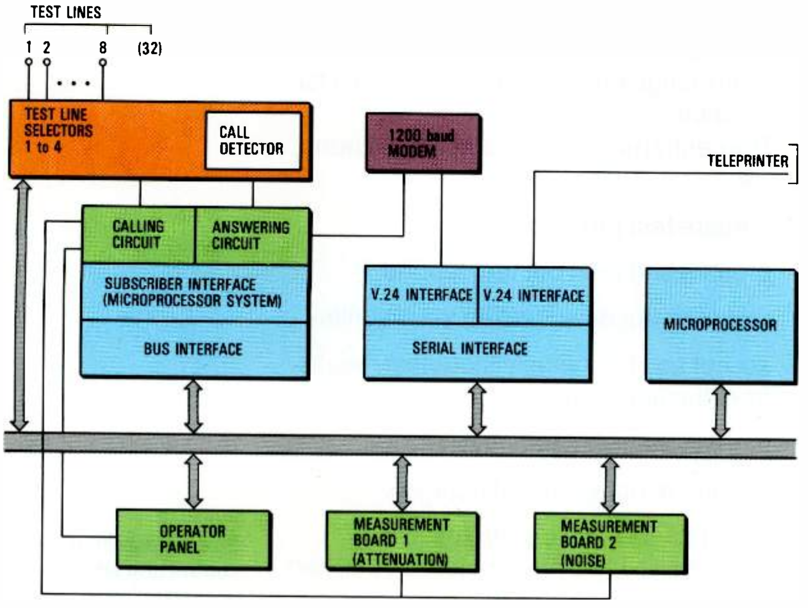


Figure 3
Block schematic of an exchange-based test unit.

All data can be requested and output at selected intervals during testing, or at the end of the test program.

Test Units

Figure 3 is a block diagram of a test unit. Individual assemblies are connected to the processor by a bus system.

Testing Facilities

The test units which establish the test calls are controlled by the central unit over dial-up lines. The following data is supplied to help set up test calls:

- accurate synchronization of timeslots
- time at which a test call is scheduled
- number of the subscriber termination in the local exchange that is to be seized for the test call
- subscriber number to be dialed
- testing mode
- test parameters
- start and stop times of test program.

Based on this information, the test unit establishes or receives test calls autonomously. Its dialing store capacity is 150 destination numbers, each with 16 digits. Test results are stored temporarily before being transmitted to the central unit.

In addition to setting up test calls, during other timeslots a test unit can act as an answering set to other test units and for the one or two simplified answering sets under its control.

Depending on the equipment and the desired testing mode, a program consists of various tests and measuring routines, as illustrated in Figure 4.

Comparison of Nominal/Actual Values

In this mode a simple pass/fail decision is made based on a predetermined nominal value; any failures are recorded. Measuring parameters can be entered to define the decision criteria for the following tests:

- maximum dial tone delay
- admissible tolerance for dial tone level
- maximum ringing tone delay
- admissible tolerance for ringing tone level
- maximum speech path attenuation
- maximum weighted and unweighted noise levels
- maximum quantization noise level
- time limits for arrival of charge pulses.

Measurements

If the NQT is programmed for detailed measurements, the above values are measured and stored. Results can then be printed out as a measurement report or used to compile statistics.

Operating the Test Units

Each test unit can be programmed and operated via the operator panel or a VDU terminal. This manual mode is normally used for troubleshooting. It is possible either to monitor the programs downloaded by the central unit or program test calls directly on the unit itself. The following tasks can be performed:

- establish calls
- stop or disconnect calls
- hold a connection on detection of a fault
- hold connection in the speech mode and transmit test tones
- set up a call step-by-step with indication of the present connection phase
- acoustically monitor a test call
- input all test parameters
- display all parameters.

Simplified Answering Set

This answering set can be used when the full range of tests and measurements is not required (e.g. in small local exchanges). This unit consists of only a subscriber interface circuit (which incorporates the microprocessor and answering circuit) and an operator panel with the most important controls. It can answer test calls and call a preprogrammed number.

Each answering set is controlled by a higher ranking test unit which calls the answering set and waits for an acknowledgment. A fault is recorded if no acknowledgment is received. Simple messages can be transmitted by keying the tone signal used for checking the speech path.

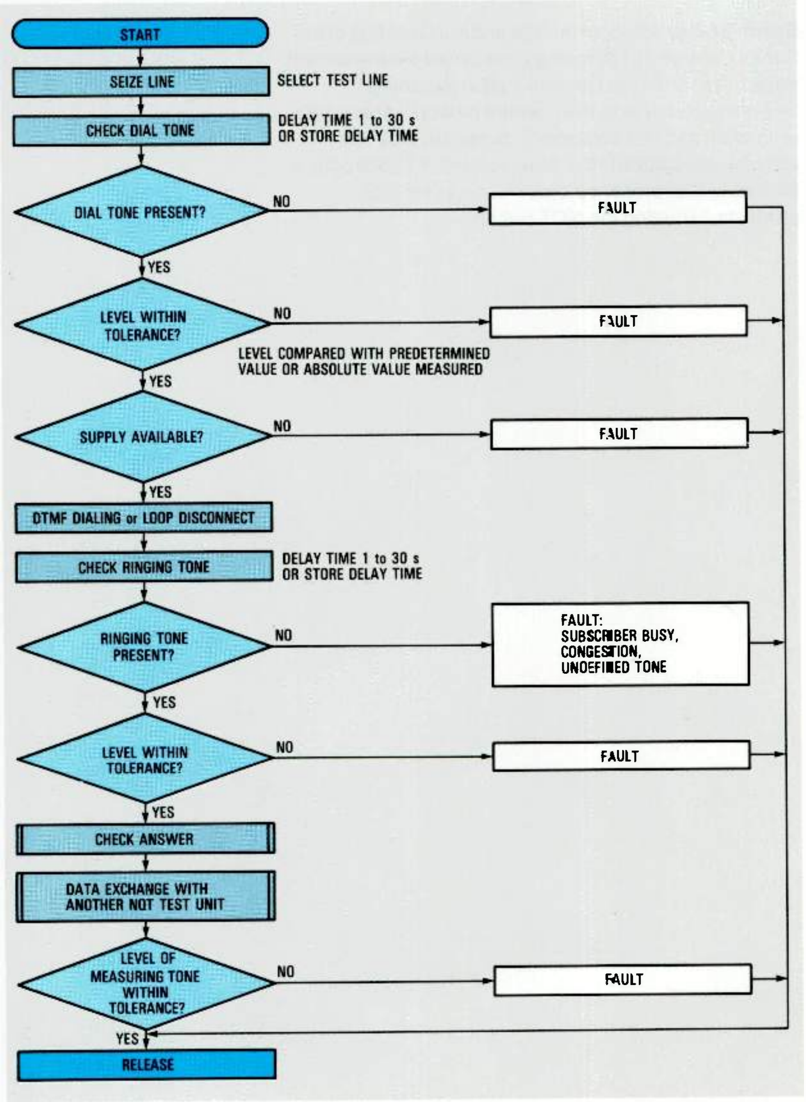


Figure 4
Flowchart for the execution of a test call generated by a test unit.

NQT Programs

Because NQT is designed to divide tasks between the central unit and the exchange-based test units, the programs are also organized by equipment type. The common interface of the central unit and test unit program blocks is the serial data channel and its modems. The simple answering set functions autonomously because it is not equipped with a modem; only the network signaling system is available to connect it with other system units.

Individual processors have a number of common functions (e.g. call setup). To take advantage of this commonality, the same operating system is used wherever these functions are required. In the central unit, this concept applies to the dialing unit which is used as a selection processor.

The operating system MIRTOS (minimum real-time operating system) was developed by ITT in the PL/M high-level language. It is based on strictly synchronous process changeover (without process displacement) and fixed priorities. This operating system performs exceptionally well because of its simplicity and transparency in microprocessor applications with fixed task assignments.

Central Unit Programs

The central unit minicomputer, which uses the RSX-11M operating system, is responsible for all functions that must be executed centrally in conjunction with remote control of the test units. Programs are written in Fortran; assembler is only used where fast execution is critical.

The main functions provided by the central unit programs are:

- man-machine communication with the operator
- network configuration maintenance
- test program preparation
- supervision of tests
- evaluation of tests.

Dialing Unit Programs

The communication protocol between the central unit and a connected test unit is handled by the dialing equipment and is thus not transparent to the minicomputer. Communication between the minicomputer and the dialing unit is program controlled.

Test Unit Programs

The main functions of the test unit programs are call processing, operation of modem and data transmission, call program execution, measurements, input/output with local operator, and results preparation.

Conflict could occur if an operator intervenes in a running test program; this is resolved by a call to the central unit which then removes the affected unit from the test in progress.

Conclusions

The NQT with its many facilities is a powerful system from which objective and reliable information about the quality of service of a telephone network can be obtained. Deterioration of the quality of service or the widespread occurrence of a particular type of fault can be detected immediately from the detailed reports. Weak points in the network can also be spotted. Troubleshooting of faults on transmission circuits or in exchanges is made easier because each test unit can also be operated manually.

Rolf Dietschi was born in 1954 and received his training as a Grad Eng of the Swiss Federal Institute of Technology at the Intercantonal School of Engineering in Rapperswil. He joined STR in 1979, where he has since been engaged in the development of hardware and programs for TSS (telecommunication service system) products.

Christoph Gessler was born in 1955; he is a Grad Eng of the Swiss Federal Institute of Technology (ETH). He joined STR in 1980 after working for two years as an assistant at the Institute for Electronics of the ETH in Zurich. Mr Gessler is at present working on the development of the NQT test unit programs, primarily on microprocessor support.

Edwin Staber was born in 1936 and is a Grad Eng of the Senior College of Technology. He joined the installation department of STR's communication switching technology division in 1967, where he worked on quality promotion and test equipment. Since 1976 he has actively participated in the development of TSS products. Mr Staber is at present project manager for TSS products, including the NQT system.

Videotex System for Trial Service in Switzerland

A trial public Videotex service will start in Switzerland in 1983/84. The trial will be used both to test the system itself and to determine the reaction of information users and providers.

Ch. A. Maurer
Standard Telephon und Radio AG, Zürich, Switzerland

Introduction

The International Telegraph and Telephone Consultative Committee (CCITT), has defined Videotex as a computer-based information and communication system for use by the general public. The first Videotex system was developed in the United Kingdom at the beginning of the 1970s under the name Viewdata[1].

All Videotex services involve three types of participant: information users, information providers, and the system operator. The information user, or Videotex subscriber, is connected to the Videotex center via the public switched telephone network, enabling him to dial up the center and then selectively access textual and graphics information from a wide range of items using a simple dialogue procedure. The requested information is generally displayed in the home on a television set equipped with a special Videotex decoder, or in the office on a special Videotex terminal. Information is stored digitally in the Videotex center. Thus a modem is provided at the center to convert the digital information into a suitable format for transmission to the subscriber over a normal telephone line. On the subscriber side, the information is again digitized by a modem, and then converted into a video image. Video images are not stored and transmitted directly, but in a compressed, coded form.

The dialog procedure enables the Videotex system to provide other facilities, such as a message service between information users, in addition to the basic accessing of information.

In Switzerland, STR (Standard Telephon und Radio) was the first company to undertake comprehensive development of Videotex. As a result, STR was able to supply a complete turnkey pilot system[2] to the Swiss PTT in 1979. The system supplied was based on the original UK Viewdata system (now Prestel) with minor modifications to cater for Swiss requirements.

The Videotex idea has attracted considerable attention internationally, which has encouraged three aspects of the original Viewdata concept to be developed further in various countries.

In Europe, screen display options were improved within CEPT (European Conference of Postal and Telecommunication Authorities)[3]. In Canada, a display technology called Telidon was developed and later taken up in the United States of America, offering considerably more facilities than the Viewdata system and the new CEPT standard. Unfortunately, the new European and North American systems are not yet compatible, even though STR has demonstrated a dual-mode system[4].

Second, by connecting external databanks to a Videotex center, it will be possible to extend the range of information, as in principle this makes it possible to access information stored in electronic data processing systems. Transaction-oriented

Videotex center with a dual VAX 11/780 configuration.

services (e.g. the ordering of goods from a mail order company) considerably enhance the attractiveness of a Videotex service.

Third, the capacity of a single information update center will not be sufficient to cope with the large number of subscribers expected to use the service, so the service functions will be distributed between several regional centers.

These trends were discernible when the pilot system was ordered. The Swiss PTT therefore decided to use the pilot system to obtain initial experience, but not as the basis of a Videotex service. Tenders for a trial system were requested in 1980, following the pilot test, in which the above trends were to be taken into account.

System Concept

The Videotex system designed for the trial service consists of two Videotex centers (see Figure 1) connected via the Telepac network, the Swiss PTT's packet switched data network. A subscriber can access the center either by dialing in via his telephone set or via the Telepac network[5]. In the longer term, a wide variety of subscriber categories, offering different functions and transmission speeds, can be connected by these access routes. Initially three subscriber interfaces will be provided:

- Interface for information users (A1) who access the Videotex system via the public switched telephone network. Data exchange will be at the same speeds as those used in the pilot test (i.e. 1 200 and 75 bit s^{-1}).
- Interface for information providers (B3) who also access the system via the public switched telephone network. The transmission rate for this category is 1 200 bit s^{-1} in both directions.
- Interface for information providers with an external databank computer (C4) who communicate with the Videotex system via the Telepac network. This category of subscriber can select any transmission speed from the Telepac range. The speed chosen for the Telepac connection with the Videotex center does not therefore have to correspond to that of the external computer connection.

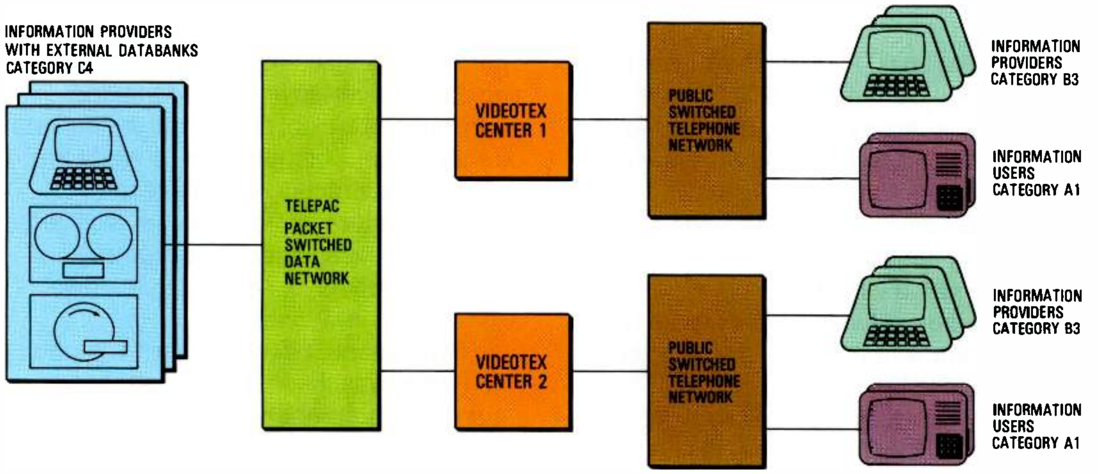


Figure 1
Concept of the Videotex system developed by STR.

In Switzerland, information and subscriber data storage, and data processing for the Videotex network, are distributed between several regional Videotex centers. Each subscriber is assigned to a regional, or home center which provides the following facilities:

- storage and management of subscriber records; these are used to check for right of access and for billing
- storage of incoming messages
- support functions to help information providers to enter information into the integral databank of the Videotex center
- support of external databanks connected via the Telepac network.

Each Videotex center has direct access to all external databank computers, enabling it to pass information between its information users and any databank in the country.

Information Storage Facilities

Information that is to be accessed by users can be stored either in one or in all internal databanks at individual Videotex centers, or in external databanks. The internal databanks at all Videotex centers make up the Videotex network integral databank, which has one common number range (see Figure 2). Part of this integral databank is operated by the PTT, and is used to guide information users to the information offered by individual providers of information. This part of the integral databank is called the operator subtree. The remaining capacity of the integral databank is subdivided and assigned to individual information providers for their exclusive use and organization (assignment of one or more entry pages). Information providers in the B3 category store their information in their allotted number range. Based on the numbers of enquiries from various regions, they can decide whether this information should be stored only in their home center or in all centers. Information stored in the home center is also available to users assigned to other centers, although access times are longer.

Information providers with external databank computers (category C4) can also use their allotted number range for storing part of their information in the integral databank. The minimum they have to store in all the Videotex centers is their entry page, which is also their gateway page and thus enables users to access information stored in the external databank.

Since three languages are used in Switzerland, the entire number range of the integral databank is available in all three languages. Information providers are therefore able to structure their information in all three languages simultaneously (i. e. to use the same page numbers in each language). The three sections used for the various languages can, however, have different contents.

Interface for External Databanks (C4)

To be connected to the Videotex trial service as an external databank, an information provider must have the necessary equipment for supporting on-line sessions initiated by the Videotex center. As well as an X.25 hardware and software interface, an enhanced Prestel gateway protocol has been defined for Switzerland to take advantage of the latest CEPT presentation level functions in conjunction with advanced transactional services.

Within the number range allotted by the integral databank, an operator of an external databank can arrange the gateway pages in any sequence so as to define various entry points into the information or service range stored in the external databank. Information stored in the external databank can be structured as required. In particular, the external databank can use a different numbering scheme from that of the integral Videotex databank. Pages which are stored in the integral databank must, however, lie within the allotted number range.

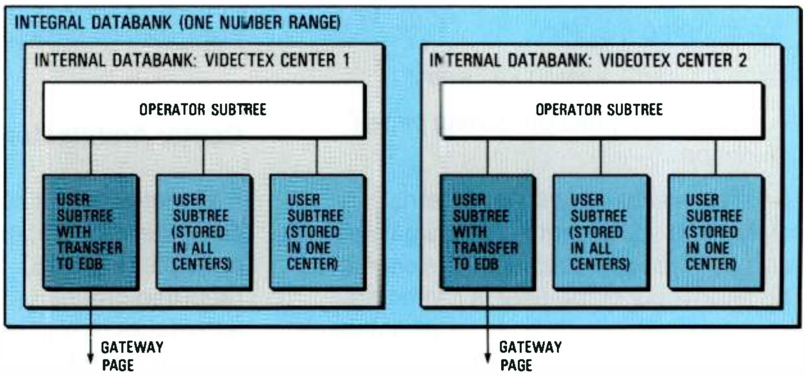
In addition to storing information, an external databank can offer transaction services backed up by the advanced CEPT data collection function in the Videotex center. Thus an external databank must contain data collection pages within its range which can be retrieved by information users. The center then supports dialog with the user, helping him complete (fill in) data collection pages by providing form-filling rules and plausibility checks. Inputs from the information user are stored in buffer memory in the center, and forwarded to the external databank only after the user has been able to check his entry and approve transmission of his data.

Welcome page when accessing STR's allocated number range on the integral database.

Interface for Information Providers using the Telephone Network (B3)

In order to participate as an information provider in category B3, a subscriber must

Figure 2
Organization of the integral databank.



be equipped to prepare information autonomously. This requires equipment for off-line editing of single pages, and for structuring pages into subtrees. Information trees prepared in this way are then transmitted in the form of a file to the Videotex center via a telephone line (bulk update). The received file is first stored in buffer memory, from which the integral databank is then updated according to instructions from the information provider, who can also choose when the update will take place. The center issues a report when the information has been updated; this report can be called up by the information provider. Protocols for bulk updating have been prepared jointly by the Swiss PTT and STR as a basis for the bulk updating of information subtrees from external databanks via the Telepac network.

Unlike subscribers in category C4, subscribers in category B3 can also use the information user (A1) functions within an information provider session without logging off. The PTT will be organizing and managing the integral operator subtree via the B3 interface at each center. STR will be supplying the four new CEPT database production and management systems necessary for this function.

Interface for Information Users (A1)

To take part in the trial service, information users will require a television set with CEPT decoder, a numeric keyboard equipped with the keys * and #, and a special Videotex modem.

The functions described below can be used both via the subscriber's own connection (i.e. in connection with the home center) and any other connection to the public telephone network. The following options are available for calling up pages from the integral databank:

- menu selection using one or two digits followed by the clearing signal (#)
- page number entry
- alphanumeric search key entry
- using the 'request preceding pages' function.

To set up a session with an external databank, a subscriber selects a gateway page and then enters at least one character. The anonymity of the user is protected. During a session with an external databank, the Videotex center supports the information user in filling in data collection pages, if offered by the external databank. The user has the option of interrupting the session at any time.

Information users can also take advantage of the message service for the input and delivery of prepared or free-format messages to one or more addressees in the same or a different area. When sending a message, the period for which it remains valid should also be defined. After this period has elapsed, the message is automatically erased in the home center at the receiver's end.

Messages can be called up either sequentially in the order of arrival, or by means of a menu. The menu thus permits direct access to a specific message within the waiting queue. Retrieved messages can be stored in a library for further use.

The password needed to check right of access can be defined by the user and altered at any time in a dialog with the center. The subscriber can also choose his dialog language on the screen. The information user can also determine whether any alteration of the dialog language should be valid until the end of the session or until it is canceled.

Differences between Pilot Test and Trial Service

The Videotex trial service will differ from the pilot test in several ways. It will provide a full CEPT transaction service for external databanks and gateway support, as well as alphanumeric keyword access to pages, off-line editing and bulk updating of information, improved message service, and advanced data collection features in collaboration with external databanks. In addition, the trial offers multilingual user dialogs and databases, closed user group facilities, and display of page prices and billing. The range of services available can be favorably compared with the features offered by planned public services in other European countries.

Center Architecture

A Videotex center consists of a dual computer system, operating on the standby principle (Figure 3). For security, data is recorded on-line on two Winchester disks; a third disk is kept as a backup. In addition, each computer has its own system disk, and is equipped with its own X.25 interface for reliability. A common communication subsystem is used to the telephone network. Since inputs to the communication subsystem from the telephone network are occupied in random sequence, this subsystem has sufficient spare inputs to handle the planned traffic load even if faults should occur.

The traffic connection capacity (without spare inputs) of the Videotex center is 48 ports to the telephone network for category A1, 16 ports to the telephone network for category B3, and one CCITT X.25 port to the Telepac network operating at 48 kbit s^{-1}. With the expected volume of traffic, this configuration allows at least 1 000 subscribers to a Videotex center. The 16 category B3 ports correspond to the number of information supplier ports in the pilot test. Traffic using these ports should therefore be easily handled, since information input sessions will in future be much shorter. The CCITT X.25 connection is more than amply dimensioned for the expected initial traffic. Centers can be extended at any time to cater for increased traffic or a larger number of subscribers.

The memory capacity of the internal databank is approximately 50 000 pages and approximately 7 500 message units. This capacity can be extended at any time by connecting bigger additional peripheral equipment.

Conclusions

The first Videotex center for the public trial is planned to go into service in Berne in November 1983, by which time the system will have undergone full operational tests.

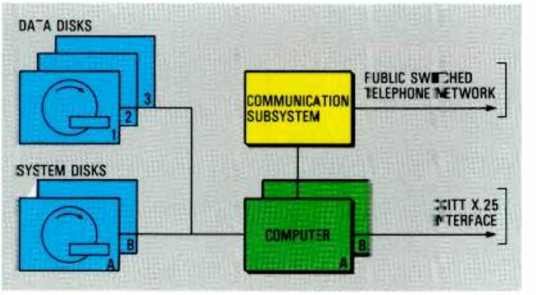


Figure 3
Architecture of a Videotex center for the trial service in Switzerland.

The second center, in Zürich, is scheduled to start operation in May 1984.

The trial service will include a parallel investigation into user acceptance. The number of users is expected to increase only slowly in 1984 due to poor availability of the new CEPT decoder, but thereafter numbers should increase more rapidly.

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5 M. Schaeren: Das Pilotnetz Telepac: *TMt PTT* 60, 1982, pp 81 – 92.

Ch. A. Maurer was born in 1951 and completed his studies in electrical engineering at the Swiss Federal Institute of Technology, Zürich, in 1975. He then worked for a telecommunication company on remote control and security systems (hardware and software). Since joining STR in 1980, he has been involved in the development of information systems. In August 1982, Mr Maurer became project manager for public Videotex systems.

Standard Radio & Telefon AB

Standard Radio & Telefon (SRT) is ITT's associate company in Sweden. The company, which started operations in 1938, now has around 900 employees and a turnover in excess of 50 million dollars. An important factor behind the company's evolution is the Swedish home market – one of the most sophisticated in the world in terms of communication and manufacturing technologies.

The focus of SRT's operations is the fast-growing telecommunication industry. The company develops, manufactures, and supplies products and systems that utilize advanced electronics, computer, and programming technologies to solve vital communication problems.

Military radio communication is an important product area. Research and development are frequently undertaken in close cooperation with the Swedish defense authorities, giving a good base for high quality products. In many countries, complete communication networks have been based on SRT's short-wave equipment. The company's radio systems can be installed in vehicles or in special radio cabins, operated by remote control, and equipped for various kinds of communication (speech, telex, facsimile, morse, and data). One application in which SRT's short-wave radio systems have been given special recognition is as a link between various countries' foreign ministries and their embassies.

The company is a specialist in the development of high frequency point-to-point radio equipment and can offer complete systems for various applications, including receivers and transmitters, peripheral equipment, installation, and training. The modular, solid state equipment features frequency synthesis and built-in fault detection, thereby ensuring high frequency stability, excellent reliability, and low maintenance costs.

Way back in the era of the spark transmitter, SRT was an established supplier to the shipping industry and is now a leader in the field of marine radio equipment. In the 1960s the company launched single sideband telephony, which is today mandatory. In the 1970s SRT introduced automatically tuned transmitters, and is now making the system of the eighties for automatic unattended ship-to-shore telex communication.

SRT is one of the world's largest manufacturers of marine high power 1.5 kW single-sideband/telex/telegraphy transmitters. All new equipment, including the ST950 automatically tuned transmitter with full remote control and the PNW20 Navtex receiver, is fully solid state. One in four new ocean-going merchant vessels has SRT radio equipment on board.

Transmission equipment has been another important product area for 30 years or more. Data modems, for which SRT has a complete product programme, represent the most extensive product group. Handling transmission speeds ranging from 300 to 64 000 bit s^{-1}, these modems can be used in public and private communication networks and data networks.

Multiplex equipment for telegraph and telephone transmission and subscriber carrier equipment is bought by PTTs in Scandinavia, and recently USTS, part of the ITT COINS group in the USA, has become a major purchaser of multiplex equipment.

SRT's loudspeaking internal communication systems are used by more than 100 000 users worldwide. The new microprocessor-controlled ITT 511 E represents the third generation of intercom systems, offering expansion up to 4000 stations.

Over the years SRT has been a major supplier of telephone equipment to the Swedish market. Through the Swedish PTT, the company today offers telephone subscribers a complete range of terminal equipment.

In many ways, Sweden is a pioneering country on safety issues. An example is the ATC (automatic train control) system developed for the Swedish State Railways to improve safety on lines carrying particularly heavy traffic. This fully automatic signaling and communication system gives the driver a completely automatic "assistant".

Under the banner of ITT Data, SRT supplies data terminals and network systems in Sweden and Finland. ITT Data's aim is to satisfy the customer's total data communication requirements, from mainframe computers via the data network to individual terminals. By virtue of ITT's coordinated international resources, users in Sweden and Finland have access to complete system know-how in this area.

The majority of the products manufactured and marketed by SRT have been developed entirely within the company. Product development is a precondition for survival. Research and development are therefore central to the company's operation. More than one in every eight employees is engaged in new product development.

P. O. Lindholm
Managing Director
Standard Radio & Telefon AB
Vällingby, Sweden

Flexible Data Circuit Terminating Equipment for Circuit-Switched Data Networks

Data circuit terminating equipment has more functions than a modem, including envelope formatting and supervision. New equipment designed for the Nordic public data network provides all the necessary functions while retaining flexibility to operate with practically any type of line and many different terminals.

T. Hedberg
Standard Radio & Telefon AB, Vällingby, Sweden

Introduction

The need for data communication has grown rapidly over the past ten years. When the telephone network was first used for data transmission, the conversion of digital signals to line signals was performed by modems designed to the CCITT V-series of recommendations. However, data communication users have different requirements from those of a telephone subscriber. Typically they require quick and reliable connection and disconnection, low bit error rates even at high speeds, and a high degree of data security. To meet those new requirements, private data networks were built, mostly using lines leased from the local Administration.

To provide data users with the benefits of a general switched network similar to the public switched telephone network, in the early 1970s Administrations began to plan the introduction of switched data networks. One consequence was the CCITT X-series of recommendations, which aims specifically at data networks just as the V-series relates to the use of the public telephone network for data communication.

In Scandinavia, the Nordic public data network started operation in 1980/81. This circuit-switched network uses DCE (data circuit terminating equipment) developed by SRT (Standard Radio & Telefon) for this network. However, the differences between circuit-switched networks are generally small as a result of early standardization within CCITT.

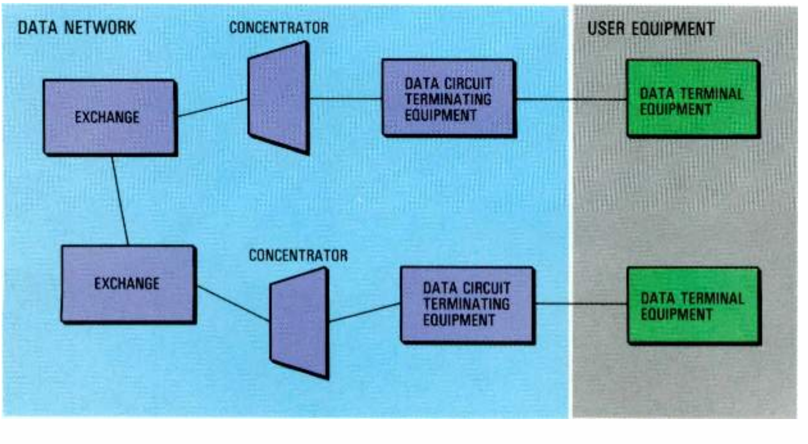


Figure 1
Basic schematic of a data network showing how data circuit terminating equipment is connected.

Description of the DCE

Functional Overview

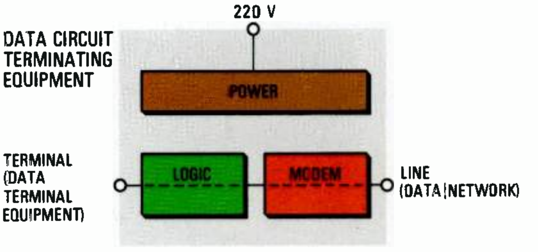
The DCE is an integral part of the data network, performing both signal conversion and supervisory functions. Indeed, as Figure 1 shows, it is the connection port to the data network. Traditionally, the DCE has been a modem which adapts the digital signals to the characteristics of the transmission medium. In a circuit-switched data network, however, the DCE must perform additional functions, including:

- envelope synchronization
- network supervision
- signaling conversion for different terminals.

The functions of the DCE can thus be divided into two groups: modem functions which adapt the signals to the line characteristics, and logic functions which adapt the DCE to the DTE (data terminal equipment).

The DCE developed by SRT is designed to be flexible, which implies that it should be adaptable to different lines and different DTEs. This has been achieved by a modular design that allows different logic units and modem units to be assembled into a complete DCE (Figure 2).

Three logic units provide the DTE/DCE interfaces shown in Table 1. In addition there are four modem units plus a modem adapter for external modems that cannot be housed in the DCE cabinet (Table 2).



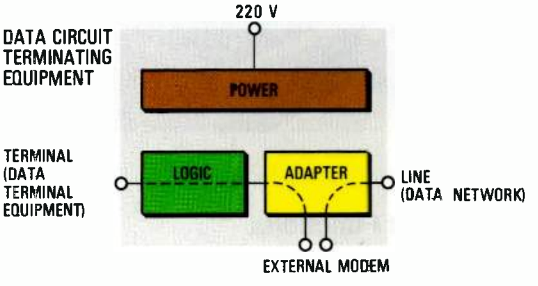


Figure 2
Modular design of SRT's flexible data circuit terminating equipment.

Supervision Functions

Several functions are built in for supervising the DCE's performance and, in the event of a failure, rapidly detecting and locating the fault:

- test functions, initiated either by the subscriber or from the network
- power-off signal generation, which raises an alarm when the DCE is unpowered and performance tests cannot be carried out
- alarm and control signal generation.

The two tests performed from the subscriber or network side are basically the same. They exercise practically all DCE functions as well as the subscriber line. The test sequence varies slightly depending on the type of logic unit, but includes operations such as disconnection, connection, and data transmission.

The power-off generator is powered via the line. When the mains supply fails, it starts sending a *simulated carrier* which is keyed on/off at 1.7 Hz, causing the carrier detection circuit in the station end modem to toggle. This is recognized as a power-off indication.

When a V-type DTE/DCE does not want to be accessed (e.g. during maintenance), a *local* button can be actuated which generates a *local signal* to the station. This bars incoming and outgoing calls, but does not initiate any alarms.

An *alarm signal* is sent to the network when the DCE is out of synchronization. The signal, a stream of *ones,* causes the station-side alignment circuitry to lose synchronization, thus giving the correct fault indication. In the same way, loss of the incoming signal suppresses the outgoing signal from the DCE, giving the fault diagnosis *no line signal.* Together, these functions enable the network management system to detect practically all faults within seconds and to monitor performance by testing each subscriber connection.

DTE/DCE Interfaces

The DCE supports essentially every kind of DTE, from asynchronous (start-stop) terminals to the latest CCITT X.32/X.25 computer-based DTEs.

Both the transmit and receive clocks are controlled by the DCE, which is in turn controlled by the incoming bit rate. That is a consequence of the synchronous nature of the data network. The CCITT X.21 interface is designed for such networks: it only provides clock signals from DCE to DTE.

Table 1 — DCE logic unit characteristics

Logic unit type	Interface	Mode	Subscriber bit rates (bit s^{-1})
X logic	X.21	synchronous	600, 2400, 4800, and 9600
VP logic	X.21 *bis*	synchronous	600, 2400, 4800, and 9600
VPC logic	X.20 *bis*	asynchronous	50, 100, 110, 134.5, 200, 300, 600, 1200, 2400, and 4800

Table 2 — DCE modem unit characteristics

Modem unit type	Application	Modulation	Bit rates (bit s^{-1})
SCM biphase	short-haul	biphase	750, 3000, 6000, and 12000
SCM AMI	short-haul	AMI	6000
LCM 750	long-haul	frequency	750
LCM 3000	long-haul	phase	3000
LCMA	adapter	–	6000 and 12000

SCM - short-haul customer modem
LCM - long-haul customer modem
LCMA - long-haul customer modem adapter.
The term customer implies that it is located at the subscriber end.

Data circuit terminating equipment used with a video terminal.

Logic Unit Functions

Envelope Synchronization

The underlying requirements for the DCE (e.g. data transparency), require a higher bit rate between the DCE and the network than between the terminal and the DCE.

The interface between the DCE and the network is essentially a serial form of the CCITT X.21 interface, as Figure 3 shows. Eight data bits and a status bit *S* are assembled in a 10-bit envelope. The tenth bit *A* is an envelope alignment bit which has an alternating *0101*... pattern. In the transmit direction, the logic units generate the alignment bit pattern and assemble the envelopes, while the inverse operations are performed in the receive direction. The algorithm for finding the alignment bit is a field-proven technique based on a state model with five states.

Network Supervisory Functions

Network supervision functions are performed by the logic unit, except for the

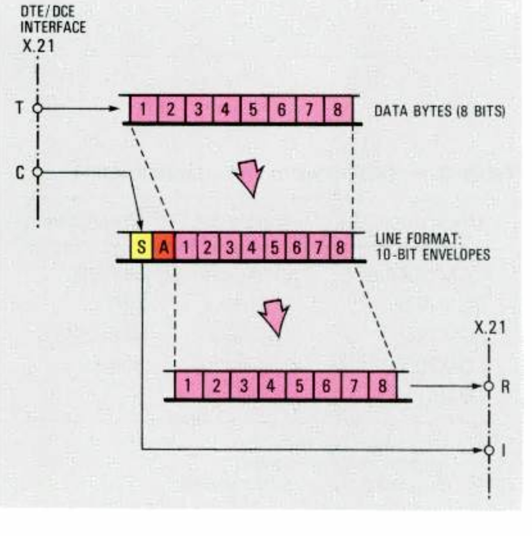


Figure 3
Envelope formatting and deformatting for the CCITT X.21 DTE/DCE interface.
T - transmit
C - control
R - receive
I - indication
A - alignment bit
S - status bit.

power-off generator. The test procedure from the DCE is very simple. One of several keys is pressed, and the optical indicators are checked. This test can be performed by the data subscriber, thereby minimizing unnecessary service calls.

Signaling Conversion

Signaling conversion differs markedly between X and V (VP and VPC) type logic units. The X-type unit does not carry out any conversion since the DCE/network interface uses a serial form of X.21 (see Figure 3), and is therefore transparent in normal operation. Exceptions are the test mode and loss of synchronization.

The X-type logic unit handles both synchronization options specified in Recommendation X.21. In one alternative, the DTE sends control signals synchronously with the *byte timing signal* provided by the DCE. In the other, the DTE sends control signals without byte synchronism, and the logic unit must align on the preceding synchronization characters. Outgoing control characters are thus always adjusted so that each envelope contains one complete character.

The V-type logic units perform V.24/X.21 conversion. Originally the V-type interface was intended for use with a subset to provide dialing facilities. In a data network this function must be integrated in the DCE (or DTE), and therefore V-type logic units are equipped with a 16-key keyboard and a 2-digit call progress display. Call progress signals show the cause of a connection failure; for example, congestion, wrong (incompatible) bit rate, closed user group. Depending on the exchange functions, several dialing methods, from full number selection to direct calling, can be used. Direct calling from the DTE is also a strap option.

Further strap options allow control of the DTE/DCE interface to be adapted to match practically any terminal.

The VPC-type logic unit consists of a VP logic unit with a plug-in asynchronous/synchronous converter. The data stream on the network side is always synchronous. Again, straps are used for adaptation to different bit rates, character lengths, and stop bit lengths. The converter transforms each asynchronous bit rate to the next higher synchronous rate.

Block schematics of the logic unit are shown in Figures 4 and 5. All the main functions are performed by a microprocessor.

Description of Modem Units

Present DCE modems are all of the 4-wire type. Both transmit and receive lines are designed to carry a DC current, which has two functions: to reduce the risk of a bad contact for low-level signals and to power the power-off generator.

The power-off generator is controlled by the power unit. When the mains supply fails, the transmitted line signal is disconnected and the power-off generator is turned on. This generates a 1 800 Hz sinewave in the LCM750, LCM3000, and LCMA (adapter) modems, and a 1 000 Hz squarewave in the short-haul SCM biphase and SCM AMI modems. The signal is modulated on/off at a rate of 1.7 Hz.

As a result of the very high signaling rate in CCITT Recommendation X.21, special protection circuitry against sequences of *zeros* is incorporated in all modems with the exception of the LCMA adapter. If a sequence of 20 or more *zeros* is generated (e.g. due to a short loss of carrier), this is interpreted as a clear signal, which clears the connection. The protection circuitry therefore inhibits the *all zero* signal when the line signal is interrupted. In the LCM750, LCM3000, and SCM biphase modems, a particularly quick-acting carrier detector clamps data to 'binary *1*' when the carrier is lost. In the SCM AMI modem, this technique is impossible to implement, so a type of scrambler is used that inverts every bit following three consecutive *ones* or *zeros*.

The transmit line signal is controlled by the modem which squelches the transmit signal if both the receive line signal detector and the internal *request to send* signal from the logic unit are *off*.

All modems have a front test point that allows the eye pattern of the received signal to be monitored during installation.

Long-Haul Modems

The LCM750 is a frequency shift keying modem designed according to CCITT

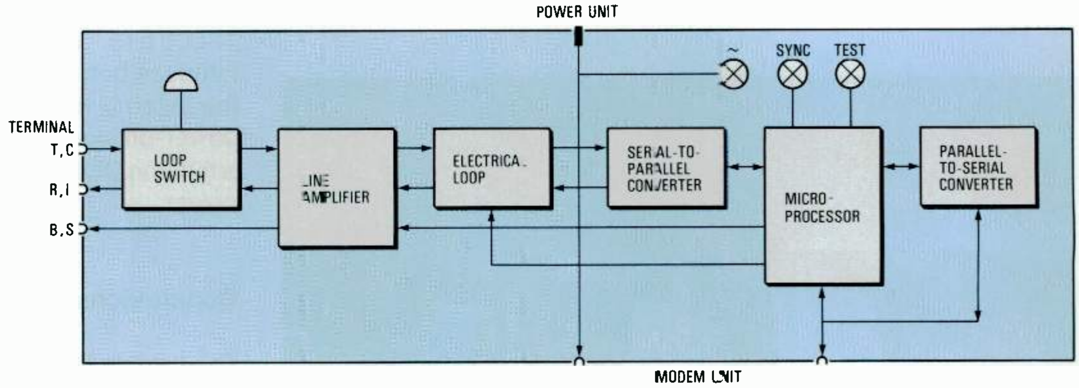


Figure 4
Block schematic of the X logic.
B - byte timing
S - signal element (bit) timing.

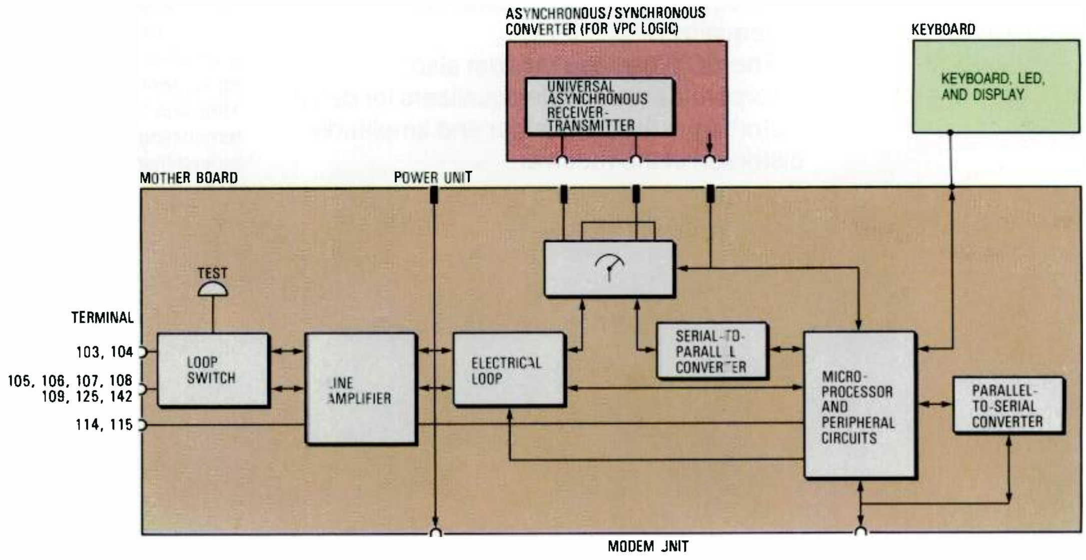


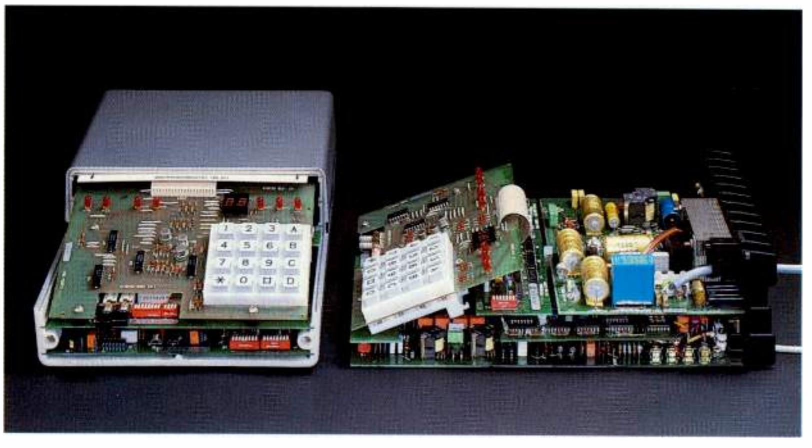
Figure 5
Block schematic of the VP/VPC logic. The designations on the left hand side are the pin numbers in CCITT Recommendation V.24. From top to bottom they are for data, control, and timing signals.

Recommendation V.23, but with 20% higher bit rate.

The LCM3000 is a V.26 phase shift keying modem which incorporates equalizers for up to three carrier frequency sections in each direction, and an amplitude equalizer in the receive direction. If further equalization is required (e.g. for loaded lines), it is provided by the station side modem.

The transmitter incorporates a microprocessor.

Views of the data circuit terminating equipment with the VPC logic.



Short-Haul Modems

The SCM biphase modem employs the biphase-space modulation method. Straps can be selected for transmission rates of 750, 3000, 6000, and 12000 bit s^{-1}. It is possible to transmit the two lowest bit rates over loaded cables. When a mix of loaded and unloaded cables is used, impedance transformers connect the two line types, causing attenuation distortion at the low frequency end. This has been overcome by incorporating a fixed low frequency boost circuit in the transmit direction as a preequalizer.

The SCM biphase modem also incorporates strappable equalizers for delay distortion in the transmitter and amplitude distortion in the receiver.

The SCM AMI modem is specifically designed to transmit 6000 bit s^{-1} on loaded cables. It therefore employs the AMI (alternate mark inversion) modulation technique, which requires half the bandwidth of biphase modulation. This modem uses the same strappable equalizers as the SCM biphase modem. Transmit signal shaping and equalization are performed by a digital transversal filter.

Modem Adapter LCMA

The LCMA adapter operates with 6000 and 12000 bit s^{-1} modems, providing the interface between the built-in logic unit and the external modem, and generating the power-off signal. The latter is turned *on* if either the DCE or the external modem loses power.

Conclusions

The flexible DCE developed by SRT is able to cater for both old and new subscriber terminals. Further development can be envisaged. For example, the present 4-wire modems could be replaced by full duplex, 2-wire modems, thereby saving lines.

A full performance model of a short-haul modem for the DCE using echo canceling has been demonstrated. It operates at 750, 3000, 6000, and 12000 bit s^{-1}.

The X-DCE can be used for X.25 terminals, which operate in the packet mode. In this application, the circuit-switched network is used as a carrier to the packet switching exchange. The connection to the packet switching exchange can be set up using the X.21 direct call procedure.

Tomas Hedberg was born in Östersund, Sweden, in 1950. He studied electrical engineering at the Royal Institute of Technology in Stockholm, where he graduated in 1975. He worked with digital transmission equipment at Ericsson until 1979 when he joined SRT. In 1980 and 1981 he was project leader for the data circuit terminating equipment project. Mr Hedberg is currently project manager for the development of modems.

Surveillance and Communication Receiver CR91

The CR91 complies with all the requirements for a communication and sweep and scan receiver. It is based on the state-of-the-art receiving section of the CR90, together with comprehensive sweep and scan facilities. The latter can be tailored to the user's needs simply by using different program modules.

R. G. Jonsson
Standard Radio & Telefon AB, Vällingby, Sweden

Introduction

Standard Radio & Telefon (SRT) has been designing and manufacturing HF (high frequency) communication equipment since 1940. The current HF radio equipment range offers users a comprehensive set of compatible units – transmitters, receivers, antenna tuners, demodulators, remote control units, etc. Together these building blocks form the System 90 radio communication range.

The latest addition, the CR91 receiver, has been designed to meet the requirements for communication, surveillance, and search receivers. The electrical performance meets the highest standards for receivers that are to be used in environments where strong interference is present. The front panel layout and operation have been optimized for surveillance and search applications. The CR91 receiver, which covers the frequency range 10 kHz to 30 MHz, is designed for continuous service in fixed, mobile, or shipborne applications.

Front panel layout of the CR91 receiver showing the logical layout of controls.

Electrical Design

The receiver circuits can be divided into three groups: RF (radio frequency) circuits or signal path, frequency generating circuits, and control circuits. The RF circuits and frequency generating circuits are identical to those of SRT's CR90 communication receiver, but the control circuits have been completely redesigned using microprocessor techniques. Figures 1 and 2 show the CR91 receiver and its control unit.

Signal Path
In the range 1.6 to 30 MHz, the required input selectivity is obtained by nine suboctave filters that are switched by PIN diodes. The suboctave filters give protection against second-order intermodulation products by attenuating the signals causing the intermodulation. Below 1.6 MHz, a low pass filter is used. A 20 dB antenna input attenuator can be switched in manually from the front panel, and is switched in automatically if the input signal exceeds 4 V. In this way the input attenuator forms part of the input protection circuit and can handle a sustained input overload of 60 V into 50 ohm.

For operation at the very low frequency end of the range, the antenna input can be switched to a high input impedance unit gain amplifier, enabling useful input signals to be obtained from electrically short open antennas down to 10 kHz.

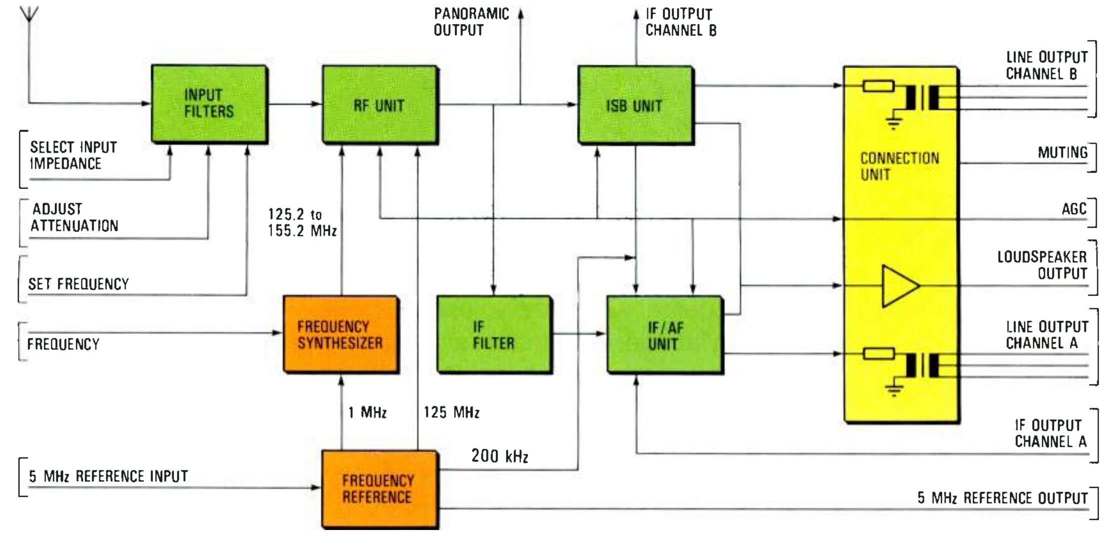


Figure 1
Block schematic of SRT's CR91 communication and surveillance receiver.

In the RF unit, the signal is first amplified by about 8 dB in a wideband gain-controlled amplifier before being input to the high power double balanced FET (field effect transistor) mixer and up-converted to the first IF (intermediate frequency) of 125.2 MHz. At this frequency, selectivity is introduced by one 2-pole and one 4-pole monolithic crystal filter. Partial compensation for filter loss is provided by an isolation amplifier between the filters. The signal is then down-converted to the second IF at 200 kHz and passed through a mechanical roofing filter*. The bandwidth at this point is 6.8 kHz.

The RF unit provides three outputs: panoramic output (for displaying a range of frequencies around the chosen frequency) and one output for each IF amplifier. The IF/AF (audio frequency) board or channel *A* IF amplifier is preceded by a filter unit which provides the final selectivity using mechanical filters. These filters, which incorporate piezoelectric transducers, have excellent selectivity and minimize intermodulation products. The selectivity of the channel *B* IF amplifier is provided by the optional ISB (independent sideband) board using a mechanical filter that extracts the lower sideband from the IF signal.

Almost all the receiver gain is provided by the IF amplifiers on the IF/AF board (channel *A*) and the ISB board (channel *B*). Both these amplifiers incorporate AGC (automatic gain control), so that the level at the AM/SSB (amplitude modulation/single sideband) detectors is virtually constant. Delayed AGC is fed back to the RF amplifier to improve the overall handling of high amplitude signals. Constant level IF outputs are provided for channels *A* and *B*, as well as audio line outputs that can be adjusted continuously by +/−10 dBm. A loudspeaker is provided on the front panel and there is an external loudspeaker output at the rear. The loudspeaker can be switched between channel *A* and *B*, or switched off.

* A filter with mechanical resonators and coupling; the term *roofing* implies that the overall bandwidth cannot exceed that of the filter.

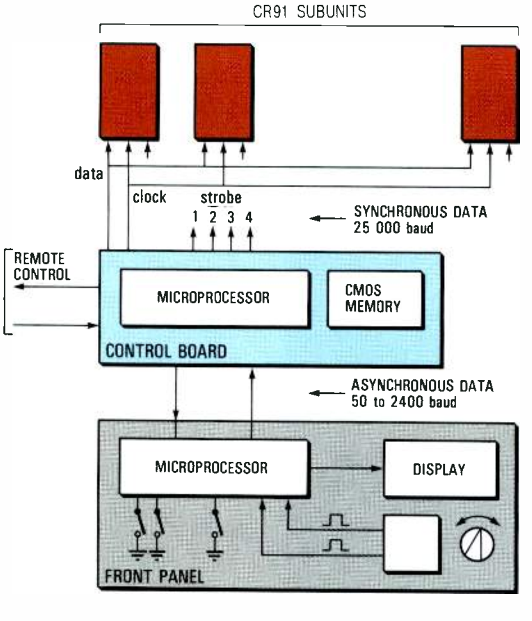


Figure 2
Control unit for the CR91 receiver.

Frequency Generation

All the frequencies that are generated, including the beat-frequency oscillator and 125 MHz second local oscillator frequencies, are derived from the 25 MHz VCXO (voltage controlled crystal oscillator) on the refenerence board. This oscillator is phase-locked to the built-in 5 MHz reference oscillator or to an external frequency standard.

The synthesizer covers the band from 125.2 to 155.2 MHz in 1 Hz increments.

The high Q inductance-capacitance circuit in the VCO (voltage controlled FET oscillator) is band switched in 32 steps by PIN diodes and tuned continuously by two low-loss varactor diodes. The VCO is controlled by a single phase-locked loop that utilizes a new phase interpolating technique. Thick-film hybrid technology has made it possible to construct the synthesizer on a single printed board.

Control Circuits

The receiver subunits are controlled via a high speed synchronous serial bus on the motherboard. The serial bus carries data, clock, and four strobe signals. The bus signals are connected to serial-to-parallel converters (shift registers) in the subunits. In this way only three pins on the synthesizer board connector are required to enter 30 bits of frequency information.

A microprocessor on the control board assembles information that is set up on the front panel into messages for transmission on the serial bus. This microprocessor communicates in serial format with the front panel and a remote control/computer control input using standard ASCII code. A battery backed-up CMOS memory provides nonvolatile storage for 100 programmable channels. System programs include the nucleus of a real-time multitasking operating system with interrupt driven input/output handlers for the serial ports.

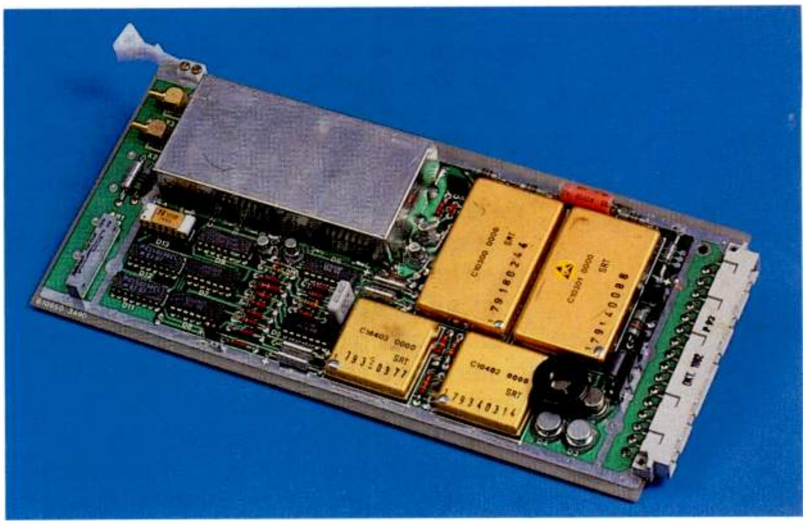
The front panel functional unit is also built around a microprocessor. All communication with the receiver takes place in serial form using ASCII code, normally at 2400 baud. When a key is pressed, an ASCII character is sent to the control board in the receiver where the memory image of the receiver setting is changed accordingly, and the ASCII character is echoed to the front panel. Echoing of the character updates the display, thereby confirming the new setting.

Analog functions are handled via the serial port to ensure maximum design freedom when the receiver is used as a building block in larger systems. On the CR91, the analog functions (main tuning knob and potentiometers) appear to be independent and 'live' to the operator just as if they were coupled directly to the receiver subunits.

Systems Approach

A radio operator and the receiver form a 'system' with specific interface requirements. The receiver must have suitable controls and displays, as well as an ergonomic panel layout, if the operator is to be able to work effectively and without fatigue. Similarly, the radio operator must have relevant training.

In a modern receiver application, the receiver must interface to a system with many parts. Using the 'systems' approach, this is recognized as the most important guideline during design. The CR91 receiver represents a total commitment to these ideas and every effort has been made to identify and meet all system interface requirements.



Synthesizer printed board for the CR91 receiver; wide use has been made of thick-film hybrid technology to keep the synthesizer compact.

Remote Control

Remote control over telephone lines or a radio link is possible using the remote/computer control serial port, provided there is a modem at the receiver end. At the remote end, the control unit has a front panel identical to that of the receiver; as communication between the remote control unit and the receiver is identical to that between the local panel and the receiver, the operator cannot tell the difference between the two operating positions.

Changeover between local and remote control can be initiated from either end. In the local mode, the receiver also monitors the remote port and if the correct character (ASCII *X*) is received, control is relinquished to the remote port. The same result is obtained by pressing the REM key on the local panel. Control can be regained by the local unit by pressing the DIR key on the local panel. As before, the functions appear identical at the remote end (i.e. control can be taken over by the remote operator by pressing the DIR key on the remote panel).

Pressing the REM key on the remote panel returns control to the local operator.

If the receiver is used for surveillance under remote control, the signaling speed on the control line should be 1200 or 2400 baud if the maximum achievable search rate (in kHz s^{-1}) must be used. At lower signaling rates, the search speed is reduced. If the receiver is used as a communication receiver, response time is not so critical and the signaling rate can be reduced. Any popular speed down to 50 baud can be selected, allowing several control signals to be multiplexed over the same line.

Computer Control

Computer control uses the same serial port as for remote control, but normally there is no modem. Instead CCITT V.28 control signals are connected directly to the computer and the maximum signaling rate of 2400 baud is used. The computer can send any ASCII character and therefore has access to receiver functions that cannot be used from the front panel. As an example, continuous back-signaling of the selected meter function (normally the signal strength meter) can be stopped and the value requested by a command from the computer. In this way a much faster response is possible; the computer can set a new frequency and measure the signal level within a few milliseconds.

Operation

All the receiver's operating parameters are displayed on the front panel, either by illuminating the relevant pushbutton controls or using LED displays. The controls are functionally grouped, and the panel is painted in different shades of gray to enhance grouping. The layout follows a logical signal flow from left to right with antenna impedance and attenuator pushbutton controls to the left, and the loudspeaker and loudspeaker on/off pushbuttons to the right.

The frequency display and main tuning knob are in the center, permitting convenient operation with either the left or right hand. Four keys below the frequency display allow the main tuning to be set to fast, medium, slow, or lock. The corresponding characteristics for the tuning knob are: 250 kHz per turn in 100 Hz steps, 25 kHz per turn in 10 Hz steps, 2.5 kHz per turn in 1 Hz steps, and disabled. The frequency can also be entered as a sequence of digits via the keyboard.

Apart from setting the frequency (the normal function), the keyboard can be used for auxiliary function, channel function, and AGC function by pressing the relevant select key. The auxiliary function allows the operator to control equipment external to the receiver from the front panel; this can be very useful with remote control.

Receiver parameters, such as mode of operation and bandwidth, are selected with a single keystroke, permitting fast operation in electronic countermeasures applications. The CR91 has a special programming mode that allows new parameters to be set up (and stored in the channel memory, if required) without changing the receiver setting.

Sweep and Scan Modes

Automatic receiver functions, such as the frequency sweep and channel scanning modes, are provided for surveillance applications. On the CR91, controls for these modes are located on a separate control pad rather than on the front panel. This has several advantages:

- freedom of layout; 10 additional keys would have cluttered and caused confusion
- ergonomics; the keypad is connected by a flexible cord to the front panel and can be placed where it is most convenient, allowing the operator to change position
- the control pad and associated programs are optional and can be left out if the sweep and scan modes are not required.

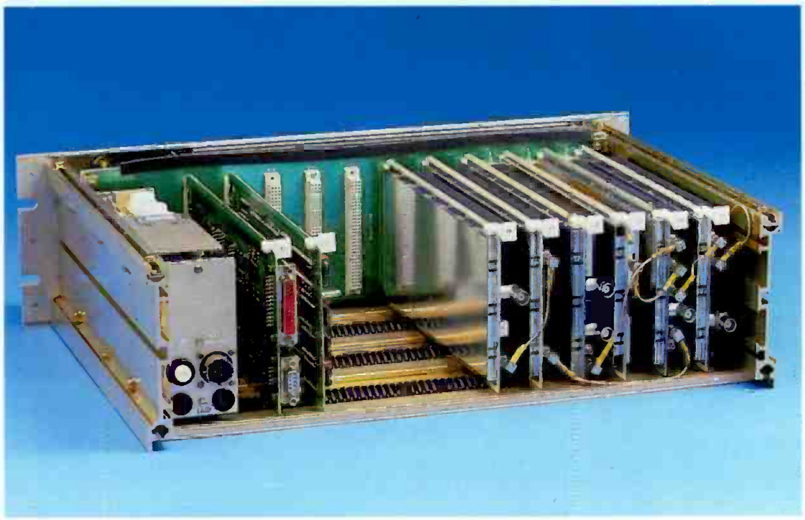
In the frequency sweep mode, the receiver frequency is automatically swept from a preset start frequency to a preset stop frequency. When the stop frequency is reached, the sweep is automatically restarted. The sweep is stopped manually or automatically when a particular carrier level is exceeded. When the SET LOWER pushbutton on the control pad is pressed, the frequency displayed on the receiver front panel is accepted as the lower limit of the sweep. Similarly the upper limit is entered by pressing the SET UPPER key.

The sweep is started by the START SWEEP key on the control pad and can be stopped manually by the STOP key. The speed of the sweep can be adjusted using the SPEED/DWELL-TIME potentiometer.

The following keys are available on the control pad for the channel scan mode:

- SET SCAN: channel number displayed on the front panel is included in the scan sequence
- RESET SCAN: channel number displayed on the front panel is excluded from the scan sequence
- NEXT: receiver is manually stepped to the next channel in the scan sequence

Rear view of the CR91 receiver with the top cover removed.



- PREVIOUS: receiver is manually stepped back one step in the scan sequence
- START SCAN: automatic scan is started.

The dwell time on each channel is set by the SPEED/DWELL-TIME potentiometer and scanning is stopped manually by the STOP pushbutton. Another possibility would be to scan the channels as rapidly as possible until the signal level on a channel exceeds a predetermined value or exceeds the value measured the previous time by a chosen amount. The scan is then stopped, either indefinitely or for the time set by the SPEED/DWELL-TIME potentiometer, permitting the operator to decide what to do.

The exact requirements on the sweep and scan functions (as well as on many other functions) vary from one installation to another. Thus the surveillance receiver must be sufficiently flexible to meet differing specifications. The CR91 control program employs a multitasking operating system and all software is written in a high-level language (PL/M). This allows program modules to be modified and new modules to be included without affecting the overall program structure. Hence, simply by changing programs, operation of the front panel, and of the control pad in particular, can be tailored to a user's requirements.

Mechanical Design

The CR91 receiver has a modular design, as can be seen from the adjacent photograph. The sturdy mechanical construction is based on extruded aluminum profiles. With the exception of the power supply and the front panel, all subunits are printed boards which plug into a motherboard. From left to right the subunits are: power supply, connection board, control board, ISB board (occupying one of the three spare slots in the motherboard), IF/AF board, filter unit, reference board, synthesizer, RF unit, and input filter unit.

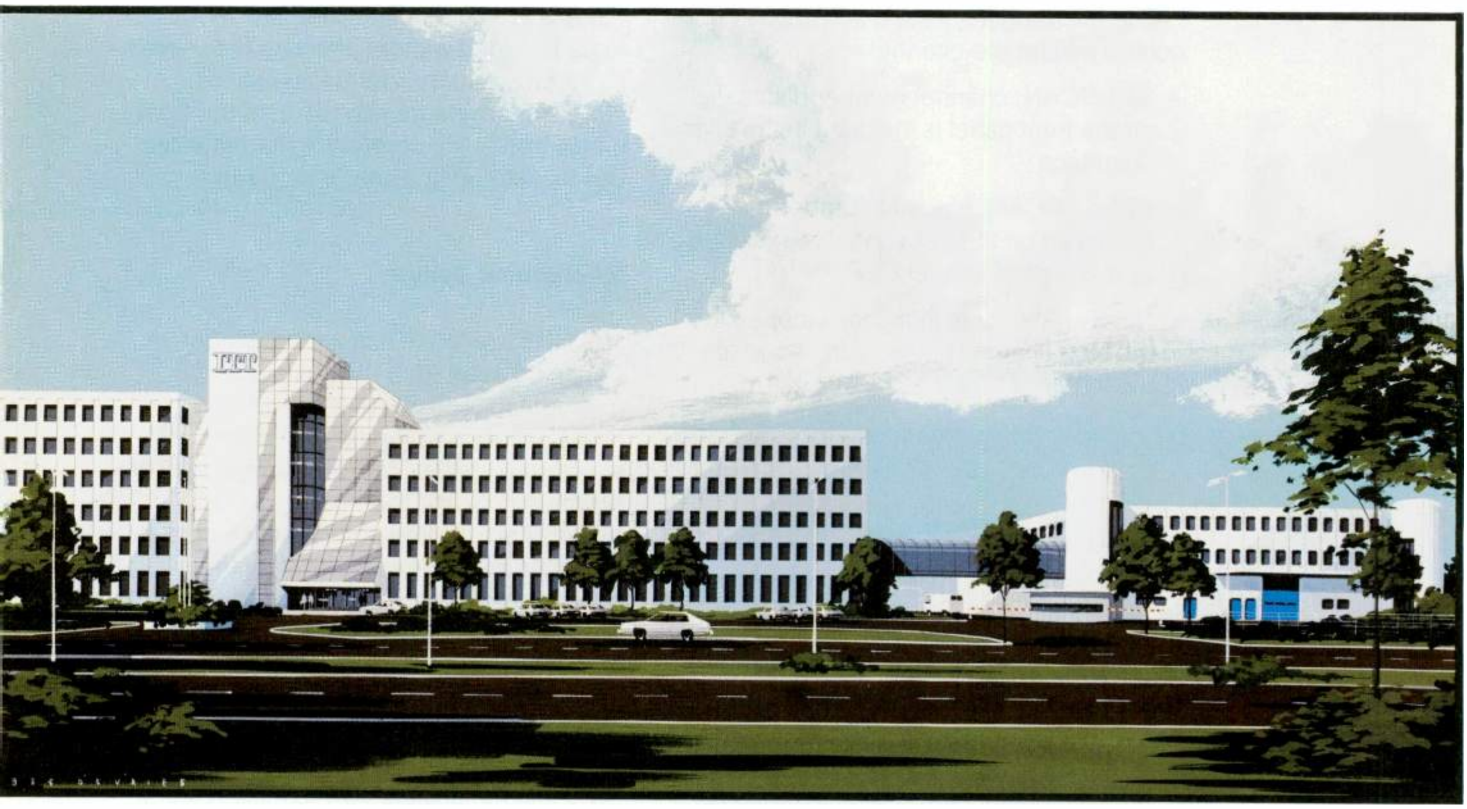
As maintenance is now a major factor in the lifecycle cost of electronic equipment, it is more important than ever to facilitate servicing and repair. On the CR91, all subunits are easily accessible and can be replaced without soldering or realignment.

Conclusions

With the exception of the special surveillance receivers used in tactical electronic countermeasures applications, microprocessor technology is eliminating the traditional distinction between communication and surveillance receivers.

The two logically distinct attributes of a radio receiver – its ability to receive radio signals and its ability to be controlled – can also be made physically distinct and the next generation high frequency receivers (and transmitters for that matter) might very well be 'black boxes'; the operator and the means of control can be located elsewhere.

Robert G. Jonsson was born in Stockholm in 1940. He was awarded a BS degree in 1960, and in the following year he joined SRT as a design engineer. Since then he has participated in the development of three generations of HF communication equipment. Mr Jonsson is at present a senior design engineer involved in the system design of processor controlled HF radio terminals.



Nederlandsche Standard Electric Mij BV

Nederlandsche Standard Electric Mij (NSEM) was founded in 1911 as a branch office of Bell Telephone Manufacturing Company (BTM), Belgium. Thus, when BTM joined the ITT System in 1925, NSEM also became part of ITT. In 1940 NSEM was formed into a limited company, with the shares held by BTM, and in 1967 became a direct subsidiary of ITT. The company is experienced in the development, manufacture, installation, and maintenance of advanced telecommunication and electronics systems and products.

NSEM is structured into five operational divisions, each with responsibility for its own results: Telecom Division, Business Systems, Electronic Security Division, Installation and Service Division, and Manufacturing Division. These divisions are supported by central departments for finance, personnel, information systems, quality control, engineering, and new business development. In all the company employs around 940 people.

ITT Telecom Division

This division has been supplying telephone exchanges for more than 70 years. As a result, a large part of the network in the Netherlands is of ITT manufacture. The entire Netherlands telex network was and is supplied by NSEM, and more recently a digital public data network has been commissioned. As a logical result of the traditional ties with BTM, exchanges are of BTM design and were originally of BTM manufacture. However, over the years local adaptation engineering and local manufacturing have increased.

During the past seven years, NSEM has been developing telecommunications service systems (TSS) to provide older stored program control and electromechanical exchanges with many of the features of the latest generation exchanges. One such product, the Unilink signaling converter, makes it possible for exchanges using different interexchange signaling systems to communicate directly. This is particularly useful in the Netherlands network which includes exchanges from several manufacturers.

In the area of end-user equipment, the product range has been expanded and now embraces subsets, key systems, small PABXs, special purpose terminals (e.g. for the handicapped), and secretarial systems. As an example, the Pentaphone II PABX offers two exchange lines and five extensions, making it suitable for small businesses.

Research and Development

Six percent of NSEM's manpower is dedicated to research and development into the technologies that will be used in the company's next generation of products. Many of the engineers are specialists in state-of-the-art microprocessor technologies and programming.

At present NSEM is closely involved in digital switching. With the Netherlands on the brink of introducing digital exchanges, NSEM is dedicating considerable resources to capturing a substantial share of this market with the ITT 1240 Digital Exchange. In this field, as indeed in all technical areas, there is a constant technology interchange with other ITT companies. In the field of switching systems the major flow is into NSEM, but in the TSS and end-user equipment areas the flow is in both directions.

Future Development

The challenges of the domestic and export markets are being met by NSEM. A major programme is in progress to strengthen the organization in both marketing and engineering. A move to new and larger premises in 1985 illustrates the company's success to date and its confidence in continuing development over the next decades.

A.O. Schaap

A. O. Schaap
Managing Director
Nederlandsche Standard Electric Mij BV
The Hague, Netherlands

Unilink Signaling Converter

Most telephone networks employ a variety of interexchange signaling systems, frequently making it impossible for exchanges to communicate directly. The UNILINK* signaling converter solves this problem by interfacing exchanges that use different signaling systems. It will be of particular benefit during the coming transition to an integrated digital network.

R. J. A. Brood
F. M. Buijs
Nederlandsche Standard Electric Mij BV,
The Hague, Netherlands

Introduction

Although electromechanical and semielectronic exchanges using analog transmission are still going into service in many countries, the latest generation of exchanges integrates digital switching with digital transmission. These new exchanges must interwork with the many types of exchange already in service in national networks.

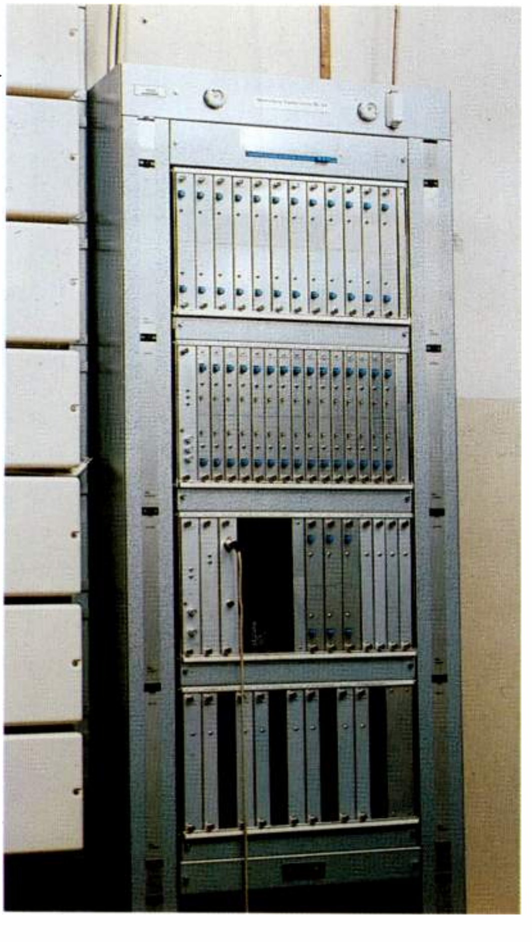
A broad historical view illustrates the variety of interexchange signaling systems that are in use. Older exchanges used much the same type of interexchange signaling as that used by subscribers: line voltage levels for line signaling and pulse trains for selection (register signaling). Manufacturers tended to develop specific line and register signaling methods for their own equipment. This meant that many exchanges could not communicate, or could do so only after modification.

Nowadays the situation is even more complex. Many of these older exchanges are still in use, while at the same time the more sophisticated signaling systems standardized by CCITT are being used for analog and digital SPC (stored program control) telephone exchanges. Digital transmission is now widely used between exchanges. Line signaling is based on channel associated signaling in timeslot *16*, while register signaling can be based on CCITT R 2 MFC (multifrequency code) signaling transmitted over the digital speech channel.

In view of the existence of such a variety of signaling systems, it is not surprising that many exchanges still cannot communicate directly but have to route calls via a hierarchical structure.

National telephone networks currently use some 20 or more combinations of line and register signaling. The design and installation of a new type of exchange that is compatible with all the other exchanges with which it must communicate, is costly and time-consuming in both hardware and program engineering.

* A trademark of ITT System



Front view of a Unilink signaling converter rack.

The efforts of telephone authorities to achieve compatible interexchange signaling in both national and international telephone systems has resulted in a trend towards two combinations of line and register signaling:

- CCITT R2 register signaling combined with a line signaling system that is standardized for a particular country.
- Digital system with line signaling in timeslot *16* combined with digitized MFC register signaling in the subscriber's timeslot.

In an effort to overcome these communication problems, many administrations have sought not only to provide economically more cross-connections between exchanges on different hierarchical levels, but also to provide a convenient and adaptable stepping-stone to the complete modernization of telephone networks based on integrated digital switching and transmission.

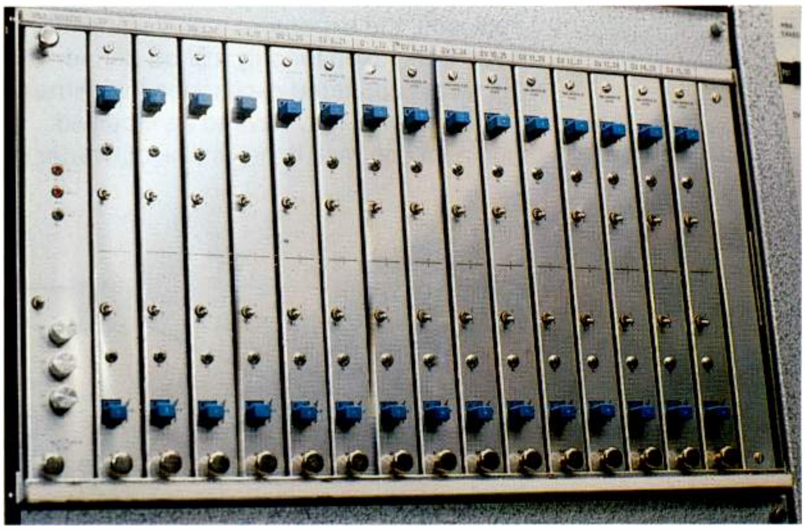
The present situation in the Netherlands is that the telephone network consists of electromechanical and semielectronic exchanges of several different generations from various manufacturers. Siemens F-systems, Philips UR, UV, and PRX, Ericsson AGF, AKE, and AXE, and NSEM Rotary 7D, 7E, and 7EN exchanges coexist in the present network.

Most of these electromechanical systems are installed on a manufacturer-grouped basis, although semielectronic exchanges are used throughout the country. With telephone traffic becoming increasingly dense, the need arises for direct connection in a meshed type network topology. This leads to a greater awareness of the diversity of line and register signaling. The introduction of PCM/TDM (pulse code modulation/ time division multiplex) exchanges into the Netherlands telephone network in the near future together with the increasing number of digital trunks required an advanced and 'future safe' solution.

The Unilink signaling converter was designed to interface the well-used, but older, types of telephone exchanges with modern exchanges using SLS (standard line signaling)/MFC or PCM/MFC signaling. Table 1 shows the main mechanical and electrical characteristics of the Unilink signaling converter. Already the Unilink converter is being used in the Netherlands network to link older electromechanical exchanges to more modern, semielectronic exchanges.

Table 1 – Mechanical and electrical characteristics

Dimensions:	2.70 m high; 0.64 m wide; 0.237 m deep
Weight:	complete rack, without station cables, 224 kg. This includes subracks and boards for 60 lines
Power:	– 48 V DC or – 60 V DC at 200 W
Speech channel attenuation between 300 Hz and 3400 Hz	without hybrid: 0.5 dB with hybrid: 3.5 dB
Echo suppression:	2-wire interface circuit, old system or standard line signaling, 150 to 600 Hz > 10 dB 600 to 3400 Hz > 14 dB 4-wire PCM, 300 to 600 Hz > 15 dB 600 to 2400 Hz > 20 dB
Symmetry:	better than – 60 dB
MFC sender/receiver:	in accordance with CCITT signaling system R2 specification send level: – 8 dBm ± 1 dB receive level: + 2 dBm to – 28 dBm send impedance: 800 ohm (600 ohm available)



Fully equipped subrack of the Unilink signaling converter.

When a Unilink converter is fitted in an exchange, both that exchange and the one with which it communicates 'sees' a mirror image of itself via its connection with the converter.

Unilink Signaling Converter

The basic idea behind the Unilink converter was to provide an add-on unit – a telecommunication service system – to interface existing exchanges at the trunk level. This may be needed where existing exchanges do not provide R2 signaling in combination with a standardized analog or R2 digital line signaling system. In particular, NSEM's Unilink simplifies the introduction of new types of exchange, such as SPC exchanges with analog or digital trunks, or PCM/TDM digital or hybrid exchanges. Moreover, Unilink comes into its own when there is a need for network rationalization by decreasing the number of trunk signaling systems or implementing new routing schemes.

A major benefit of Unilink compared with other methods of interworking between exchanges is that it is 'future safe'. Modification of an existing exchange without Unilink implies removal of the old trunk circuit and installation of a new one, together with modification of the registers. The disadvantage of this 'solution' is that it uses old technology with its high cost of maintenance; in addition it does not prepare the way for going digital. In many cases signaling conversion will only be required for a limited number of years, because older exchanges will be replaced by digital exchanges using modern signaling systems. When the upgraded exchange is replaced, the expensive modification may have been used for only a short period. Similarly, upgraded exchanges that remain in the network may have to be modified again. In contrast, if the Unilink converter is used, the unit can be simply taken away and reused in another exchange as modernization of the telephone network proceeds.

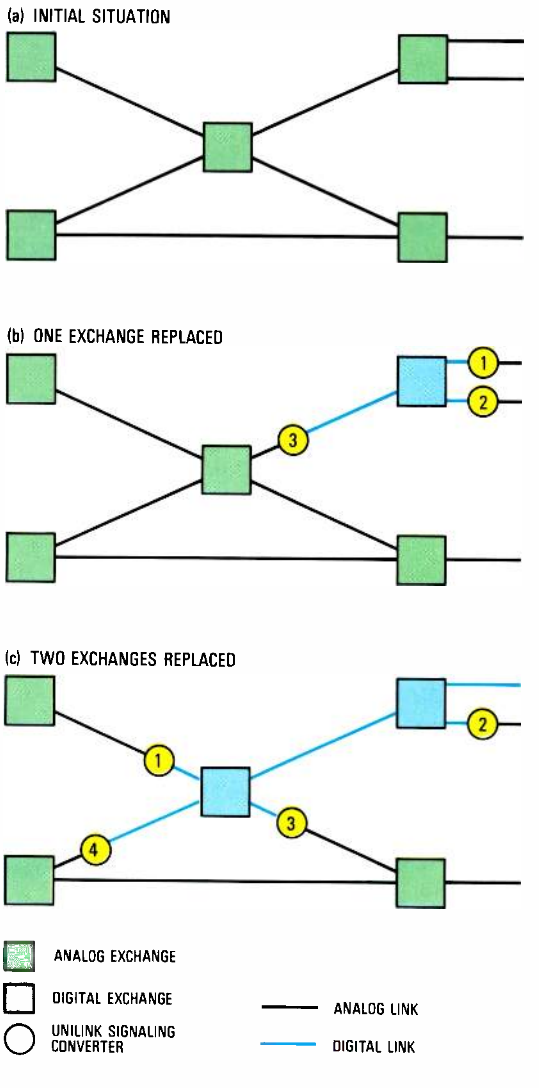


Figure 1
Evolution of a telephone network using the Unilink signaling converter.

Figure 1 illustrates how Unilink converters could be utilized during the evolution to a digital network. When a new digital exchange is introduced (Figure 1b), three Unilink converters (1, 2, and 3) provide the necessary signaling conversion for operation with existing analog exchanges. The introduction of further digital exchanges makes the Unilink converters 1 and 3 redundant, but they can then be reused for other links as shown in Figure 1c. Depending on the network configuration, reuse may not involve any equipment modification. At most, it requires reprogramming of the microprocessor and replacement of one or more printed boards.

Modifying a new exchange without Unilink requires the development of the hardware and programs for a special trunk module. The disadvantage is the need to develop and maintain a variety of hardware and programs for the different types of trunk already in existence. Often these trunks occur in quite small numbers, and may exist for only a short time as a result of network modernization.

NSEM has designed Unilink converters for 7D and 7E Rotary exchanges. They are used in the Netherlands network between Rotary exchanges and Ericsson's AXE for local or toll traffic over analog trunks or with AXE-D's trunk module, and will be used between Rotary exchanges and Philips' PRX-A with analog or digital trunks, and between F-type exchanges and AXE exchanges.

Functional Structure

Figure 2 shows the structure of the standard line signaling converter which consists of four main units: the system dependent interface, the interface for standard line signaling, an MFC sender/receiver unit, and the microprocessor controller. One converter can incorporate interface circuits for up to 30 trunks (the minimum is two, with increases in steps of two trunks). A unit of four MFC senders/receivers can be provided in conjunction with a crosspoint matrix which connects them with the standard line signaling circuits. Each MFC sender/receiver requires one sender board, one receiver board, and one matrix board.

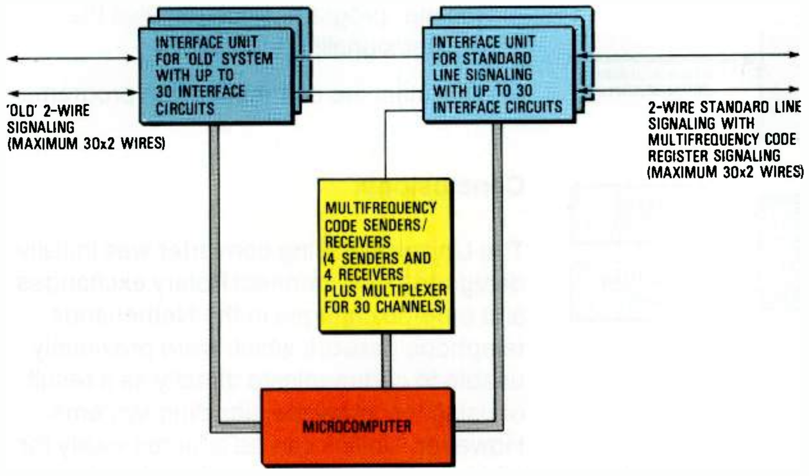


Figure 2
Structure of the Unilink converter for standard line signaling.

In the case of outgoing traffic, the sender in the Unilink converter transmits numerical digits (of the originating exchange) in the higher frequency band; a receiver accepts A and B signals from the terminating exchange in the lower frequency band. A converter used for incoming traffic includes a receiver which accepts numerical digits in the higher frequency band and a sender which transmits A and B signals in the lower frequency band. In common with the old system interface circuits, the standard line signaling units are equipped as two circuits per printed board. Up to 15 boards can be equipped, giving a maximum of 30 circuits per interface unit.

Figure 3 shows the digital converter which consists of a signaling multiplexer and a 2-/4-wire adapter (hybrid). The signaling multiplexer sends and receives line signaling information in timeslot *16*. This multiplexer, constructed on a single board, handles the line signaling for 30 channels. The microcomputer block consists of two boards: a microprocessor unit and input/output circuits. Instead of the standard line signaling interface, a 2 Mbit s^{-1} digital multiplexer is equipped for 30 channels.

In every Unilink converter, the microcomputer unit is equipped with 4 kbytes of random access memory and 16 kbytes of erasable, programmable read-only memory. The digital converter includes a direct memory access facility for the input and output of data via the signaling multiplexer. The microprocessor unit is built on a single board and includes a universal serial asynchronous receiver-transmitter for man-machine communication. The converter is equipped with two power supply units.

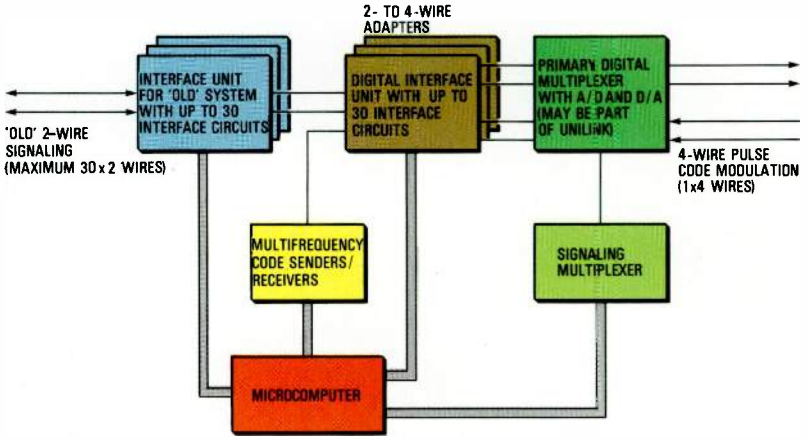


Figure 3
Structure of the Unilink converter for digital exchanges.

Man-Machine Communication

A connector on the front panel of the microprocessor board allows a keyboard terminal to be connected to the microprocessor. Functions provided by the man-machine communication facility are:

- Servicing tool functions; the ability to read the contents of operational memory and, under password protection, to write to that memory to set up specific conditions within the converter and/or the connected telephone lines.
- Hardware alarm functions: a printout giving details of a fault or group of faults can be provided either on demand or when called by a program. Faults include converter or network timeout, or converter faults such as power supply failure or multiplexer failure.

Fault Printout

Faults detected by the register program (i.e. converter or telephone network timeouts) are written into a fault table. Faults relating to the converter power supply, signal multiplexer, or primary digital multiplexer cause a program interrupt and are handled by a separate alarm program. Using the man-machine communication facilities, the *autoprint flag* can be set; this results in a print routine being run each time a fault is written into the fault table. A fault threshold in the fault table is set to a default value which can be altered from a keyboard terminal. When the number of faults reaches the threshold, an indication is given to the alarm routine. A light emitting diode displays the alarm, and details of the fault are printed out if the teleprinter is connected.

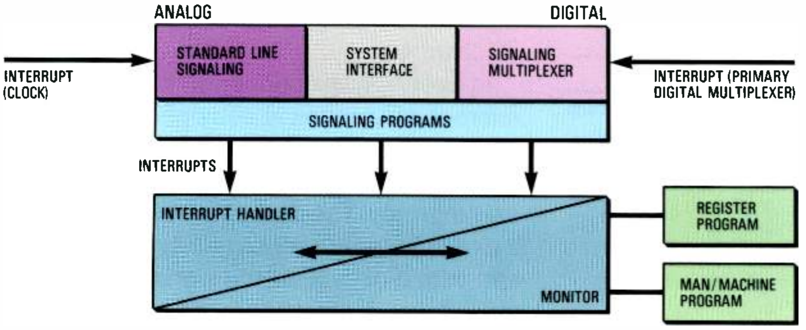


Figure 4
Program structure of the Unilink signaling converter.

Program Modules

Figure 4 shows the program structure of the Unilink converter, which consists of the following modules:

- Monitor program.
- Interrupt handler, which includes a clock program.
- Standard line signaling program, which is initiated every 10 ms by the interrupt handler; it scans and drives the 30 standard line signaling trunk circuits and stores their status in memory.
- System interface program, which controls and communicates with the 30 system interface circuits, in the same way as the standard line signaling program.
- Signaling multiplex program, which communicates with the signaling multiplexer using direct memory access.
- Register program which handles the register signaling functions.
- Man-machine communication program.

Conclusions

The Unilink signaling converter was initially designed to interconnect Rotary exchanges and other exchanges in the Netherlands telephone network which were previously unable to communicate directly as a result of using incompatible signaling systems. However, Unilink can be adapted easily for other networks or to any other circuit signaling system. All that is required is reprogramming of the microprocessor and replacement of one or more printed boards. An interesting application would be during the evolution to a future integrated digital network.

R. J. A. Brood was born in The Hague, Netherlands, in 1944. He attended the Technological College in The Hague where he graduated in 1970. In 1966 he joined NSEM to work in the installation division. Since 1971 he has worked in the engineering department, and since 1980 has been chief engineer for switching and transmission.

F. M. Buijs was born in Geldrop, Netherlands, in 1946. He studied at the University of Technology at Eindhoven, graduating in 1970. After military service, he joined NSEM in 1972. He was seconded for three years to work at BTM, Antwerp, on switching system development. After returning to the Netherlands he worked on various switching and transmission projects, and is now product manager for telecommunication service systems.

Pentaphone II Private Automatic Branch Exchange

Small businesses require private branch exchanges with only a few internal and exchange lines. Equipment for such an environment should be compact, silent, and reliable. Pentaphone II has been designed to meet all these needs.

J. J. C. M. Hoefsloot
R. A. Steinberg
Nederlandsche Standard Electric Mij BV,
The Hague, Netherlands

Introduction

PENTAPHONE II* is a multifunction small private automatic branch exchange which can be equipped with one or two exchange lines serving up to five internal extensions. Two internal lines are also available for an intercom service between the extensions. While Pentaphone II has been designed primarily as a small office system, it is also useful in larger homes where several extensions or two exchange lines are required. The maximum cable length of 200 m allows it to be used in quite large buildings, or even in outbuildings.

Pentaphone II is microprocessor controlled and has a built-in power supply. If the optional DTMF (dual tone multifrequency) receiver is equipped, pushbutton dialing telephones can be used provided that DTMF register signaling is supported by the local exchange.

Any of the wide range of facilities provided by the system can be initiated from any extension. No operator position is required.

* A trademark of ITT System

Pentaphone II equipment being installed in an office. The unit measures only 175 mm × 275 mm × 76 mm.

Design Considerations

Pentaphone II is intended for use in small businesses and at home. In these environments, the following design considerations prevail:

- distances between subsets and the Pentaphone II unit are relatively short
- use in small businesses requires secure internal and external calls
- as the system will be installed in normal rooms in houses and offices, the equipment should be small, reliable, easy to install, and silent in operation
- it should be possible to connect standard and specialist subsets that match the home or office decor.

In order to meet these requirements, an asymmetrical telephone circuit interface was chosen. This cost-effective solution is well suited to the relatively short distances involved, takes up little space on the printed board, and has a low power consumption.

Electronic solid-state crosspoints were used because of their flexibility, small size, and low power consumption. In addition they require fewer microprocessor outputs and operate silently.

Two-wire connection of telephone subsets enables any standard or specialist subset to be connected to the Pentaphone II unit.

An additional requirement for the Netherlands is the facility for connecting DTMF subsets. This is possible if the optional DTMF receiver is equipped.

Facilities

Pentaphone II offers a wide range of facilities for both internal and external calls. These features can be initiated from any extension simply by dialing a 1- or 2-digit code.

The system ensures totally secure internal and external calls. A number of internal tones are used to indicate line conditions; ringing of incoming calls, paging, and general calls can be disabled on an extension to prevent interruption. However, ringing cannot be disabled on all five extensions at the same time, so an incoming call is always heard. Facilities for external calls include call transfer, enquiry to a second extension, the inclusion of a second extension for a conference call, and paging or general calls to all free extensions.

Internal call facilities include conferences using all extensions; two separate internal calls, one of which can be a three-way conference; general call to all extensions; and personal paging, using coded ringing tones, to all extensions.

Hardware Design

As shown in Figure 1, the Pentaphone II consists of six basic building blocks: a microprocessor, a switching matrix of electronic crosspoints, a network circuit, an asymmetric telephone interface, power supply, and the optional DTMF receiver.

All the circuits for one exchange line and five telephone interface circuits are mounted on a single printed board. An additional printed board for a second exchange line can be mounted on top of this. As an option, a DTMF receiver board can also be plugged in the main printed board.

Microprocessor

The microprocessor, an Intel 8050 with a 4 kbyte read-only memory, has 15 inputs and 13 outputs. The inputs are connected to the various building blocks for loop detection and for detection of the ring-trip, watchdog, and DTMF receiver signals, as well as for monitoring equipment status. The outputs are used to operate the relays and electronic crosspoints and to output internal tones.

The program controls the operation of the crosspoints and relays in accordance with the detected signals. A hardware watchdog monitors correct operation of the microprocessor. The watchdog circuit is connected to the crosspoint data output line. If no data is available for a predetermined period, the watchdog automatically resets and restarts the microprocessor.

Electronic Crosspoint Switch

The Pentaphone II PABX uses custom LSI electronic crosspoints originally designed

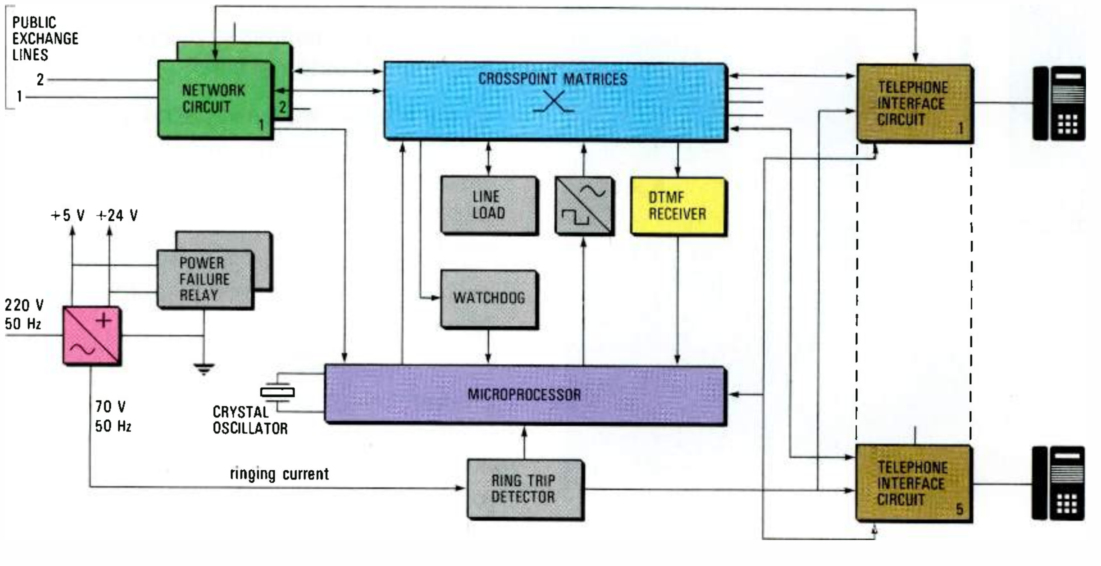


Figure 1
Block schematic of the Pentaphone II small private exchange showing the six main building blocks.

for ITT's UNIMAT* PABX family. Although the choice of electronic crosspoints makes the interface circuits more complex, the programs can be simplified and memory space used efficiently. However, a larger printed board is required. Two electronic crosspoint LSIs are used in the configuration shown in Figure 2.

The status of the crosspoints is controlled by the microprocessor. The use of a shift register in which data is written serially means that only one data output and two control outputs of the microprocessor are used for both switching circuits. The interface circuits for the subsets are connected to the five upper horizontals of the matrix shown in Figure 2. The sixth one is used to hold the connection. The last two are connected to the microprocessor which generates a constant tone. Alternating tones are achieved by switching the crosspoints. All verticals, with the exception of the line for DTMF dialing (TT line) can be used as internal circuit connections. The line relays are switched off if no external call is present. The TT line is connected to the DTMF receiver. Scanning of the telephone lines is achieved by switching on the respective crosspoints under program control. When the microprocessor detects a digit, a check is made to ensure that the signal is sent by the extension that originated the call.

In the event of an enquiry call, the trunk circuit is connected to the hold resistance. The enquiry can then take place via another free vertical. If no verticals are free, the system is in full use. A conference can be set up simply by adding another telephone circuit to the respective vertical. For internal connections there is no restriction on the number of participants, so that all telephone circuits can be connected in conference by means of the crosspoint matrix. However, for external calls the number of internal circuits is limited to two for attenuation reasons. This feature also offers an easy implementation of the group call facility.

Network Circuit

The network circuit consists mainly of an adapter to match the impedance to the public exchange line, a ringing current detector, a power failure relay, a line relay, and a transformer.

In the event of failure of the + 24 V, + 5 V, or 220 V AC supplies, the crosspoint matrices are bridged by a hardware

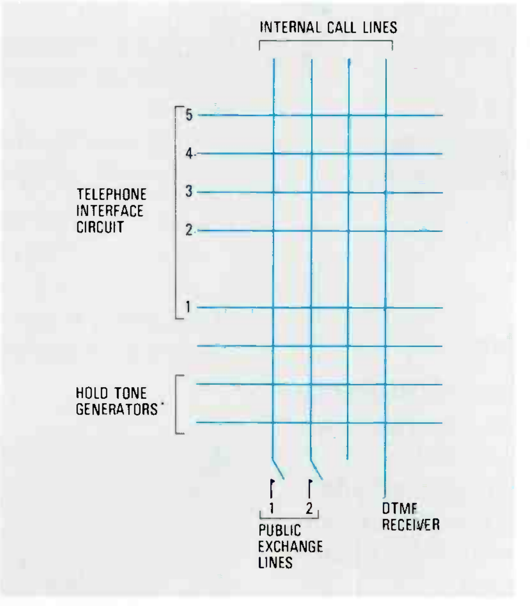


Figure 2
Configuration of the electronic crosspoint matrix used for the Pentaphone II. It consists of two custom designed LSI circuits.

connection. Telephones *1* and *5* are then connected directly to public exchange lines *1* and *2*, respectively. The ringing current detector utilizes an optocoupler to isolate it from the microprocessor. The line relay has two functions:

- switchover function for incoming call detection and voice
- as a pulse relay operating under the control of the microprocessor during call setup.

During outpulsing, a second relay (mask relay) shorts out the speech circuit.

The impedance adaptation circuit acts as a DC loop completion for the public exchange line.

Telephone Interface Circuit

The main functions of the telephone interface circuit are power supply to the telephone subsets, loop detection, and ringing current supply.

A simple asymmetric current source providing 30 mA is sufficient to supply power for the telephones and loop detection. During ringing, the hook status is detected by a ring-trip detector. Lifting the handset completes the DC loop in the telephone subset between the *a* and *b* lines, causing a high output in the loop detection circuit which is monitored by the microprocessor. Dialing pulses are also detected by this loop detector and monitored by the microprocessor.

A relay connects the telephone line to the constant ringing current generated by the power supply. A driver drives the relays under microprocessor control. In this way,

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internal and external ringing currents are provided under program control.

Power Supply
The power supply transformer has one primary voltage and three secondary voltages:

- 12 V for the +5 V
- 28 V for the +24 V
- 70 V for the ringing current.

The transformer is protected by an integral thermal fuse. If any fuse in the system is blown, telephone *1* is automatically connected to public exchange line *1* and telephone *5* to line *2*. The microprocessor and the crosspoint matrix are reset when the power supply is restored.

A ring-trip detector, incorporating a differential amplifier, ensures that ringing stops as soon as the handset is lifted. The 70 V ringing current and a +5 V bias voltage are connected to the detector. When the handset is lifted, the additional load reduces the signal to the amplifier causing a constant low signal to the microprocessor.

DTMF Receiver
Pentaphone II can be equipped easily with the optional DTMF receiver; this receiver is constructed on a small printed board which can be plugged into the main board. The DTMF receiver consists of a hybrid filter separating the high tones from the low tones. These signals are transformed to a binary code in an integrated decoder. When the decoder detects a valid code, the DTMF receiver provides a strobe for the microprocessor which then reads the digit.

The DTMF receiver is connected to the TT line. Scanning is accomplished by closing the respective crosspoints.

Programs for Pentaphone II

All programs are held in the 4 kbyte read-only memory of the microprocessor. A simple initialization procedure is carried out when power is applied to the unit and also when the watchdog function is activated.

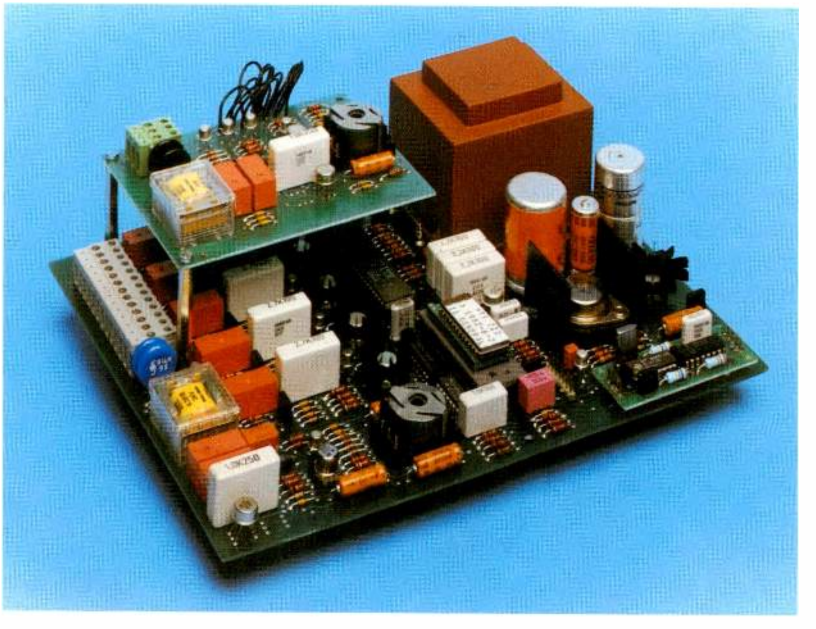
Two programs control the Pentaphone II: a timer interrupt program, which is started at 1.2 ms intervals, and the main program which consists of a number of processes scheduled in a loop.

Main Program Loop
A simple sequential scheduler is used to initiate the various functions in the main program loop. These are peripheral handling and call handling. The start of each main program loop resets the timer interrupt counter.

Peripheral handling is achieved by the following two programs:

- Line circuit handling program which determines the status of the extension line circuit (e.g. on-hook, off-hook, dial pulses). It is run five times in each loop – once for each telephone interface circuit.

Pentaphone II printed board including the optional DTMF receiver.



- DTMF (detector) handling program, when this option is provided. The output of the DTMF receiver is scanned by this program once per loop. When a valid digit is available to be read, the DTMF receiver provides a strobe for the microprocessor, which then reads the digit.

Call handling is carried out by three programs:

- DTMF (call) handling program which handles all activity on the TT line. When the handset of a free extension is lifted, it connects dial tone and the DTMF receiver to the extension in preparation for dialing; when a first digit is dialed, it recognizes that digit, clears the dial tone, and passes control to the public exchange or internal call handling process, as appropriate. Periodically the program connects calls to the DTMF receiver and scans its output to check whether any digit sent from a DTMF extension originated from the caller. In the event of congestion, this program connects the busy tone to an extension.

- Public exchange handling program which deals with all aspects of a call to or from the public exchange; this program handles outgoing digits, sets paths, rings extensions, monitors line status and takes the appropriate action, and handles enquiries and external conference calls. It is run twice, once for each public exchange line. If only one line is equipped, the program detects this when it is run for the second line.
- Internal call handling program which handles dialed digits, connection and disconnection of extensions, and ringing and ringing tones; it also handles enquiries and internal conference calls, and monitors the line status, taking any action that is necessary. This program also runs twice – once for each internal call line.

Timer Interrupt Program

The timer interrupt program, which is initiated at 1.2 ms intervals, varies in length according to the tasks to be executed. For example, at approximately 100 ms intervals, tone and ringing requirements are assessed. At the end of each timer interrupt program run, the main program loop counter is reset. Two pulse driver processes detect incoming calls, seize and release the public exchange lines and send dialing pulses to the public network.

Watchdog Function

Mutual resetting of the main program loop counter and the timer interrupt counter is the basis of the watchdog function. When the main program loop counter has not been reset within 256 loops, the timing system is reset and restarted. When the 1.2 ms timer interrupt counter is not reset within 256 timer intervals, tone generation stops, the watchdog resets and releases the microprocessor, and a system restart is carried out.

Conclusions

The Pentaphone II PABX is a small, versatile unit specifically designed for use in small businesses or even a domestic environment. It is very simple to install, and as it can be wall mounted takes up no desk or floor space. Operation is straightforward, with any of the five extensions able to initiate the various facilities, such as call transfer and conference calls.

J. J. C. M. Hoefsloot was born in Arnhem, the Netherlands, in 1951. He attended the University of Technology at Enschede where he graduated in 1980. The same year he joined NSEM. After a period working on electronic security systems, in 1981 Mr Hoefsloot became involved in the development of end-user equipment.

R. A. Steinberg was born in Oisterwijk, the Netherlands, in 1946. He attended the University of Technology at Eindhoven, graduating in 1970. After military service he joined NSEM in 1972. He was seconded for three years to work at BTM, Antwerp, on switching system development. After his return to the Netherlands, he remained involved in switching and also worked on military projects. In 1977 Mr Steinberg was appointed NSEM product line manager for end-user voice equipment.

Standard Electric Kirk

Standard Electric Kirk was established in 1971 by the merger of Standard Electric A/S (SEA) and Kristian Kirk's Telefonfabrikker A/S (KKT).

SEA was itself founded on 6 February 1931 as a subsidiary of ITT; initially it was primarily involved in importing products from other companies within the ITT system. However, at that time the Danish Government wanted facilities for the development and production of telecommunication equipment to be established in Denmark, and as a result a factory was built in Copenhagen. Four years later, in 1941, the production area had already doubled, and growth continued over the following years.

In addition to being a supplier to the four Danish Telephone Administrations, SEA undertook contract work with the authorities, supplying products as different as telex equipment for the Danish Police and transmitters and studio equipment for the Danish Broadcasting Service. SEA thus became a leading supplier of public exchanges and PABXs – the company's main products when production was transferred to Horsens during the 1970s.

While SEA was primarily operating in and around Copenhagen, KKT had been active in Jutland for decades. KKT was founded in 1892 under the name of the Emil Møllers Telefonfabrikker, after the founder, Emil Møller. Under his direction the company expanded steadily, and in 1917 he bought a factory producing electromechanical equipment in Aarhus. As a result, the company became an important supplier to the Danish Telephone Administrations.

In 1936 the company name was changed to Kristian Kirk's Telefonfabrikker. Under the management of Kristian Kirks and then Gregers Kirks, the company continued to grow, finally becoming a conglomerate of some 30 Danish and foreign companies.

In October 1971, the telephone factory was bought by ITT, and over the next few years all production at SEA was transferred to the new KKT plant at Horsens. In 1976, both companies were amalgamated under the name Standard Electric Kirk A/S.

Today SEK has more than 1 300 employees of which 85% are working at the Horsens plant. About 10% of the workforce is occupied with product development. The concentration of considerable resources in this area has brought SEK into a leading position among telecommunication equipment producers. The DIGITEL* 2000 subset is known and used all over the world. Its unique, modern styling and advanced facilities have made it the standard by which all other subsets are judged.

However, the Digitel 2000 is not the only important SEK product. The company is currently marketing a new digital PABX, the ITT 5300 business communication system, which provides a wide range of user facilities. As it uses digital technology it is geared to the widespread introduction of

* A trademark of ITT System

digital switching and transmission in national telephone networks. At the same time, SEK is pioneering the development of digital subsets. The DT80 has been designed for use with the ITT5300 BCS to enable office communication systems to take advantage of digital technology before digitization of public networks.

Four years ago at Telecom 79 in Geneva, SEK was the only company to have a working digital subset on display. This was the forerunner of the DT80 which in 1980–81 was used in the world's first field trial in which digital transmission was provided all the way from subscriber to subscriber. The trial was undertaken in cooperation with the Jutland Telephone Company; other ITT units supported the trial with exchange equipment.

In cooperation with other ITT units, SEK has produced and maintained the MINIMAT* PABX range which is now being followed by the ITT5300 BCS. In public switching, SEK and BTM have cooperated to deliver some of the first large ITT1240 digital exchanges; SEK now has contracts for the delivery of approximately 100000 equivalent lines of this unique system.

SEK is also cooperating with the Danish computer company RC-Computer on the development and launch of DATABOCS – an office automation system based on the ITT5300 BCS combined with the RC3904 IAPX-based 16-bit microcomputer. This office automation system uses the DT80 digital subset to provide access to integrated voice and data facilities, including gateway functions to local area networks, packet switching networks, etc.

SEK has a firm foundation based on successful products such as the Digitel 2000 subset series and the ITT5300 BCS. These ensure that the company is in an excellent position to take advantage of the future merging of traditional subsets with tomorrow's intelligent terminals.

K. Jakobsen
Managing Director
Standard Electric Kirk A/S
Horsens, Denmark

Digital Subset for the ITT 5300 Business Communication System

The DT 80 family of digital subsets can be used with the ITT 5300 business communication system to expand the existing voice facilities. In particular, the combined system supports data switching of low to medium speed data from data terminals connected to the DT 80 subset.

D. Andersen
E. Stridbaek
Standard Electric Kirk A/S, Horsens, Denmark

Introduction

During 1981 and 1982, a 12-month field trial was carried out in Horsens, Denmark, involving the use of 50 digital subsets spread throughout the city's local network. The equipment was designed and produced jointly by SEK (Standard Electric Kirk), the Jutland Telephone Company, and Standard Telecommunication Laboratories. The main objectives were to prove that digital subsets could operate using existing wiring and to determine how useful subscribers found the new facilities. After a successful field trial, which gave very promising results, SEK decided to use the technology to develop digital subsets for use with modern digital PABXs (private automatic branch exchanges) operating in an office environment.

DT 80 digital subset.

DT 80 Digital Subset

The DT80 digital subset is the result of further development of the digital subset used in the Horsens field trial. The reasoning behind the development of a new generation of digital subsets was to use them in conjunction with SEK's ITT 5300 BCS (business communication system) – a new generation digital PABX – in the local network to take advantage of the ability of digital transmission to carry voice and data simultaneously.

Transmission Format and Protocol

The transmission format used for the DT80 subset is an 80 kbit s^{-1} AMI (alternate mark inversion) code, which is divided into an 8 kbit s^{-1} channel for synchronization (4 kbit s^{-1}) and signaling (4 kbit s^{-1}), an 8 kbit s^{-1} channel for data traffic, and a 64 kbit s^{-1} channel for digital speech. Synchronization and signaling information is transmitted as 8-bit command words; error correction is achieved by repeating important signals. The 8 kbit s^{-1} data channel is transparent, allowing the maximum data rate of 8 kbit s^{-1} to be used.

Hardware

All the circuits for the DT80 subset are mounted on a single printed board. The DT80 line interface utilizes low power operational amplifiers, enabling it to be powered by a 40 mA current from the ITT 5300. The subset is controlled by a CMOS microprocessor with a 2 K CMOS EPROM and 256 × 4 bits of CMOS RAM

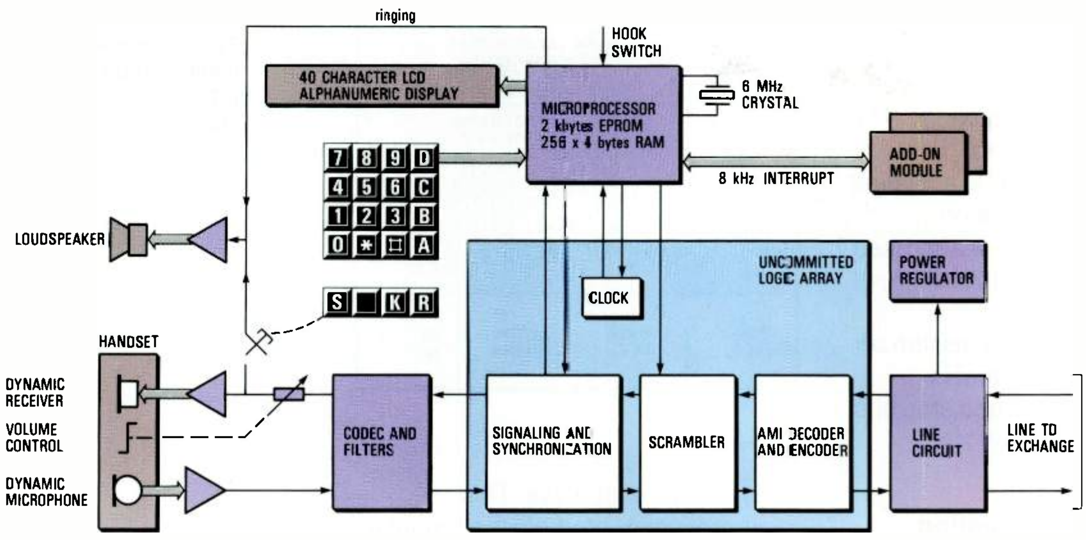


Figure 1
Block schematic of the DT80 digital subset developed by SEK.

(random access memory). A CMOS single chip codec and filter is used for the digital voice channel. Figure 1 is a block diagram of the DT80 digital subset.

The discrete CMOS logic is included in a new ULA (uncommitted logic array) circuit with 560 gates which has been developed by SEK. The ULA circuit includes an AMI coder and decoder, pseudorandom scrambler generator, digital phase-lock loop, and 80 kHz and 8 kHz clock generators; the quartz clock of the microprocessor is used as a master clock in the DT80 subset. The ULA circuit communicates with the microprocessor over the synchronization and signaling channels. This circuit also includes an interface for the 8 kbit s^{-1} data channel.

The DT80 printed board includes two interfaces for optional boards: one for an LCD (liquid crystal display) of 2 × 20 ASCII characters and one for expansion of the DT80 subset by add-on printed boards.

DT80 Program

The DT80 is programmed in assembler. The program is designed with two interrupt levels. When the DT80 has received 10 bits from the 80 kbit s^{-1} line, the microprocessor is interrupted, and the signaling and synchronization bit is read; the 8-bit speech sample is forwarded to the codec and the data bit is routed to the add-on module. After eight such interrupts, the extended interrupt program is called. This program transfers the collected signaling bits to the main program, checks the synchronization pattern, and then executes the keypad scanning, display refresh, and communication routines for the add-on modules.

The main program is designed as an interpreter that controls the execution of subset functions. As the main program is interrupted every 125 μs, only a few main program statements are executed between two interrupts. The microprocessor provides the necessary fast interrupt response and return.

Extension of the DT80 Facilities

As mentioned previously, the facilities of the DT80 can be extended by add-on printed boards. The multifunction terminal module is used to extend the DT80 keyblock by up to sixteen keys, eight of which are provided with LED (light emitting diode) indicators. This module is powered from the ITT 5300 BCS over the same wire as for the DT80 subset.

The terminal access module adds a CCITT V.28/V.24 data interface to the DT80. This module is constructed on a daughter board which is built into the DT80 standard housing. The terminal access module may be programmed from the controlling exchange for use with different data transmission rates: asynchronous 300 to 9600 bit s^{-1} and synchronous 300 to 4800 bit s^{-1}. This module communicates with the ITT 5300 via the DT80 signaling channel; when the user wishes to set up a V.24 data call, the keypad is used to select the number of the called data port. During call setup, the ITT 5300 BCS initializes the terminal access unit to the required format.

The terminal access unit uses the 8 kbit s^{-1} channel for data communication.

Synchronization of this channel is achieved by the two terminal access units communicating over the path that has been set up by the exchange. Character synchronization between the two connected terminal access units is achieved using a 4-character synchronization sequence. The same sequence is also used to convert the 8 kbit s^{-1} synchronous channel to different data rates. Asynchronous 9600 bit s^{-1} transmission on the 8 kbit s^{-1} synchronous channel is achieved by removing the start and stop bits from the asynchronous channel.

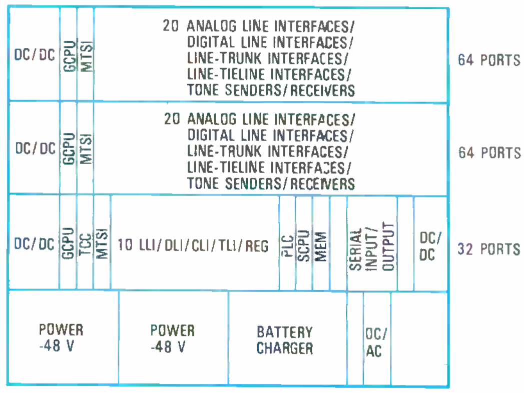


Figure 2
Typical hardware configuration of the ITT 5300 BCS with 160 ports.

ITT 5300 Business Communication System

The ITT 5300 is a digital PABX with up to 352 ports, which may be configured to provide a very flexible combination of extensions, trunks, tielines, and tone senders/ receivers.

The main units of the ITT 5300 are three types of shelf which are mounted one on top of the other as individual units. The power shelf incorporates the −48 V power supply, battery charger, and ringing current generator for analog extensions (see Figure 2).

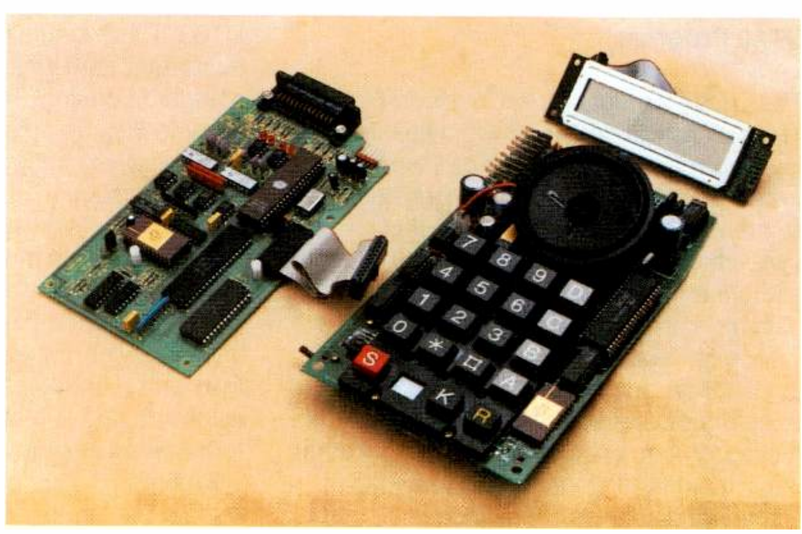
Above this is the system shelf which consists of two parts: the system controller and the first group controller. The system controller performs resource management and call handling, and maintains customer data (a battery backup is provided). It communicates with the first group controller and other shelves using a serial format. Connected directly to the system controller are up to four operator sets and the man-machine communication unit which is used for maintenance operations, either locally at the exchange or at a remote center via a modem connection. Remote maintenance is also possible using the low cost DTMF (dual tone multifrequency) signaling, but the response is slower.

The system microprocessor and memory are located on two printed boards designated SCPU and MEM, respectively (see Figure 2). Other shelves and operator sets are connected to the three SIO (serial input/output) boards which together provide 12 connections. Voltages for the electronic equipment are generated on each shelf using a DC/DC converter which is powered from the −48 V main supply. The same type of converter is used on all shelves.

At the left side of the system shelf is the GCPU (first group controller), which is also used on higher shelves. This microprocessor controls the modular timeslot interchanger; the tones, clock, and conference board; the paging and loudspeaker control board; and 10 line positions. The tones, clock, and conference unit stores the digital tone patterns in EPROM; these patterns are transmitted on fixed timeslots in the system so that all ports can read a specific tone and convert the digital pattern to a corresponding analog signal. The conference part can handle six 3-party conferences and one 8-party conference simultaneously; the conference unit is based on the "instant speaker" principle. A small part of this board is used for the accurate 2.048 MHz clock and a frame clock generator.

The remaining 10 positions in the system shelf can be used for any combination of four line extension interfaces (analog LLI and digital line interfaces), four tone sender/ receivers (REG), 2-line tieline interfaces (TLI), or 2-line trunk interfaces (CLI). Any combination from 20 trunks/tielines and zero extensions to zero trunks/tielines and 32 extensions can be configured. If tone

View of the DT 80 digital subset showing the major components.



sender/receivers are required, they can replace either four or two extensions.

The 10 positions and their interface boards are controlled by the microprocessor bus. Ports are connected to the switch via an ordinary 32-channel 2.048 MHz digital system in which all 32 timeslots are used for voice as control and synchronization are generated centrally by the microprocessors and the tones, clock, and conference board.

Further shelves are of a type known as group shelves; up to five such shelves may be equipped in one ITT 5300. In addition to the DC/DC converter, this shelf contains a group controller, a modular timeslot interchanger, and two 10-board positions which can be configured in exactly the same way as the 10 positions in the system shelf. Thus a fully equipped exchange provides either 240 trunks/tielines or 352 extensions, or any variant within these limits. A commonly used configuration provides 312 extensions, 32 trunks, and eight tone sender/receivers.

The control structure is based on a central microprocessor and distributed microprocessor controllers for each shelf of 64 ports. The controller performs the basic primitive actions, such as scanning and timing at the port level, when analog interfaces are equipped.

In the case of digital lines, the distributed microprocessor controls the on-board single chip processor, which is in charge of subset synchronization and signaling.

Digital Line Interface

When the ITT 5300 business communication system is used with the DT80 subset, it must be equipped with a digital line interface board. One such board can interface up to four DT80 subsets. The function of this printed board is to interface the 2.048 Mbit s^{-1} digital bus of the ITT 5300 to the 80 kbit s^{-1} line speed of the DT80 subsets.

The digital line interface incorporates a multiplexer which merges the speech samples with the signaling and synchronization bits generated by the onboard processor. The interface to the DT80 uses the same ULA circuit as in the subset. The digital line interface includes the ULA circuit, AMI encoder and decoder, pseudorandom scrambler generator, digital phase lock to the receiver clock, and variable delay which is used during synchronization hunting.

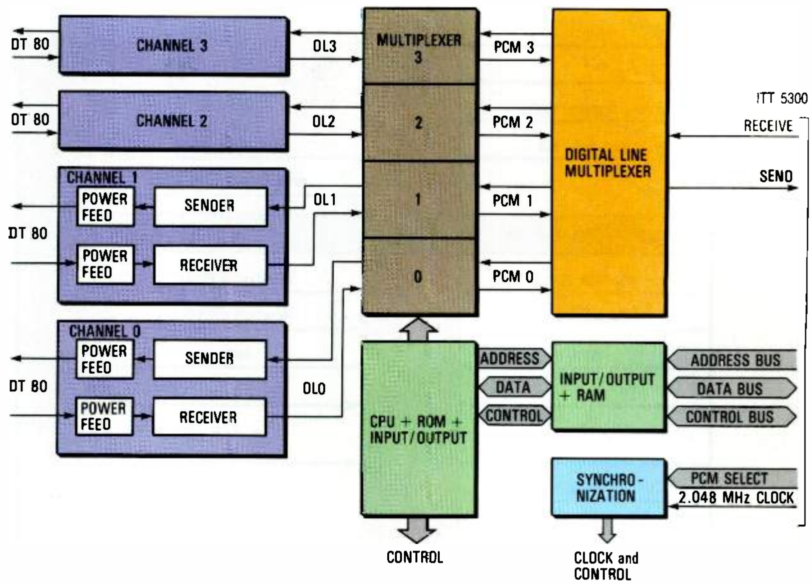


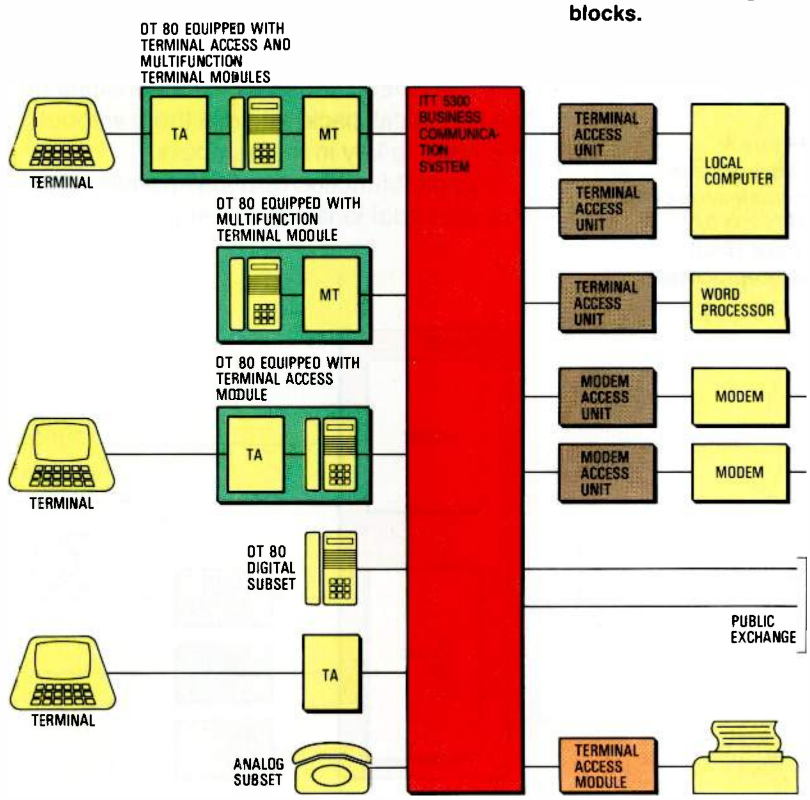
Figure 3
Block schematic of the ITT 5300 digital line interface.

The digital line interface board includes two current feeding circuits for each DT80 subset; one for the DT80 itself and one for the optional multifunction terminal module.

Digital PABX Voice Features

The initial version of the ITT 5300 is equipped with analog interfaces for conventional subsets and trunks. By adding the digital line interface and the DT80 subset building blocks, an extended range

Figure 4
The ITT 5300 BCS with DT80 building blocks.



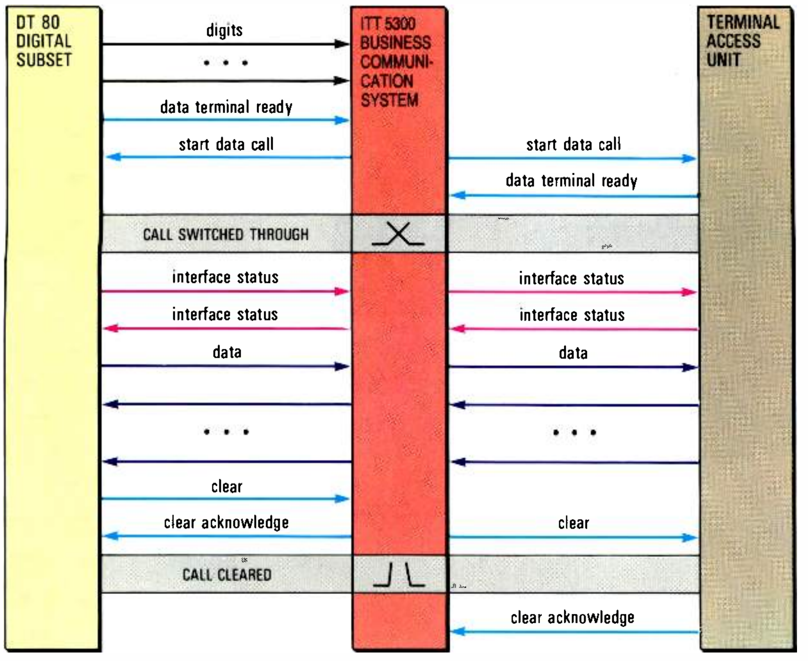


Figure 5
Typical signaling sequence for data call setup and clearing.

of facilities is available for voice and data switching.

Figure 4 summarizes the interface possibilities with variants of the DT80 subset based on the optional modules already described.

The DT80 offers conventional subset facilities, but with an improved quality of transmission. In addition, the DT80 features built-in abbreviated dialing, last number redial, and an LCD display to show the called or calling number and call status. If a DT80 subset is called and there is no answer, the calling extension number may be stored for recall by the absent user on his return. The user can then choose which extensions he wishes to call back, and dial them without the need to key in the numbers.

The multifunction terminal module offers the user additional keys to support existing voice facilities, with one function per button. Also, LED indicators may be provided to show the status of lines or extensions in the group (i.e. busy, calling, holding, or idle).

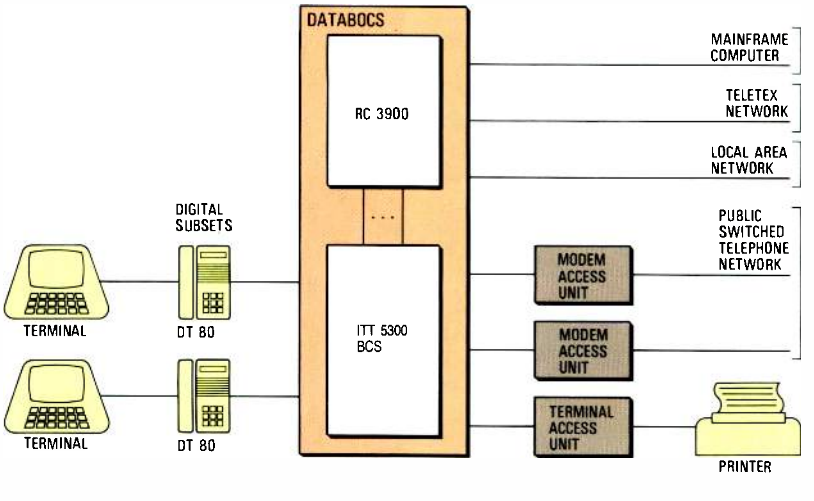


Figure 6
Databocs – a combination of the ITT 5300 BCS and a 16-bit microprocessor.

Data Switching Facilities

The DT80 line signaling structure reserves 8 kbit s^{-1} for data. In order to offer data switching, a terminal access module must be added to the DT80. This printed board provides a V.24/V.28 interface for the connection of low to medium speed data terminals. Two modules can be combined to offer a stand-alone data plug based on the same blocks as the DT80, but without voice facilities. The two modules comprise a terminal access unit for the connection of terminals, or a modem access unit for the connection of conventional analog modems, the required version being selected by straps.

Data calls can be set up in two ways; either directly for a predefined data port or under user control via the subset keypad or using one of the additional keys on the multifunction terminal module.

In the case of direct call setup, the signal *connect dataset to line* (CCITT no 108) is used to initiate a data call and clear it.

If the data port is defined by the user, he must key in the called data port number as the interface does not directly support call setup procedures. During manual call setup, the CCITT no 108 signal is used as a *data terminal ready* indication, and call setup is only accepted if both data ports signal the true condition for this V.24 interface signal. A typical call setup sequence is shown in Figure 5, which indicates the main control flow between the DT80, the ITT 5300 BCS and, for example, a called terminal access unit.

To support user-friendly procedures for data calls, the system supports user-selectable bit rates, multiple data ports with the same number, and redirection of data calls during maintenance.

Several types of barring are also supported, with the facility for preventing some data ports calling data ports outside their own group. The facility for having multiple data ports with the same number allows the use of common lines to local computer equipment or modems to be shared among several users, as the PABX automatically selects a free port in the group when the common number is called.

If a data port or pool is busy, the call waits automatically, provided the called port supports this facility.

As no tone signals are supported during simultaneous operation with voice and data, the user is guided by text output on the DT80 display.

The total system supports both asynchronous and synchronous data transmission at the conventional bit rates of up to 9600 bit s^{-1} used on existing data terminal equipment. If the simultaneous transmission of voice and data is not required, the transmission rate can be as high as 64 kbit s^{-1}.

Databocs

In order to extend the data transmission facilities of the ITT 5300 BCS, SEK and Regnecentralen have started to integrate it with a 16-bit microprocessor. This multi-user microprocessor was specially designed for the growing market in office communication and data transmission equipment. The system supports connections to teletex, well-known mainframe computers, and local area networks (Figure 6).

Using this configuration, the ITT 5300 BCS acts as a terminal concentrator to the microprocessor, based on the pooling principle, offering the user a low cost connection for standard V.24 data terminals to word processing facilities, the Teletex network (including telex), the local area network, and mailbox facilities. At the same time it retains the user's connection to existing local computers and modems via the PABX data switch.

The microprocessor may be a stand-alone unit, but in the PABX version it occupies a conventional PABX shelf, enabling it to be powered from the PABX, giving the option of an uninterrupted power supply if the PABX has this facility.

Conclusions

Based on the results of a successful field trial in the town of Horsens, SEK has used the DT 80 digital subset and the ITT 5300 BCS to develop a flexible integrated voice and data facility for use in the office and local area, where digital subset technology seems likely to penetrate faster than in the public network. In this way the experience gained from the field trial will be expanded by the introduction of new products in the field of digital networks, which will evolve rapidly in the eighties.

Databocs shelf stack, showing the PABX shelves and microprocessor shelf.

D. Andersen was born in Aarhus, Denmark, in 1950. He graduated with an MSc in electrical engineering at the Technical University of Denmark, Copenhagen, in 1975. He then worked on operating system software at Regnecentralen, before joining SEK in 1979 where he is now project leader for the ITT 5300 BCS.

E. Stridbaek was born in Vejle, Denmark, in 1948. He graduated with a BSc in electrical engineering from the University of Aalborg, and in 1980 received a degree in organization theory from the Highschool of Commerce. In 1978 he joined SEK where he was responsible for the field trial in Horsens. Mr Stridbaek is currently project leader for the DT 80 subset and operators' desks for the Minimat 64 and ITT 5300 BCS.

ITT Austria GmbH

With a century of experience behind the company, ITT Austria is one of the longest established units in the ITT System. Founded in 1884, with one of its branches formed even further back, the Austrian company started building large telephone exchanges in Vienna as early as 1889. Today, this aspect of production retains a predominant position in ITT Austria's business, with almost equal emphasis being accorded to the development and installation of railway signaling and remote control equipment. Two of the principal customers are, accordingly, the Austrian Postal and Telegraphic Administration and the Austrian Federal Railways system.

Through its affiliation with ITT and its forerunner Western Electric, since 1905 ITT Austria has benefited directly from international technology. For its part, ITT Austria undertakes development work for other ITT affiliates in Europe.

Facilities

ITT Austria has its main office and factory in Vienna on the left bank of the River Danube overlooked by the vine-clad slopes above Grinzing. It is directly accessible from the main motorway network which also provides a convenient link with Vienna International Airport. A second factory, located at Eggenburg, 50 miles from Vienna, has virtually identical manufacturing facilities, offering considerable production flexibility, depending on market needs and the current availability of labor. Branch offices serve all of Austria's nine federal provinces.

In all, ITT Austria employs a staff of around 2750, including 450 engineers and technicians. Research and development plays a major role in the company's activities with, for example, 175 million Austrian Schillings being allocated to R&D in 1982, an increase of almost 48% over the previous year and representing more than 8% of turnover. During 1983, the proportion is expected to rise to 9%.

Switching Equipment

Telephone switching equipment, by far the ITT Corporation's largest manufacturing product line, is also dominant in Austria. Crossbar and METACONTA* systems are now being superseded by digital exchanges. The ITT 0802 microcomputer, which is designed and manufactured by ITT Austria, is well suited to a variety of applications, including large public exchanges and small PABXs. The ITT 0802 is essentially a control device which can be used either to upgrade existing electromechanical telephone exchanges or as the 'heart' of a PABX. It is already widely used as a standard component within ITT Europe (e.g. in Switzerland, Finland, and Denmark). This equipment is conceived for a much longer lifecycle than, for instance, a personal computer. Whereas the latter is likely to become obsolete within a few years, the control device of a telephone exchange must remain operational for a much longer period, and thus requires much higher quality standards.

In the field of PABXs, ITT Austria has recently launched the sophisticated ITT 5200 business communication system, also known as the Amanda. This is geared to both European and overseas markets.

Even the most sophisticated public telephone switching systems still require operators to perform certain services. The fully digital System 12 operator position, which can be adapted to any type of telephone exchange, is designed to increase operator efficiency and improve operator comfort. This digital operator

* A trademark of ITT System

position, which was developed and is being produced by ITT Austria, is to become standard throughout ITT.

Railway Signaling

The ITT 0802 microcomputer is also used in railway signaling systems, another important aspect of ITT Austria's activities. The company enjoys technological leadership in computer-based electronic systems for railway signaling and control used by the Austrian Federal Railways. Major projects have included equipment for the shunting yards at Wolfurt in Vorarlberg, and the stations at Heiligenstadt in Vienna and Kufstein in Tyrol. Further projects include computer-based train-routing systems for the Taurern line and for the high-speed 'S'metropolitan line in Vienna.

Export Markets

Altogether 12% of ITT Austria's production is exported, including worldwide sales of the UNIMAT* 4010 PABX. In addition, components, microcomputer hardware and programs, and support software systems are exported. Principal export markets are the Federal Republic of Germany, the Arab countries of Africa and the Middle East (especially Saudi Arabia), Iraq (ITT Austria has a branch office in Baghdad), and Iran.

Future Prospects

ITT Austria's management is confident that the future continues to be assured by its policy of employing skilled engineers and technicians on high technology projects and products. Every opportunity is taken to utilize these skills to develop products that are of high quality and relevant to the rapidly evolving needs of users.

R. T. Stasek
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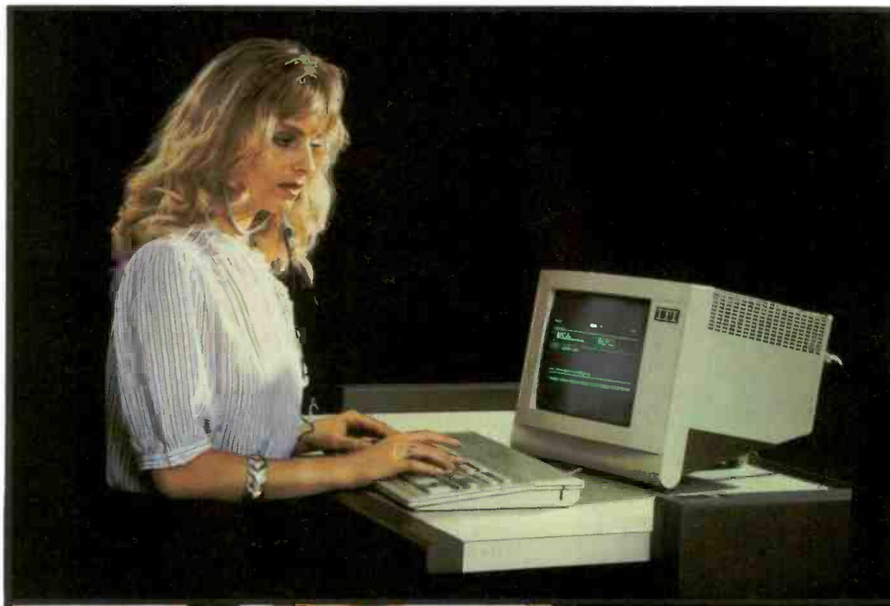
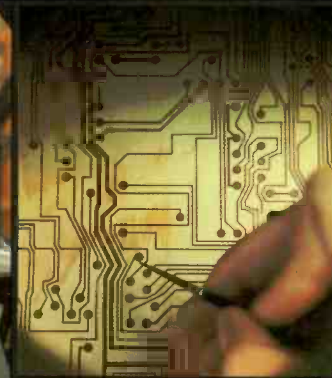




Videopult System for the Management of Railway Stations

The Videopult system offers considerable improvements in the management of railway station operations. In particular, the use of a light pen to input commands speeds up operations, while ensuring that the operator retains overall control.

K. Lukaschek
ITT Austria, Vienna

Introduction

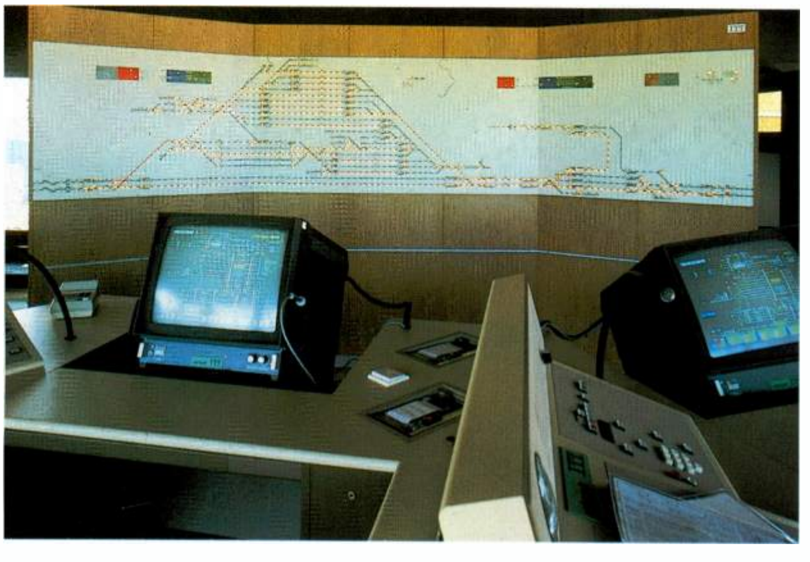
The Wolfurt freight station in Vorarlberg serves as the central reshipment point between road and rail traffic for the region. This new station handles the rolling stock, and controls 92 points and 227 main and shunting signals. The central signaling unit necessary for these operations was supplied by ITT Austria. The Videopult system is the man-machine interface for this signaling unit.

In addition to the conventional panoramic display, a Videopult system is available to the traffic superintendent and chief signalman. This system was developed by ITT Austria and the ÖBB (Austrian Railways) as an alternative to the conventional command keyboard console. The operation of a large route interlocking system is not in general controlled by pushbuttons on the panoramic display but through a command keyboard at the traffic superintendent's workstation. Each pushbutton on the panoramic display is allocated a three-digit number. These numbers are selected, as required, by the traffic superintendent and entered into the route interlocking system via function keys on the command keyboard.

The Videopult system was originally installed for field trials at the Wolfurt station, the aim being to test the concept of an integrated facility for controlling train movements and shunting operations. The man-machine interface between the operator and the route interlocking system was made the cornerstone of the development.

It was important for the indication and operation equipment to be independent of both the technology and make of the equipment that it was to control. To achieve this objective, the latest communication technologies and ergonomic principles were employed during design. In addition, the ÖBB set out clear cost limits on the project.

Fully adjustable operator position for the Videopult system.



Videopult System

The interface between operator and equipment was at the heart of the system design. Not only the classic indication and operating functions for the route interlocking facilities, analogous to the conventional command keyboard, were to be taken into account; operation of the system in a railway station was to be considered as a general task which required an integrated management system. As the essential management tasks are based around the workstations of the traffic superintendent and chief signalman, priority was given to assisting them to perform their jobs.

Management should be assisted by the Videopult system, but retain its role as decision maker, particularly for complex decisions. At the same time, human error can largely be prevented by the system. This required the design of an equipment layout for the indication and operation functions that is clear, logical, and provides the necessary operational flexibility.

A cathode ray tube was the natural choice as a display for the indication function as it has proved valuable in train control centers for many years. A solution to the problem of entering instructions was, however, more difficult to achieve. From the outset it was clear that integration of the operation of all technical facilities would only be acceptable to operating personnel if the system allowed associative interactive operation. For this reason, after careful analysis a light pen was chosen for inputting instructions on the "indicating monitor".

Semi-Graphic Picture System

Once a cathode ray tube had been chosen, the decision to use a color screen followed because of the complexity of the tasks to be performed. A color display makes it easier for the operator rapidly to assimilate a wide variety of information.

Another difficult question was which of the proven systems available should be selected. Not only was the cost of equipment important, but also the cost of symbol and picture creation. After extensive analysis, including consideration of the chosen method of inputting instructions, the process video system PVS 1050 (1100) was chosen. This *semi-graphic* system met all requirements with its 2048 screen fields and 256 freely definable symbols (per picture). The simple, rapid creation and programming of process pictures by a light pen had a major influence on the choice of system.

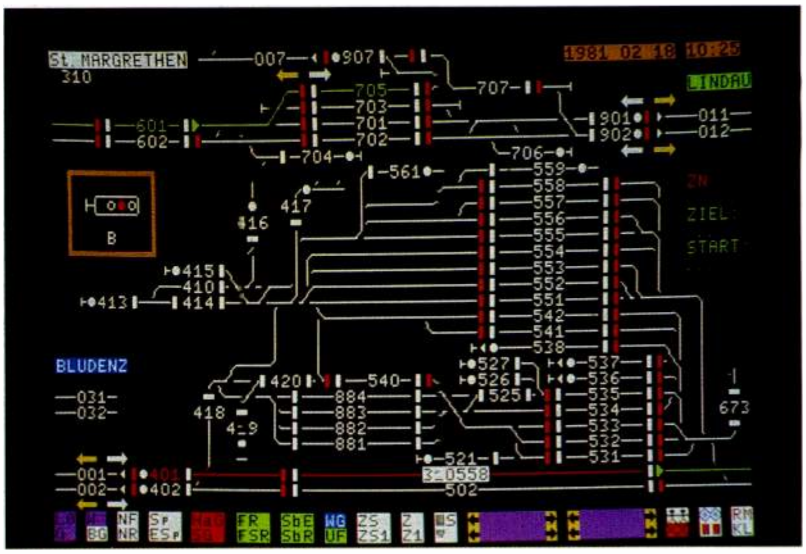
Symbols

Special attention was given to the design of symbols for illustrating the wide range of rail operations. Basically the aim was to use a representation based on *symbolic association*; this means the choice of simple symbols that are familiar to and easily interpreted by a wide spectrum of the population. As an example, a main signal in the stop position is symbolized by a red diagonal bar across the track. When the signal condition changes to "go", a green arrow replaces the red bar.

The PVS 1050 (1100) system can show one symbol with one symbol color and one background color for each screen field.

Human factors were used when considering the possible combinations of background color with symbol color; this made an important contribution to achieving the aim of symbolic association for the operating condition symbols.

In addition to the approved display of the track diagram, an enlarged display was provided that could show any track element (e.g. main signal B) requested by the operator, including detailed information in the familiar form of a panoramic display. In practice this display has not proved necessary.



Videopult system display of the railway system track diagram provided for the traffic superintendent.

Input of Commands using a Light Pen

In the Videopult system, commands are input using a light pen. The light pen is placed on the desired symbol (e.g. signal, points) on the surface of the cathode ray tube and gently pressed. Pressing the light pen operates a contact which switches on the photodiode in the pen, thereby generating the corresponding signaling command for the master computer.

The system confirms input of the command by flashing the appropriate symbol(s). Commands from the master computer are only passed to the signal box when the appropriate enabling key at the lower edge of the picture is pressed. These enabling keys correspond to the group buttons of the panoramic display; in this way the proven two or three button operation method of the panoramic display has been retained.

Command input using a light pen is quicker than using a keyboard. As the field trial has shown, an alphanumeric keyboard is not necessary for inputting control functions in a railway management system.

Input of Instructions by Numeric Keyboard
The numeric keyboard with a six-digit input control corresponds to that of the conventional number signaling console. This keyboard was provided in the field trial to ease operational changeover to the Videopult system and to provide a backup in the event of a monitor failure.

Information Processing
At the start of the development, processing of operational information was considered to be of lower priority than the display and control functions.

Initially, existing data in the master computer was used to check all inputs (plausibility check). Since all functions of the railway failsafe equipment are available in the master computer, plausibility checking can be performed before signaling information is sent from the master computer to the route interlocking equipment. If the input is incorrect, the system informs the operator via a screen message.

In addition to plausibility checks, auxiliary functions, such as train number registration and train routing (derived from the first digit of the train number or the whole train number), have been integrated in the system.

Recording of management data is also part of information processing. During the field trial at Wolfurt station, special actions by the operator which could lead to a dangerous situation by overriding the failsafe equipment were printed out; all train movements in the station were also recorded. Registered actions are necessary in order to move rail traffic as smoothly as possible should faults occur in the failsafe equipment; this function is intrinsically dangerous since the safety of rail traffic is ensured exclusively by the attention of the traffic superintendent. For this reason, special technical and regulatory conditions must be adhered to when using registered actions. Data was recorded primarily to observe the field trial and not to replace the train time printer and fault printer.

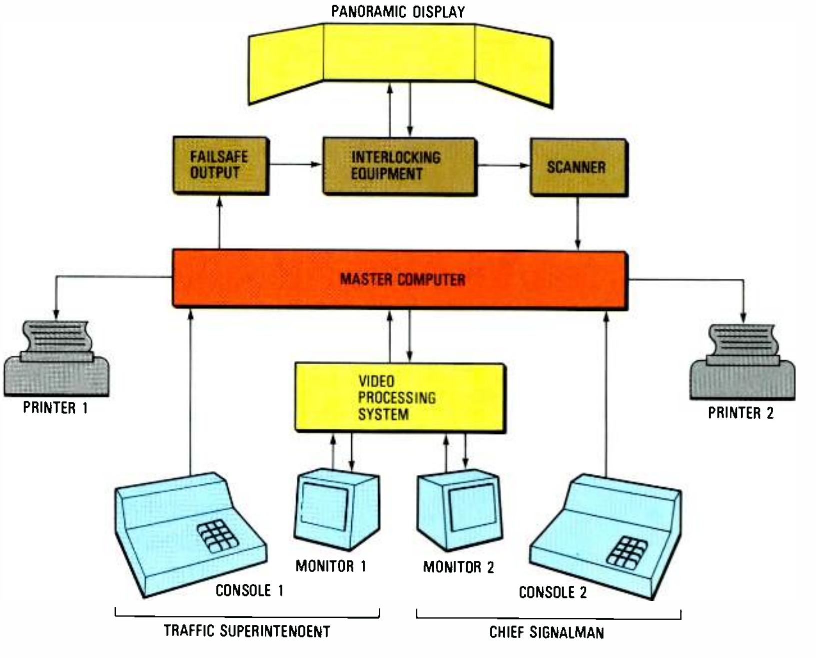
Information available in the master computer is also used to detect faults in the failsafe equipment. Fault detection for all essential equipment (e.g. axle counters, track relays, monitor loops) is easily integrated using the Videopult system, thereby achieving a higher availability of railway equipment.

Operational experience has shown that the information processing function can take over a substantial share of management tasks, without diminishing the operator's freedom of decision. Programmable shunting movements at the Wolfurt station are included in the area of information processing.

Configuration of the Videopult System

The main functional blocks of the Videopult system are shown in Figure 1. The scanner, which utilizes components from the ITT 0802 microcomputer system, is connected at the panoramic display (lamps) or the interlocking system. Depending on availability, free contacts are used for scanning. If no contacts are available, scanning is carried out at the lamps of the panoramic display via suitable interface equipment. This scanning flexibility makes it possible to connect the Videopult system to any type of technical facility. All input signals are cyclically interrogated. A digital filter suppresses transients caused by contact bounce. Sufficient logic switching is provided in the scanner for the logical evaluation of signals.

Figure 1
Block schematic of the Videopult system for the control of operations at a railway station.



At Wolfurt station approximately 1400 signal inputs are connected to the scanner. The memory requirement necessary for scanning and processing is approximately 54 kbyte. After evaluation, information from the scanner is passed to the master computer, which is also based around components of the ITT 0802 microcomputer system. The master computer provides central signaling for the Videopult system. The 124 kbyte memory stores all constant and variable data relating to the connected facilities. Essential data for display on the monitors is passed in serial format to the process video system. In turn, the master computer receives commands which have been input using the light pen. The pushbutton consoles are connected directly via their own inputs to the master computer. The output of signaling commands from the master computer to the signal box is via *compulsive guided relays* (relays used for the failsafe circuits in railway signaling). The contacts of these relays are switched in parallel with the corresponding buttons on the panoramic display board; thus commands input via the light pen have the same effect as pressing the corresponding keys on the panoramic display board.

Design of the Operator Position

Considerable importance was attached to human factors in the design of the operator position. All statutory and medical regulations and guidelines were taken into account during the design phase. In addition, electric motors rapidly after the display angle to suit the operator.

Operational Features

Track Diagram
The track diagram is based on previously proven diagrams.

The status of the failsafe facilities is continuously scanned, and the results are sent to the process video system. The displays for date and time are at top right. This data can be entered or changed by the operator via a numeric keypad on the console.

The names of the destination stations are displayed on a colored background. Train numbers are displayed with the same background colors as the designated destination stations, thus facilitating coordination of the destination and train number by the operator.

The function keys include not only group keys for the failsafe facilities but also labels (e.g. working gang) that can be placed on the desired track section using the light pen. Four track memories are located to the right side of the function border. These track memories allow rapid setting up of frequently used routes; they are analogous to facilities provided by the conventional number signaling console.

Operation of the Safety Facilities
As already described, commands are usually input using a light pen. The locking of routes or changing over of points is carried out using the familiar two or three button operation method. Registered actions are initiated in a similar way using the light pen on the monitor. If, for example, a substitution signal is set (e.g. should a green lamp fail), then the associated symbol of the main signal is activated using the light pen. The number of the track section before the main signal begins to flash as soon as the main signal associated signal relay at the output of the master computer is activated. However, the signaling command is only passed on to the route interlocking equipment after the operator is satisfied with the correctness of the track number and has pressed the enable and group buttons on the console keyboard.

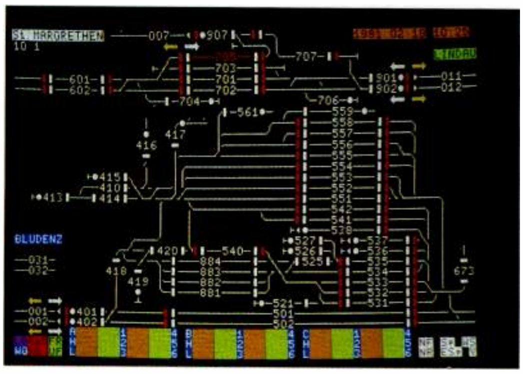
Indication of the track number is the same whether the Videopult light pen and monitor or the console keyboard is used to input the information.

Train Number Function
The indication and movement of train numbers is completely integrated in the Videopult system. This makes it possible not only to connect a desired number of stations by inputting the train numbers, but also to determine the routing of a train from the first digit of the train number or the whole number.

Changeover Movements for a Shunting Locomotive
The monitor picture available to the chief signalman differs from the traffic superintendent's monitor, especially in the function area. Here the three blocks of six lines are conspicuous and serve for entering train movement programs. The right hand block is used to enter shunting movements from one track to the next using the light pen.

The operator activates field C to advise the master computer that the following input is a program and not a signaling command. Next the track numbers are touched with the light pen in the sequence of the shunting movements. Track numbers are automatically written into the fields provided for block C. When the movement program has been entered, the operator again touches field C with the light pen. The

Monitor diagram for the chief signalman.



program or block C is run simply by activating line *1* in block C and field *V* (shunting key).

The master computer then automatically signals the first and all subsequent tracks as soon as the last common track element of two consecutive tracks is free. In this way the operator is relieved of a routine activity, the tracks are signaled at the earliest possible moment, and dead time required for human action is eliminated.

Fly Shunting Movements for Two Locomotives

Blocks A and B in the function border are provided for programming fly shunting movements. At Wolfurt station these are undertaken simultaneously from tracks 414 and 673.

The operator enters the movement program in a similar way to the changeover movements. He then activates, for example, field *A* with the light pen and subsequently the track from which shunting should commence. Subsequently only the numbers of the fly shunting movement destination tracks are touched with the light pen in the sequence that they arrive over the shunting radio. At each touch of the light pen, the track numbers are written into the corresponding fields of block *A*.

At Wolfurt station all three blocks are used at the same time, taking full advantage of the programmability of fly shunting and changeover movements.

Monitoring Functions and Records

These functions are not at present used for system operation at Wolfurt station. However, the demonstration of these facilities has proved valuable in the development of an automatic operation recording method using the Videopult system.

Results of the Field Trial

The field trial was evaluated by experts in the ÖBB and ITT Austria. It was clear from the trial that the Videopult system makes a major contribution to the management of railway stations. In particular, interactive operational control using a light pen proved to be excellent.

Training of operating and maintenance personnel did not present any problems, and there were no technical problems.

Finally, the stipulated cost target was met; the system offers a better cost/utilization ratio than the conventional system has achieved for the ÖBB.

The trial has also provided information that will be valuable for future Videopult system installations.

Conclusions

The field trial at Wolfurt station has provided a range of suggestions as to how the capabilities of the Videopult system can be enhanced to further improve the management of railway stations. Although at present it is too early to consider all the proposals, two changes have been implemented. One is the tying in of catenary group switching (switching of the traction current) with automatic interlocking of the disconnected track. The second is a computer assisted reporting procedure that could eventually replace all existing manual reporting books.

K. Lukaschek was born in 1938 in Vienna. He studied telecommunication engineering at the Technical University of Vienna, where he graduated in 1963. Since 1973 he has worked with ITT Austria where he is head of the department for railway control systems.

Enhanced CHILL Tasking Concept and Language for a Business Communication System

The CHILL programming language has been enhanced by the introduction of new facilities to support timeout handling, access to a database system, and specific input/output operations. These and new test features make CHILL suitable for use in business communication systems, and other applications such as railway signaling.

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ITT Austria, Vienna, Austria

Introduction

The major characteristics of the third generation office communication systems are the ability to switch digital voice and data traffic, a hierarchical multiprocessor control structure, and an advanced software architecture using a high level language and a real-time operating system. A typical example is the ITT 5200 business communication system (Amanda) developed by ITT Austria.

The programming language used for the ITT 5200 and other ITT Austria microprocessor products is CHILL, the CCITT high level language for telephone switching. CHILL is a real-time programming language for processor controlled switching systems. The static language properties of CHILL reflect today's state of the art by providing PASCAL-like constructs for structured programming and user definable data types.

The real-time multitasking capability of CHILL, which supports concurrent execution of processes, is essential to its use in switching systems. The main features supporting concurrent execution of processes are dynamic process management, process synchronization and communication, and mutual exclusion.

The ITT 5200 is based upon a complete CHILL environment, including compiler, operating system, and test tools. Implementation of this environment also aimed at supporting the use of CHILL in other real-time applications with different structures and demands, including public and private switching, and even railway signaling.

To meet the requirements of these applications, the tasking capabilities of CHILL were enhanced by introducing features such as timeout handling, special input/output operations, and database access. The additional tasking features were implemented without changing or extending the CHILL syntax, by using the concept of operating system primitives – built-in procedures with a predefined parameter and exception list.

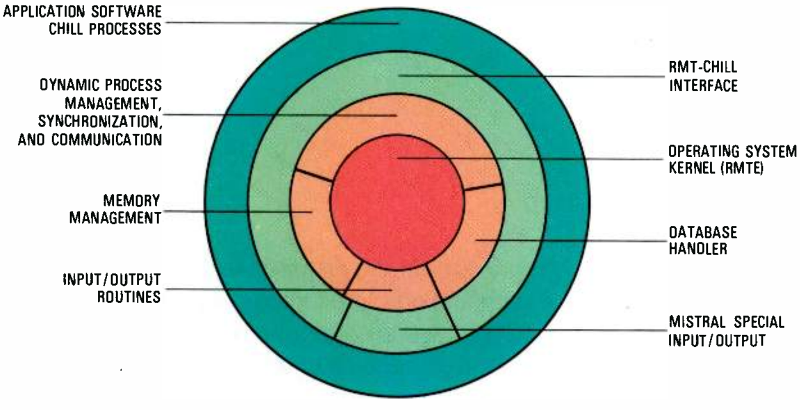
CHILL is much more than a programming language – rather it can be seen as a design philosophy for real-time systems. Unlike a static programming language without any real-time multitasking capabilities, a complete CHILL environment for embedded systems must also contain an operating system supporting the specific tasking concept on the target computer. The operating system developed for this implementation – RMT (real-time multitasking system) – supports the full set of enhanced CHILL tasking functions.

As different application areas often require different operating systems, a major design goal for the CHILL environment was to keep such changes transparent to the application program and to minimize their influence on the compiler. A hierarchical software model has been developed which meets these requirements.

When CHILL is used in switching systems, testing is important. Real-time applications in general provide restricted software debugging facilities because of the high cost of built-in test facilities. Most tests, therefore, have to be carried out in a special test environment which must permit a stepwise execution of CHILL programs and language level debugging. Displaying, tracing, and modifying user program data structures, as well as operating system data, must be possible on a logical symbolic level.

Especially in the case of real-time applications, tests must be performed under true real-time conditions. The test tool has to interact with the system under test without influencing its real-time behavior because changing the behavior of the tested system will, in most cases, affect the error symptoms of the faults under investigation.

Figure 1
Software hierarchy in a CHILL application.



As a result, a major effort has been put into the development of appropriate test tools for this CHILL environment. In particular, a real-time monitor has been developed for interactive or program controlled debugging of applications running under the RMT operating system.

Real-Time Multitasking Capabilities

A major aspect in the definition of CHILL was to incorporate all the necessary mechanisms for processor controlled telecommunication systems. The main objective is to support concurrent execution of processes.

In CHILL a process is defined as a possibly parameterized sequence of actions that may be started for concurrent execution from different places in the program. Execution of a *start* action installs a new copy – a so called *instance* – of a process. This new instance starts to execute concurrently and can stop and delete itself from the system by executing a *stop* action.

In a real-time application, concurrent execution of process instances must be synchronized. An instance must be able to delay itself until a certain event has occurred (hardware event, software event, timeout, etc). After the event has occurred, the delayed instance must then be reactivated. CHILL provides three message types for the synchronization and communication of processes events, signals, and buffers. Any process instance can delay itself to wait for one or more messages by executing a *delay* action (for events) or a receive action (for signals or buffers). After receiving a message, the instance is reactivated to continue processing.

Features such as timeout handling, database access, and specific I/O (input/output) operations, which are needed for industrial real-time applications, have been added to CHILL using the concept of operating system primitives. This method has made it possible to avoid extending the basic CHILL syntax.

In addition to its control and tasking capabilities, CHILL includes an efficient mechanism for handling run-time errors – *exception handling*. Exception handlers enable run-time errors to be processed directly in the process where they occur.

Software Hierarchy in CHILL

The software architecture of the CHILL implementation in the target computer is based on a 4-layer hierarchical model (Figure 1). The two innermost layers contain the RMT operating system. The operating system kernel is formed by all the system routines running without any interaction with the application software (e.g. scheduler, dispatcher, timer). The system support routines (second layer) fulfill resource management and input/output functions, such as dynamic memory management and process management according to requests from the application software.

A basic idea behind this concept is that the two operating system layers are logically and physically protected against direct access from the application program (outermost layer). Access to the operating system functions and resources can only be gained through the RCI (RMT-CHILL interface). This is a set of parameterized routines that can be called by the user program. The CHILL compiler translates all tasking actions and operating system primitives to calls of RCI routines. There are several advantages to introducing this extra level of control:

– A clear interface between user software and operating system improves program security (dynamic memory protection and privilege level protection can easily be implemented).

- Easy compilation of all CHILL tasking actions.
- Standard parameter and exception handling.
- Flexibility of using different operating systems supported by the same compiler.

Different application areas in many cases have different operating system requirements. Although a general purpose operating system would be the ideal solution, experience has shown that in some areas (especially in switching) operating systems with specific features are needed. However, as a CHILL compiler always represents a major tool investment (this implementation consists of about 250000 PASCAL lines), the last point has a major influence on the cost of development tools.

With a clear interface definition at the CHILL level, the target operating system can be replaced together with the operating system-CHILL interface without any change in the compiler (see Figure 2). The required effort is limited to the development of a new set of interface routines. A precondition is that the new operating system meets the CHILL tasking requirements.

Real-Time Multitasking Operating System

The RMT has been developed to support the enhanced CHILL tasking concept in microcomputer-based real-time applications. The RMT operating system, which was first used in the ITT 5200 business communication system, meets the following general requirements:

- many concurrent process instances
- large number of context switches
- short tasks (number of actions between two wait states)
- high message transfer rate
- short response times to hardware events.

These requirements could only be met by introducing an efficient context switch mechanism (context switch duration about 350 μs) and an optimized memory management algorithm.

In addition to dynamic process management and process synchronization using signals and events (both of which are contained in the Z.200 CHILL definition), additional tasking functions have been implemented to support the enhanced tasking concept:

- database handling for memory resident data relations
- timeout handling for signals and events
- dynamic memory management.

The RMT also contains specific support functions for switching applications, including generation of ringing cadences and dialing pulses.

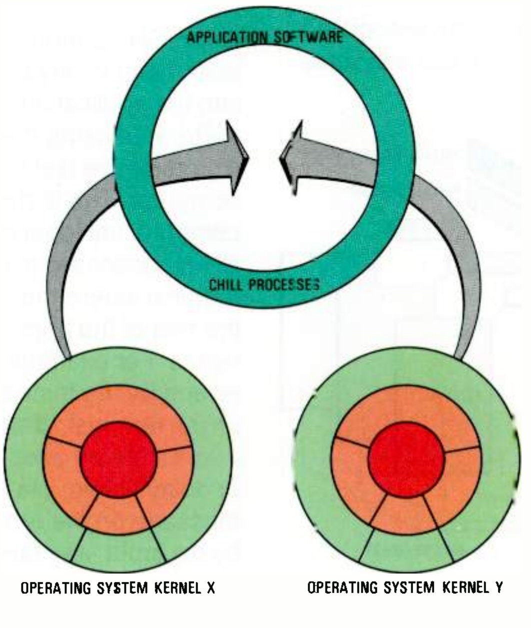


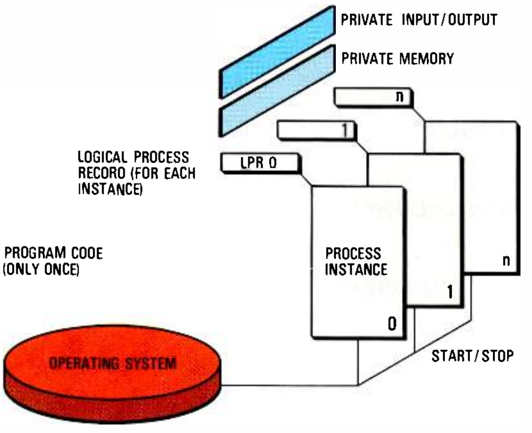
Figure 2
The operating system-CHILL interface allows the same compiler to be used with different target operating systems.

Dynamic Process Management

The RMT supports dynamic process management as defined in the CHILL tasking concept. The code for each application process physically exists only once in the system. When a CHILL program executes a *start* action, a virtual copy (an instance) is installed by the operating system (Figure 3). The basic idea behind this concept is that the actual program code is not duplicated, but only a small RAM (random access memory) buffer is allocated – the *logical process record*. This buffer contains all system data necessary for the concurrent execution of this instance. The logical process record also contains application variables which must be kept separately for each process instance to allow reentrant execution.

As soon as a process instance is installed it begins to execute independently until either it suspends itself to wait for a message or a stop action is executed. Execution of a stop action purges an

Figure 3
Dynamic process management.



instance from the system and the logical process record is freed.

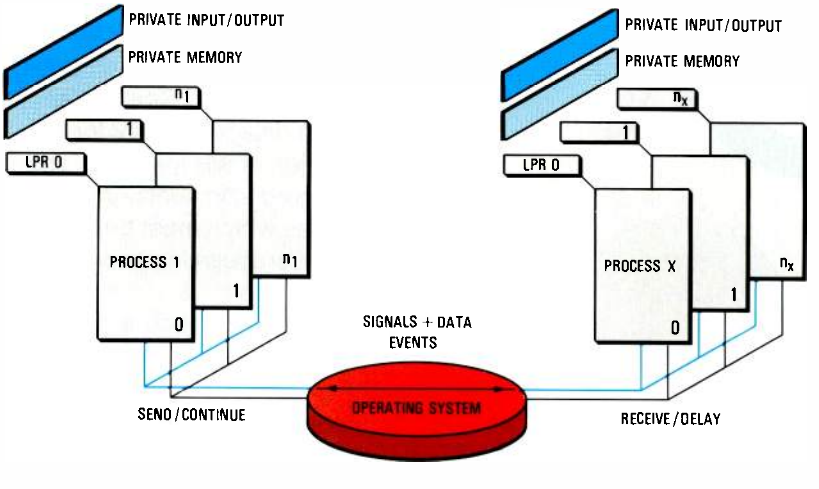
Process Synchronization and Communication
In the RMT, synchronization and communication of process instances is performed via signals and events; both mechanisms are recommended in CCITT Recommendation Z.200 (Figure 4).

Signals are messages that can also carry data. In the RMT they establish a direct communication path between two process instances. Using the *send* action (the *to* part is obligatory), any process instance can send a signal to any instance. By executing a *receive case* action, signals can be received by their destination process; a receive case action can contain one or more signal alternatives defining a set of actions to be executed after the signal has been received. Execution of a *delay* action with an event as parameter causes a process to suspend itself to wait for this event. In the RMT events are realized as mailboxes with one queue for the waiting processes for each event. When a process delays itself for an event, an entry is made in the associated queue. When any active process executes a *continue* action with an event as parameter, the process instance with the oldest entry in the queue is rescheduled and dispatched. The priority mechanism for events conforms to CCITT recommendations; when a priority is provided in the delay action, continuation of the process is in accordance with that priority.

Database Handler
Many applications, especially in switching, require large amounts of memory-resident data. The database handler is an independent subsystem of the RMT for managing the user access of two-dimensional data relations. These relations are physically protected against direct access by the user programs.

Access to data records and elements uses special operating system primitives (PUT_RECORD, GET_RECORD, etc), as shown in Figure 5. Additional security is provided by a special read/write access check mechanism. For each element in each relation, read and write access can be defined separately for each process type. Access rights are checked at run-time and an exception is raised should a violation occur.

Figure 4
Process communication and synchronization.



Test Tools

Major effort has been put into the development of appropriate test tools for real-time applications programmed in CHILL. Testing is performed in three steps:

- CHILL module test
- system integration test
- field test.

In the CHILL module test, each process is tested separately before being integrated into the application system. This test is performed using the CHILL module tester, an interactive test tool running on a host computer. The tester simulates the target computer and operating system, and directs all interactions with other processes to the terminal where the test engineer simulates the rest of the application system by his inputs. For example, all signals and events issued by the tested process are displayed on the terminal. When the process goes into a wait state on execution of a receive case or delay action, the engineer receives a message on the terminal and can simulate, by his input, any other instance sending a message to the process under test.

Because it interacts with the test engineer, the CHILL module test is not performed under real-time conditions; the logic flow is verified rather than the real-time behavior. Module testing can also be performed automatically (but not under real-time conditions) by using command and output files.

In the integration test, the CHILL processes are brought together with the operating system and the original application hardware. This test must be performed under real-time conditions.

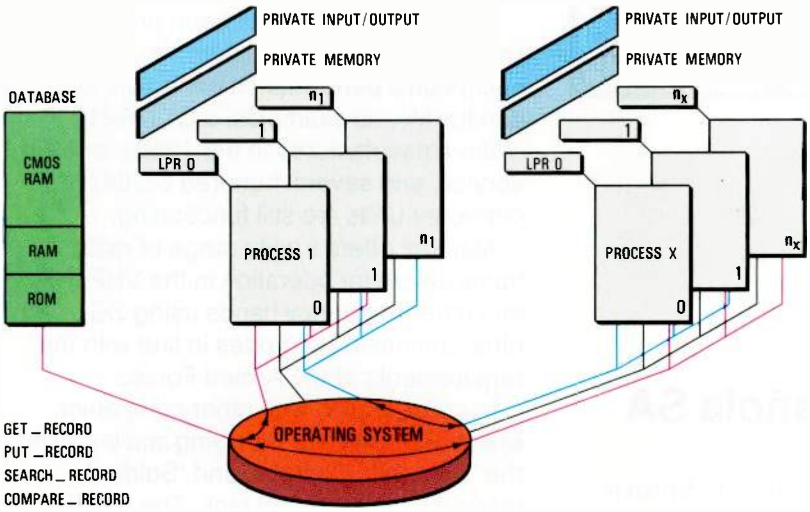


Figure 5
Database access.

Debugging a system in the field means that errors must be investigated on the running system without disturbing the system's functions. Another field test requirement is that most test procedures should be automatic, otherwise a large number of service engineers would be needed.

A real-time monitor has been developed to support the integration and field testing of CHILL applications. This monitor is a test computer which accesses the application hardware and programs via a bus interface (*field tester slave interface*) which can be inserted in the running system without disturbing its function.

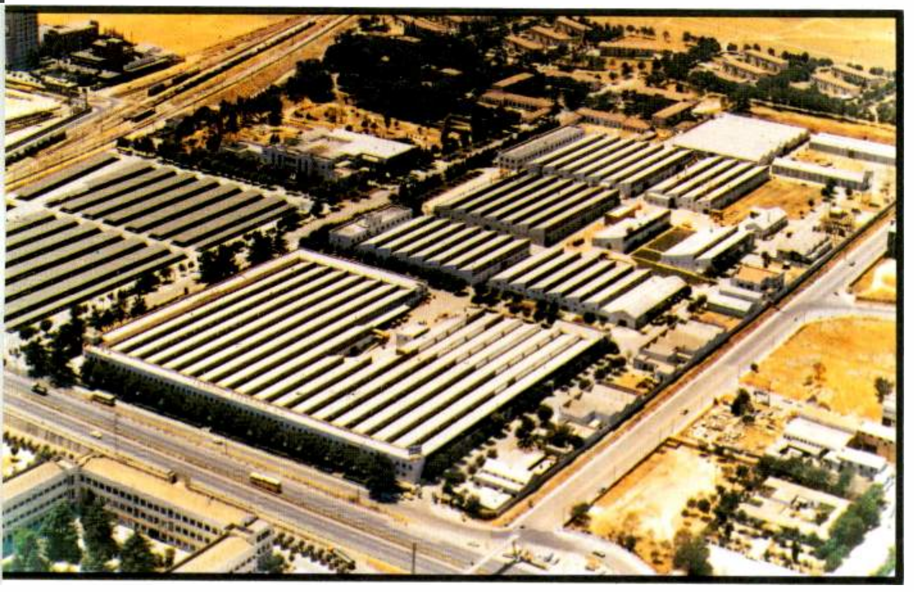
The real-time monitor enables the test engineer to display and modify system and application data structures on a logical and symbolic basis (e.g. message queues, timeout tables, and data relations can be accessed using special monitor commands). The real-time monitor has physical access to the entire address space of the slave computer and therefore allows addressing of all memory and input/output locations of the application system. Special data and address trigger hardware on the slave interface enables the real-time monitor to execute monitor commands according to trigger events. For example, a queue could be displayed whenever a new entry is added to it by setting a trigger to the queue counter and so executing a queue display command whenever it changes its value.

Using the real-time monitor, testing can be performed interactively or under program control. For this purpose the real-time monitor can be programmed in a high level programming language called PASSAT (PASCAL subset for application in test computers). In addition to the PASCAL derived language constructs, PASSAT contains a special statement for the execution of monitor commands. To support the development of PASSAT programs, the real-time monitor contains all the necessary software tools such as editor, compiler, and file manager (programs can be stored on cassette tapes).

Conclusions

An implementation of an integral CHILL real-time programming environment has been developed by ITT Austria, including compiler, operating system, and test tools. The CHILL tasking concept has been enhanced to meet the requirements of microcomputer-based real-time applications. A 4-layer software architecture allows the use of the same compiler with different operating systems. The implementation is an optimal solution not only for a third generation business communication system, but also for processor controlled products in areas other than switching.

Norbert Theuretzbacher was born in 1952. He studied telecommunication at the Vienna Technical University where he was awarded a Dipl Ing degree. He joined ITT Austria in 1975, and the following year assumed responsibility for developing real-time programming technologies. Mr Theuretzbacher is at present the manager of software development aids at ITT Austria. Recently he received the ITT Programming Recognition Award for the best programming tool chain.



Marconi Española SA

Marconi Española was formed in Spain in 1917. Initially the company produced some of the first wireless telegraphy stations of the type installed by its predecessor the 'Compania Nacional de Telegrafía sin Hilos'. The company first manufactured wireless telegraphy equipment within a few years of its invention; now, more than 65 years later, Marconi Española is still dedicated to using advanced technology, although the technologies themselves have changed. A highly qualified staff, extensive experience, and modern technical facilities are the foundations behind the company's application of microelectronics technology to both military and civil products.

While the primary market has always been the armed forces, this has not restricted development in other fields. Indeed this broad base has made it possible to branch out into areas such as railway signaling and control, television, and security devices.

With an annual production capacity of over 200 million US dollars, Marconi Española is dedicated to constant progress, providing an effective service in both the national and export markets. Its incorporation into the ITT System in 1968 provided additional technical, financial, and marketing resources.

Military Electronics

Ground Forces

Since 1917, Marconi Española has been the leading Spanish company in the field of military electronics, and has maintained a close relationship with the armed forces through changing circumstances. New products are regularly developed and manufactured, and the company also undertakes installation and maintenance. The area of military electronics includes radio and telephone communication, radar and navigation systems, and firing control and radar systems.

Marconi Española has been producing transmission equipment for many years. Its exceptional performance ensures a long service life; as examples, a C11/R210 station manufactured in the 1950s is still in service, and several hundred CC24/12 converter units are still functioning.

Marconi offers a wide range of radio transceivers for operation in the VHF, HF, and other frequency bands using SSB and other transmission modes in line with the requirements of the Armed Forces.

In collaboration with other companies, Marconi is currently supplying and installing the 'Colonel', 'Captain', and 'Soldier' models for the AMX30 tank. The RX80 coastal radar and the 'Superfledermaus' firing radar have been followed by modern gun and surface-to-air control systems.

In the paramilitary field, Marconi manufactures SSB equipment for the Spanish Civil Guard and radioactivity warning detectors for the Civilian Defense Staff and the Nuclear Energy Commission.

Finally, a group is studying an advanced digital tactical transmission system that will have a performance equal to that of the most modern military systems in the world.

Air Forces

Marconi Española supplies equipment for military aviation, including navigation aids, VHF/UHF communication sets, simulators, radiotelephones, and radar. In addition, the company maintains and repairs aircraft flightdeck equipment and instruments. Similarly, Marconi is a subcontractor to the United States Air Force, providing maintenance for electronic equipment in some of the world's most advanced aircraft.

Present negotiations with the manufacturers of the future combat aircraft to be selected by the Spanish Air Force are aimed at establishing a programme of industrial cooperation and participation.

Products for Civil Use

Close contact is maintained with the Ministry of Transport, including collaboration on projects for the supply, installation, and maintenance of communication systems (point-to-point and ground-to-air), navigation aids, air traffic control systems, and other aviation aids.

Marconi Española is also a pioneer in naval electronics. In cooperation with ITT associates, such as Standard Radio & Telefon of Sweden, the company has developed a range of marine transmitters and radiotelephones which are manufactured in Spain. These equipments provide intercontinental links through SSB HF transmitters with output powers of 100 W to 1.5 kW, and extend down to the VHF 25 W radiotelephones used by coastal services.

In 1969, a mobile radio product line was introduced. The range includes portable, mobile, and fixed multichannel equipment for both public administrations and the private market. Within the government sector, the main clients are the police and the armed forces. In the private sector, customers include taxi cooperatives, hydroelectric utilities, and petrol companies.

Another important activity is railway signaling and control. More than 30 years of experience has culminated in the following railway signaling and control systems:

- centralized traffic control systems
- geographical (TRICON-E) and conventional railway signaling systems
- automatic train braking system
- continuous control system for train operation
- remotely operated railway crossing.

In addition to the above areas, Marconi Española is continuously exploring the possible entry into new fields and products in response to customer demands, and introducing new products developed by the engineering department.

P. Regatero
Managing Director
Marconi Española
Madrid, Spain

Mobile UHF Transceiver RF Unit

A compact transceiver RF unit using analog and digital LSI circuits has been developed for use in mobile radio equipment for the 0.7 m band. It fulfills all the relevant CEPT recommendations.

J. L. Garcia Semov
E. Diez Kowalski
Marconi Española SA, Madrid, Spain

Introduction

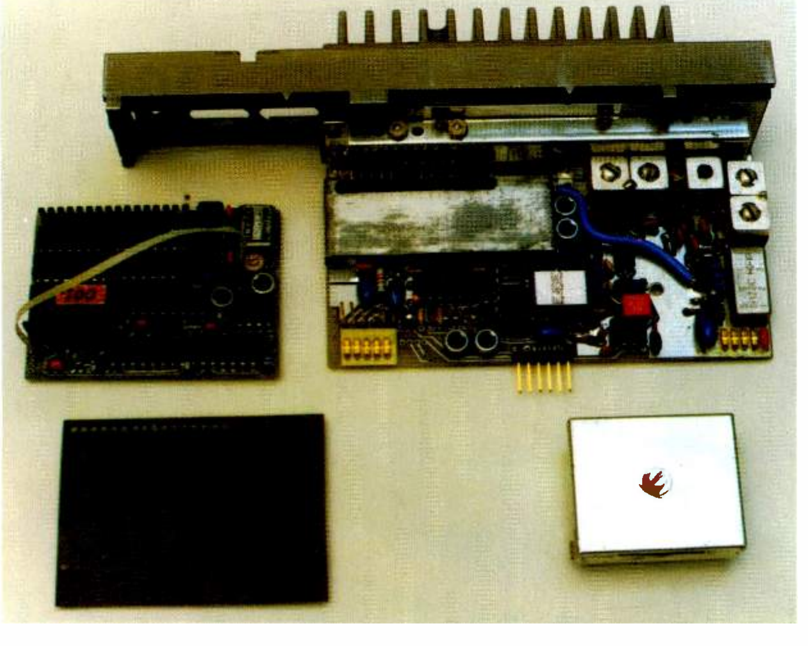
The characteristics of both professional and commercial radiocommunication equipment, mobile and fixed, currently available on the market are similar. Transmitter and receiver have a relatively narrow RF (radio frequency) bandwidth and the transmit-receive frequency spacing is limited by the design. Each transmit and receive channel requires its own crystal to determine the frequency. Multipliers are then used to produce the local oscillator signal. The RF bandwidth and the space available for the crystals limit the number of channels that can be accommodated. The channel spacing is also restricted, usually to 20 or 25 kHz but in some cases to 50 kHz. Extensive use of discrete components makes the equipment relatively bulky.

Increasingly the wide range of standards and specifications is affecting the design of radio equipment which is often required to comply with a variety of national standards, such as FCC and EIA standards (USA), EST and EFT (France), British Telecom (Great Britain), FTZ, DIN, and VDE (West Germany) as well as international standards such as those of CEPT (Conférence Européene des Postes et Télécommunications).

These considerations led Marconi Española to set seven principal design objectives for the transceiver RF unit:

- maximum use of LSI (large scale integration) technology
- high reliability
- minimum number of adjustments
- rapid and simple servicing
- use of modern manufacturing and test methods
- compliance with all relevant CEPT specifications
- compact mechanical design.

Transceiver RF unit shown assembled and as separate modules.



General Concept

The unit operates in the 0.7 m UHF (ultra high frequency) band between 440 and 470 MHz but can be modified relatively easily for use in the 0.8 m band down to 406 MHz. Channel spacings other than the basic 20 and 25 kHz, 12.5 kHz for example, can be obtained with this unit. Operation is possible in either simplex or half-duplex mode, or both, as required.

In half-duplex operation, the spacing between the transmit and receive frequencies is limited only by the receiver RF bandwidth (20 MHz) and the transmitter RF bandwidth (30 MHz).

The basic multichannel version can be programmed with up to 20 receive and 20 transmit frequencies within the RF bandwidth.

Mechanical Design

The principal modules of the transceiver RF unit are:

- common transmitter/receiver printed board with transmitter heatsink
- plug-in transmit/receive VCO (voltage controlled oscillator)
- channel printed board with reference oscillator.

The main printed circuit board holds the complete receiver front end (from the antenna input to the hybrid mixer 21.4 MHz output), synthesizer circuit including the phase-locked loop, transmitter modulation control circuit, transmitter output power control circuit, and VCO module control circuit. The compact VCO module, which acts as the local oscillator, plugs into the main printed board and can easily be changed.

The antenna filter board is soldered to the main printed board. Its cover provides shielding and a common ground. The antenna filter board is screwed to the heatsink for the hybrid power module and the + 5 V integrated regulator.

Electrical Design

Figure 1 is a simplified block diagram of the transceiver RF unit. The antenna filter board carries the harmonic filter, forward and reflected RF power detector, and electronic transmit-receive switch. The receive amplifier module carries two tunable bandpass filters and a variable gain amplifier. It is followed by the hybrid integrated mixer.

Output power from the VCO (and hence the transmit RF power) is held constant by the power control module. The transmit power module includes an input drive attenuator. Temperature compensation can be provided for the transmit power module as an option.

Several of the modules are common to transmitter and receiver in order to simplify the design. They are:

- wideband VCO
- synthesizer
- channel printed board.

The synthesizer module includes a variable modulus prescaler which divides the input frequency down to 2.5 MHz. It is followed by the loop filter module which consists of a low pass phase-locked loop filter and DC amplifier. Preceding the synthesizer is the channel module containing the 6.4 MHz reference oscillator and the channel printed board with PROM (programmable read-only memory) in which the operation mode, channel number, and transmit and receive frequency information are stored. A separate module provides temperature compensation for the reference oscillator. The common transmit-receive VCO module follows the loop filter.

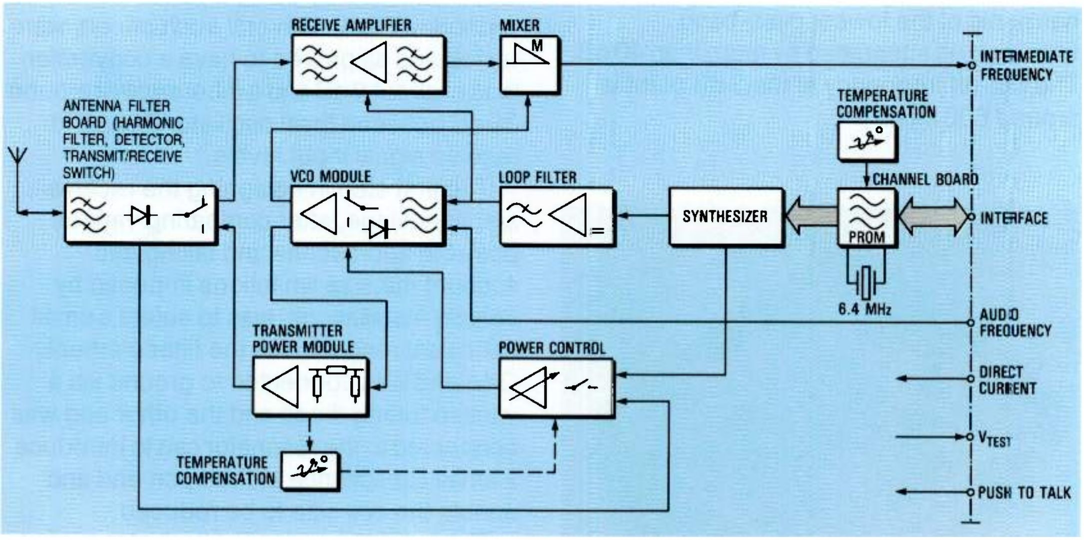


Figure 1
Block schematic of the transceiver RF unit.

Transmitter

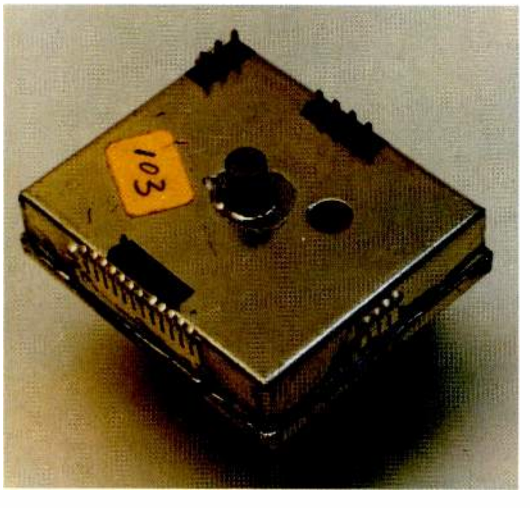
The output power of the VCO module at the transmit channel frequency is high enough to drive the transmit power module directly.

There is sufficient reserve to allow for compensation of the variations in the power amplifier output level with frequency and drive level.

The transmitter power amplifier is a commercial hybrid module consisting of three amplifier stages with an overall gain of approximately 19 dB. A power output of between 10 and 13 W was chosen as a compromise between RF unit performance and local market requirements.

The output of the transmitter power module is fed to the common antenna filter board. The electronic transmit-receive switch uses two PIN diodes, one for the transmit path and the other for the receive path. The switch is changed over by biasing off one diode or the other with 10 V DC.

The low-pass harmonic filter has 50 Ω input and output impedances, 0.5 dB transmit path insertion loss, and 1.5 dB receive path insertion loss. The first harmonic of the lowest pass-band frequency is attenuated by more than 40 dB. The roll-off frequency at the 3 dB point is around 600 MHz.



VCO module. Its small size simplifies handling and replacement.

The RF power monitoring circuit uses a Π-section filter at the module output to extract both forward and reflected power. After the capacitive divider, two RF switching diodes rectify the signals and feed the sum to the control circuit.

Regulation of the transmitter output power is carried out automatically by adjusting the drive power to maintain the output power constant at the preset level. The control circuit also reduces the power in the event of a short circuit or open circuit at the antenna connector, not uncommon in mobile installations. The transmitter output power response meets CEPT recommendations for supply voltages between 10.7 and 15.6 V.

No retuning is required after a frequency change as all sections are wideband and have an almost flat frequency response.

Receiver

The design objective for the receiver front end was to develop a wideband unit that could be tuned automatically to the receive channel frequency. It was also to be compatible with the IF (intermediate frequency) section of the SE 205 series of mobile radio equipment. This 21.4 MHz IF section provides the specified signal-to-noise ratio with an input of only 0.33 μV. As this is the level specified at the receiver antenna input, the overall RF unit gain must be unity. The number of alignment points was minimized by using a commercial wideband mixer. Several such mixers were tested and all proved to have a conversion loss of about 9 dB and a difference of around 35 dB between local oscillator drive and receive signal input levels.

The first step in designing the receive amplifier stage, after comparing various possible approaches and taking into account the size limitations imposed by vehicle installations, was to select a small helicoidal resonator as the filter element. One end was connected to ground via a varicap tuning diode and the other end was connected to the resonator can to introduce a small capacitance at the open end and enable the coil size to be reduced.

The tunable bandpass filters at the input and output of the receive amplifier each consist of two aperture-coupled resonators. A loop near the end connected to the varicap diode couples the filter to external components. The filter is tuned by varying the DC voltage on the varicap diodes. This voltage is taken from the DC amplifier output of the loop filter and varies sufficiently to tune the filter over the entire 20 MHz receive pass-band.

Voltage Controlled Oscillator

For the sake of design simplicity it was decided to develop a device that could be tuned over the whole operating range without readjustment. The lowest receiver local oscillator frequency is 418.6 MHz and the highest transmitter oscillator frequency is 470 MHz; this defines the range of

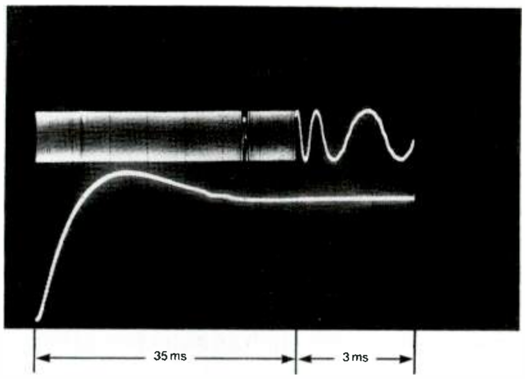


Figure 2
Synthesizer locking time for a change from 418.6 to 470 MHz. After about 35 ms the frequency error is less than 1 kHz.

frequencies covered by the synthesizer loop filter output circuit. The tuning voltage varies from 2 to 9 V to give a tuning input sensitivity of about 7.34 MHz V^{-1}.

The principal difficulty with the design of the VCO was to keep the phase noise below 110 dBc with a 10 kHz channel spacing. After analyzing and testing various active devices and several different oscillator configurations[4, 5], an FET transistor was selected in a grounded gate configuration with low feedback. The oscillator tank circuit uses an air-cored coil wound from silver wire together with a capacitor and varicap diode combination.

The signal is fed from the buffer amplifier to the distributor stage, a dual-gate MOSFET which supplies RF level to the mixer, to the transmitter power module, and back to the prescaler. The drain of the MOSFET is connected to two PIN diodes which switch the VCO output to the mixer when receiving and to the transmitter power module when transmitting. The loop back to the prescaler is taken from the drain of the MOSFET.

Synthesizer

A variable 200/202 modulus prescaler in the synthesizer directly divides the VCO output frequency down to below 2.5 MHz. This frequency is fed to the programmable dividers in the synthesizer where it is divided down. The output of the 6.4 MHz reference oscillator is also divided down to the same frequency and the two are compared. The synthesizer dividers are programmed by the PROM on the channel board.

Passive components are used in the second-order RC loop filter in order to avoid the additional noise which would be generated by active devices[6].

The following measurements were made to verify the design concept:

- Synthesizer locking time (the time required to change from 418.6 MHz to 470 MHz, the largest locking step). The result (see Figure 2) was around 35 ms.
- Measured sideband noise was −120 dBc[4, 5].
- Spurious response, especially at the sidebands generated by the reference frequency. This was found to be as low as −70 dBc.

Channel Board

The channel board consists of three circuits:

- 6.4 MHz reference oscillator
- PROM
- interfaces.

The reference oscillator frequency is determined by a 6.4 MHz crystal together with a group of inverter-buffers in a CMOS integrated circuit. The crystal circuit includes a varicap diode to correct frequency errors caused by temperature variations. The temperature sensor is a silicon diode connected in one arm of the compensation bridge, balanced at +25 °C. Voltage variations across the bridge caused by temperature changes are amplified by dual operational amplifiers and fed to the varicap diode to correct the tuning of the oscillator.

Fifteen bits are required to program the synthesizer dividers. The memory capacity is 256 × 8 bits and two words are required to program each frequency. These two words are read out sequentially from the memory. A short pulse is connected to one of the memory addressing inputs when the channel or operating mode is changed; this causes the first word (8 bits) to be read and sent to the latch circuit, the outputs of which are connected to eight synthesizer inputs. The second word (7 bits) is read and passed to the remaining synthesizer inputs when the pulse amplitude decreases to zero. This enables the full capacity of the memory to be used.

The eight address bits are allocated as follows:

- Bit one initiates a transmit-receive changeover.
- Bit two is the pulse that causes the sequential words to be read.
- Bits three to eight are used for preset channel selection. These six bits allow a

theoretical maximum of 64 channels but the number of preset channels is limited to 20 in the present design, divided into two groups of 10.

The selected preset channel is read out using pulses generated by a counter and displayed on an LED indicator on the front panel. Bit seven is used to designate the selected group; if the selected channel is in the second group this is indicated by displaying the decimal point to the left of the channel number. Bit eight is not used and is grounded.

When one of the groups is programmed for less than 10 channels then the first unused channel is programmed with the three most significant bits as zero, a combination which cannot occur in a normal channel allocation. These three bits are fed to three inputs of a NOR gate which has a logic *1* output when the selected channel number exceeds the programmed capacity of the PROM. This output resets the channel selector to channel *1* (or to channel *11* if in the second group).

Conclusions

Although Marconi Española has achieved its objective of using modern technology to design a small transceiver RF unit that meets all the relevant CEPT recommendations, they are nevertheless studying new technologies which might be used to further improve the performance of the equipment or reduce its size. Technologies under consideration include:

- LSI synthesizer circuits
- wideband hybrid amplifiers
- surface wave filters and oscillators for higher frequencies.

Some of these components are already widely used in certain domestic items such as television sets, broadcast receivers, and high fidelity sound equipment, and will soon be introduced into professional radio equipment.

Decreasing costs and increasing scale of integration now allow the use of microprocessors in mobile radio equipment. This is likely to result in the inclusion of additional facilities such as selective calling, data transmission, channel scanning, and cryptophony.

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J. L. Garcia Semov was born in 1944. He studied electrical engineering at the Polytechnic University of Prague, Czechoslovakia, where he was awarded a Dipl Eng. In 1970 he joined Marconi Española, where he was involved in the design of marine and mobile radio communication equipment. He is now responsible for the mobile radio design and system department.

E. Diez Kowalski was born in 1944. He studied telecommunication engineering at the Polytechnic University of Madrid where he obtained the degree of Ing Superior. In 1973 he joined Marconi Española to take charge of military and civil communication equipment design. He is currently head of the defense product design department.

Adaptive Differential Pulse Code Modulation Coder for Low Bit Rate Transmission of Speech Signals

The performance of speech encoders has been considerably improved by a new adaptive differential pulse code modulation coder. This is based on an improved understanding of the parameters that characterize the input signal.

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Introduction

Development work at Marconi Española has been directed towards designing an efficient speech coder capable of transmitting speech at low bit rates. Fixed prediction of the first order is based on feeding back a single sample to the quantizer; the sample is delayed by $1/f_m$ and processed to ensure the minimum difference between the input sample and the value obtained from the prediction process. In conjunction with adaptation of the quantization step size from the previously coded sample, this leads to an improvement in the signal-to-quantization-noise ratio, at the same transmission rate, of about 12 dB compared with compressed logarithmic PCM sample encoding with a compression μ of 100. This has been achieved without using especially complex circuits.

Choice of Parameters

The search for effective methods of coding speech is hindered by a lack of information on the behavior of speech signals. However, the main problems are well known:

- Over a long speaking period, the average speech signal level differs from one person to another.
- For a given average speech level, the instantaneous level changes as a result of variations in the speech sounds.
- The correlation between successive input voice signal samples varies with time.

One way of largely overcoming these problems is to use adaptive quantizers and predictors that derive an estimate of the signal parameters at regular intervals, and adapt their behavior according to these parameters.

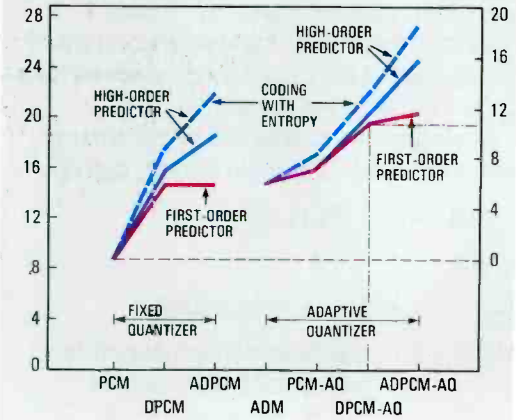
To eliminate the problems of unknown average level and variations in the instantaneous level, adaptive quantization can be used in which the encoded sample is processed to determine the size of the step to be used for the next sample. An advantage of this approach is that the information does not need to be transmitted to the receiver, thereby optimizing the use of spectral bandwidth:

$$G_t = F(I_t)\, G_{t-1}$$

where

G_t – updated quantizer gain

Figure 1
Signal-to-noise ratios due to quantization for different 3-bit digital speech encoding methods, and gain compared with standard PCM encoding (companding low $\mu = 100$). Values are given for PCM, adaptive PCM, and adaptive differential PCM with a fixed quantizer and the adjustable quantizer (AR); adaptive delta modulation is also included.



$F(I_t)$ – function, invariant in time, which determines the size of the next step from the encoding assigned to the previous sample.

Figure 1 is based on real estimates derived for the band between 200 and 3200 Hz with a sampling frequency of 8 kHz. The enunciation time was between two and three seconds.

As mentioned in the introduction, for a given transmission rate the use of a fixed predictor and an adaptive quantizer based on the previous sample coded (Figure 2) offers an improvement of about 12 dB in the signal-to-quantization-noise ratio compared with the use of logarithmic PCM encoding with a compression μ of 100 (Figure 3). The prediction criterion is that of optimizing the ratio between the input signal and the difference between the input signal and the predicted signal; this is equivalent to minimizing the variance of the difference signal:

$$G_p = 10 \log \frac{E(X^2(t))}{E(\delta^2(t))} = 10 \log \frac{\sigma x^2}{\sigma \delta^2}$$

where

X_t – input sample at instant t

$\delta(t)$ – input signal to the quantizer at instant t

G_p – factor that can be calculated directly from the standard self-correlation function of the input signal.

Modulator

Adapting the Step Size

It is assumed that the signal to be coded is 1.4 V peak-to-peak. If this signal were to be coded by a nonadaptive coder while keeping the transmission rate below 24 kbit s^{-1}, the eight possible levels (three bits per sample) would be separated by 1400/8 = 175 mV which, using logarithmic compression, would lead to a signal-to-quantizing-noise ratio of about 8.7 dB. The use of an adaptive equalizer makes it possible to increase the dynamic range of the signal that the quantizer can handle for a given signal-to-noise ratio.

The step size is adapted in accordance with the previous sample coded, that is:

$$\Delta_t = \Delta_{t-1} M(|P_{t-1}|)$$

where

Δ_t – step size at instant t

$M(|P_{t-1}|)$ – coefficient which depends on the absolute value of the previous coded sample.

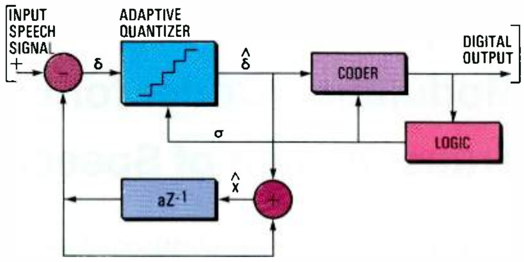


Figure 2
Block diagram of coder using differential PCM with an adaptive quantizer.

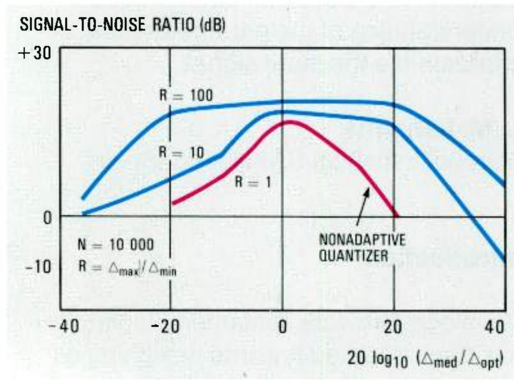


Figure 3
Improvement in the signal-to-noise ratio using adaptive quantization.
N - number of samples in the input sequence.

There will be maximum (Δ_t *max*) and minimum (Δ_t *min*) limits which ensure that the peak-to-peak range of the quantizer input signal is fully covered.

In accordance with the criteria mentioned previously, for a speech signal between 200 and 3200 Hz, the following adaption coefficients are obtained from the spectral density function of the input voice signal:

Prior output signal	Chosen multiplier
111 or 000	2
110 or 001	1.25
101 or 010	1
100 or 011	0.875

With $M = 0.875$, to decrease the signal from a sample at +20 dB to a situation with no signal present would require about 45 samples; at a sampling frequency of 8 kHz this takes about 5.6 ms.

Step Size

Knowing the values of the adaption coefficients that are ideal for speech signals, it is possible to derive a constant value which, when multiplied by any of the coefficients M_n, will cause a change by a whole number of steps. To do this, the following approximations can be made:

$2^{-1/3} = 0.7937$	to decrease one step.
$2^0 = 1$	to remain on the preceding step.
$2^{1/3} = 1.26$	to increase one step.
$2^{2/3} = 1.587$	to increase two steps.
$2^1 = 2$	to increase three steps.

The coefficient will therefore be $2^{1/3}$ since this makes it possible to approximate the coefficients as whole powers of $2^{1/3}$.

$$\Delta_{max} = 128 \text{ mV}$$

$$\Delta_{min} = 1 \text{ mV}$$

$$R = \Delta_{max}/\Delta_{min} = 128$$

$$\Delta_{mean} = (\Delta_{max}\Delta_{min})^{0.5} = 11.313.$$

This choice makes it possible to achieve a flat response over the whole dynamic range (see Figure 3).

Consequently there will be 22 step sizes given by the following table:

1	1.259	1.587	2	2.519
3.174	4	5.039	6.349	8
10.079	12.69	16	20.16	25.38
32	40.32	50.79	64	80.576
101.44	128			

Logic for the Integrator Loading-Unloading Control

This logic unit loads and unloads the integrator according to the bis and *code* signals and the sampler output levels. The *bis* signal is generated in the comparator and handles information for charging and discharging the integrator; it indicates whether the value of the previous prediction was higher or lower than the sample present at the input at that instant.

$Xi > \hat{X}i$ *bis* = '0'
$Xi < \hat{X}i$ *bis* = '1'.

The *code* signal determines the encoding period of 105 μs; *code* = 0 indicates readjustment. Therefore, the system should be capable of:

- Maintaining the previously selected step size using simultaneous positive and negative signals.
- Loading the integrator and canceling unloading, and vice versa.
- Controlling the loading/unloading switches using the *bis* and *code* signals.

Sampler

The *level* signal, which provides information on the previous step size selected, is input to the operational amplifier IC*1* (see Figure 4). The signal at the output of IC*1* is designated M^+, which is equal to M^-, the signal at the output of IC*2*, as long as the resistances RP*2* and RS*2* are the same. The current can be adjusted by P*1*.

Switch Control

Depending on the *bis* and *code* signals, the levels at the switch inputs are chosen to either saturate or cut off transistors T*1* and T*2* in accordance with the following table:

bis	*code*	T*1*	T*2*
1	1	OFF	ON
0	1	ON	OFF
X	0	ON	ON

Switches

When a low signal level is present at the base of T*1*, the transistor cuts off; simultaneously, a high level at the base of T*2* causes it to saturate. This implies that the signal M^+ will be present at the input of the integrator IC*3*. In contrast, when T*2* is cut off, M^- appears at the input of the integrator.

Resistances R*2* and R*4* control the time constant of the integrator.

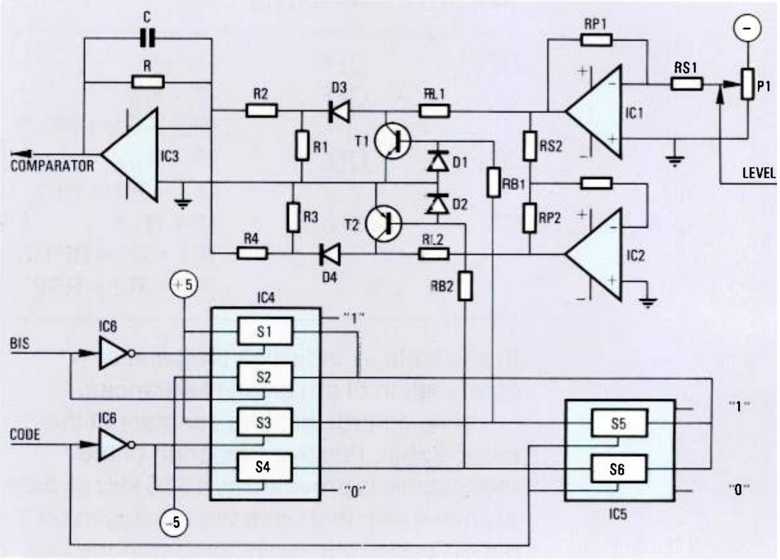
Logic for Controlling Step Size

This unit detects the coding of the previous sample and delivers a BCD (binary coded decimal) signal which selects the size of the next step (see Figure 5).

As already explained, step sizes should be selected according to the coding of the preceding sample so that the step size is shifted up or down with respect to the previous step size. For example:

Previous size	$\Delta_t = 20.16$ mV
Coding of previous sample	110

Figure 4
Circuit of the integrator and loading-unloading control.



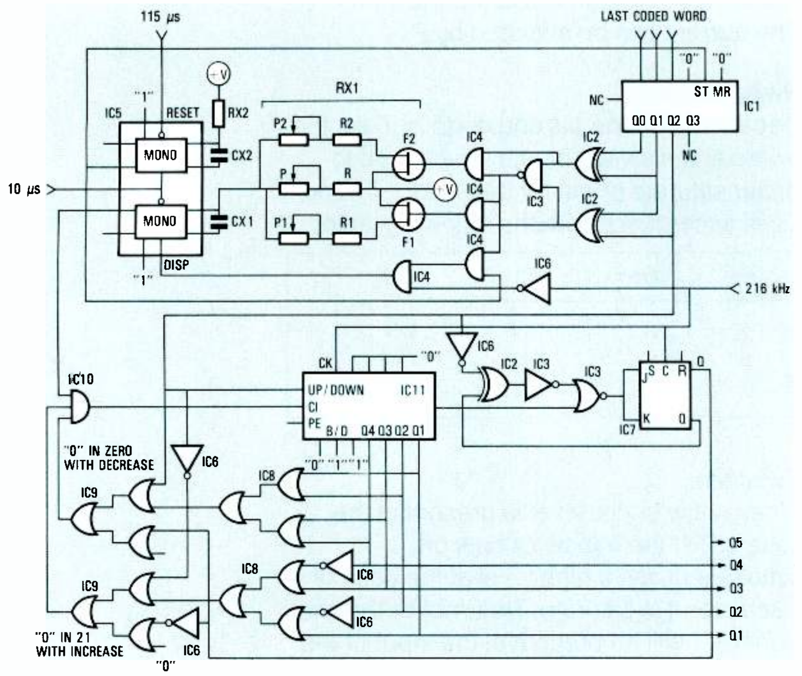


Figure 5
Logic for the selection of step size.

Coefficient to be used	M3
New size	$\Delta_t + 1 = 32$ mV

The logic should interpret that a coefficient M3 is to be used and, therefore, that the new step size will be obtained by an increase of two steps.

The three bits which constitute the coding of the previous sample are stored in the separator output; the stop control is provided by the $\overline{code}$ signal which acts as *enable*. Thus the signal is separated during the coding period.

The gates IC*2/1* and IC*2/2* together with a small logic element act on switches F*1* and F*2* which control the time constant of the monostable IC*5*, given by the product RX1.CX1. Thus the resistance RX1 at the input to IC*5* will be:

Switch F*1*	Switch F*2*	RX1
OFF	OFF	P + R
ON	OFF	(P + R)// (P2 + R2 + RF2)
OFF	ON	(P + R)// (P1 + R1 + RF1)
ON	ON	(P + R)// (P1 + R1 + RF1)// (P2 + R2 + RF2).

In this table, // indicates the parallel combination of the given resistances.

These control the time constant of the monostable. Positive triggering of the monostable is provided by a 216 kHz stream in such a way that once the readjustment period has begun, coinciding with the first fall in the 216 kHz signal, the cycle of monostable 1 begins. The signal from this monostable actuates an enabling gate of the up/down counter constituted by IC*11* and IC*7*. When the output Q of IC*7* is zero, the BCD output of the ensemble is lower than 16, since the output will be the bit of weight 16.

The input CK receives the 216 kHz signal from the clock generator. The operation of this part is as follows: once adaption of the step size has begun and the time for which CI(5) should remain active has been determined, the up/down counter is incremented either up or down, depending on the state of the up/down pin, on receipt of the 216 kHz clock from the clock generator. The duration can never exceed that of three clock pulses.

In order to achieve the required 22 steps, a small supplementary logic unit has been added; this consists of a flip-flop so that the output QJK1 constitutes the bit of weight 16.

The function of monostable 2 is to provide the *code* signal. It is triggered by the negative edge of the differentiation pulse at the 10 μs instant (start of coding). Excursions of the monostable are controlled by the time constant TX2 (= RX2.CX2); it is reset by the differentiated pulse after 115 μs.

Thus the BCD signal takes a value from 0 to 21, indicating after each sampling period what step size should be used for the next sample to be coded in relation to the preceding step size.

Comparator

The comparator produces a high or low output signal depending on the difference between the signals applied to its inputs. The samples of speech signal X_i are applied to the comparator's inverted input for the 125 μs sampling period; simultaneously the signal from the predictor is applied at the non-inverted input. The comparator then delivers at its output a signal of about +3 V if the sample $X_i < \hat{X}_i$, or no signal if $X_i > \hat{X}_i$. This signal, which will be present at the shift register input, is sampled at the instants 10, 66, 94, and 108 μs, giving a sequence of ones and zeros for subsequent transmission.

Conclusions

The coder described is a further stage in reducing the bit rate needed for speech transmission while maintaining an adequate signal-to-noise ratio. However, its field of application, as far as communication is concerned, is restricted to channels with 50 kHz of bandwidth and adequate premodulation filters.

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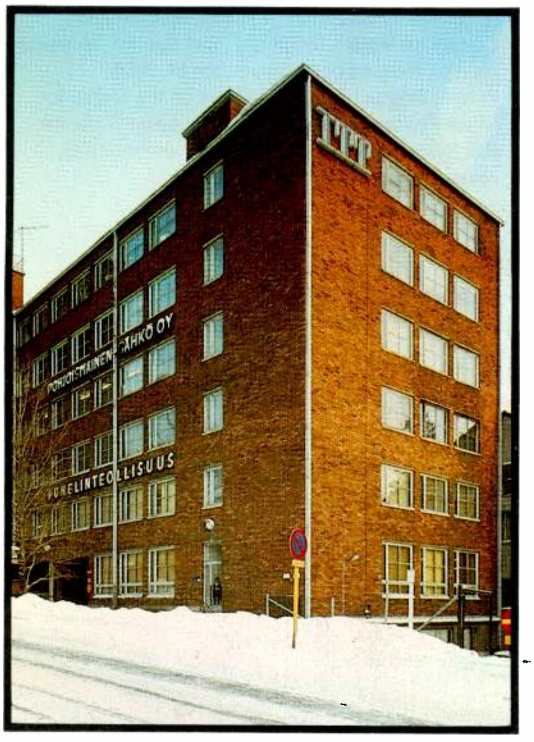
Luis M. Lafuente was born in Madrid in 1950. He studied telecommunication engineering at the Madrid Polytechnic University, graduating in 1975; he was awarded an MDA by the Instituto de Empresa of Madrid in 1982. He has experience in several areas, including radar, digital signal processing, communication, and navigational aids. He joined Marconi Española in 1976 where he is now head of the development and systems department.



Standard Electric Puhelinteollisuus Oy

The origins of the company go back to 1940 when a company was established in Finland under the name Puhelinteollisuus Oy. In 1954 ITT set up a company in Finland, and in 1962 the two were merged under the present name of Standard Electric Puhelinteollisuus Oy (SEP).

SEP has played an active part in the automation of the Finnish telephone network with the development, production, and installation of A-204 and HKS systems, of which 250000 lines have been installed. More recently, SEP has supplied METACONTA* exchanges for the Finnish network and to private telephone companies. Based on this extensive experience, the company is now able to supply the ITT 1240 Digital Exchange for use in Finland. In addition, the company is an important producer of private branch exchanges, starting with the Citomat* family and progressing to today's modern semi-electronic Minimat* range, developed by ITT associates Standard Electrik Kirk and Standard Telefon og Kabelfabrik, and a new data system development.

SEP has also installed air navigation systems and air traffic control systems at the larger Finnish airports.

* A trademark of ITT System

Resources

Traditionally SEP has had manufacturing facilities for conventional crossbar exchanges and private automatic branch exchanges, as well as telephone subsets. These basic resources have been upgraded during the past few years to keep abreast of the new electronic era. In the mid 1970s, a decision was made that SEP should possess comprehensive engineering and development capabilities. As a result the engineering department has grown steadily with the employment of many new graduates in a number of disciplines, including computer-aided design, software design, and system design.

While the primary resource is the skills of SEP's staff, they have had to be provided with modern tools, including printed board design equipment and program development and test systems.

Product Development

SEP's main products are designed in Finland. The key to these products is SEP's own microprocessor system which consists of a series of printed boards for telephone applications, together with corresponding functional program modules. These modules can be used to build a wide range of switching systems for companies in Finland, and in the USSR – the main export market for Finnish telecommunication manufacturers.

Stored Program Control Exchanges
Various functions in a conventional crossbar exchange can be modernized by using modern microprocessor control techniques. Examples are register functions, subscriber facilities, the collection of charging data, and maintenance facilities. Completely new crossbar exchanges have been built using this stored program control principle.

Network Maintenance
One of the most interesting applications in this area is the central office maintenance system which provides up-to-date maintenance facilities for conventional crossbar and step-by-step telephone exchanges. Installation of the system in conventional exchanges allows a minicomputer-based network maintenance center to manage the total network by communicating directly with modern stored program control exchanges (digital or analog) and with central office maintenance

terminal systems in older types of exchange. The same microprocessor system modules are again used to provide facilities such as traffic measurement, alarm and fault supervision, call simulation, and ATME measurements.

Semi-Electronic Exchanges
The latest project to be undertaken by SEP is the development of a microprocessor-based system to control telephone exchanges which use the ITT miniswitch – a standard component in Metaconta exchanges and Minimat PABXs for more than 10 years.

Export Activities

Up to now SEP has concentrated on exports to the huge USSR market. These projects are developed, produced, and installed by SEP personnel. All such projects require careful study of the network specifications, and close cooperation with the Soviet PTT and its local organizations.

Know-How Transfer

Within the multinational ITT corporation, know-how can be transferred in two directions and in two ways. The first, technology transfer, involves the transfer of equipment designs and programs. The second, human resource transfer, is based on seconding engineers from different ITT companies to work on specific projects. The main aim is to avoid unnecessary duplication of development work within the corporation. Also, it is desirable that engineers should work on projects in establishments where the most comprehensive development facilities are available so that the best results can be achieved. At present engineers from SEP are working in ITT houses in Germany, Austria, and Belgium.

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Central Office Maintenance System

A new microprocessor-based modular testing and maintenance system has been developed for all types of telephone exchange. The central office maintenance system can work either independently or in conjunction with an ITT 1290 operations and maintenance center.

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Introduction

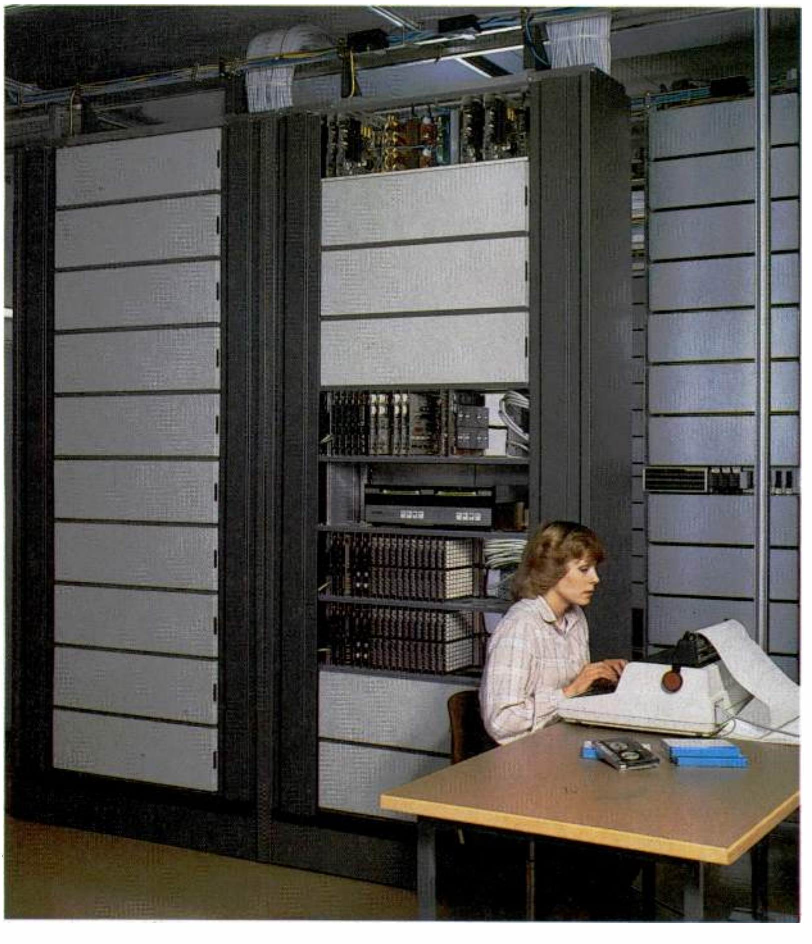
The continuous growth of telephone networks has resulted in an increasing need for telephone administrations to reorganize their operations and maintenance methods. New generation digital or analog SPC (stored program control) exchanges can be maintained via a remote terminal or from a maintenance center computer. At the same time, a large number of older exchanges are in operation, and in many countries will remain in operation for several decades. Wherever possible these exchanges should also be monitored centrally.

It is uneconomic for administrations to undertake the additional maintenance work required by continuous network growth simply by increasing the number of maintenance personnel. Instead, new methods must be found to overcome the dual problems of operations and maintenance in today's mixed telephone networks.

The method that is now being generally introduced is to install centralized systems for monitoring the operation of a wide variety of exchanges. This is possible as a result of the reliability of modern exchanges which require relatively little maintenance and therefore do not justify on-site maintenance staff. Centralized monitoring systems allow repair staff to cover a large number of exchanges.

The availability of inexpensive microprocessors and memory has made it possible to realize a reliable, automatic centralized maintenance system that operates fast enough to enable maintenance tasks to be carried out almost in real-time.

Central office maintenance system installed in a crossbar exchange.



Design Objectives for COMS

The central office maintenance system (COMS) can either work independently in a telephone exchange, or can be expanded to work as a terminal system in a centralized network maintenance system. The introduction of COMS into a telephone network offers several advantages:

- manual testing is minimized
- remote supervision is possible

Table 1 — Direct and indirect maintenance aids

Maintenance aid	Nature
Dimensioning	Direct
Network routing arrangements	Indirect
Extensions	Indirect
Adjustments	Indirect
Cleaning of devices	Indirect
Fault indications	Direct
Fault location	Direct
Fault repair	Indirect
Subscriber network maintenance	Indirect
Junction network maintenance	Direct
Carrier system maintenance	Direct
Power equipment maintenance	Indirect

- maintenance staff need not be permanently located at exchanges
- fault detection is fast and accurate
- comprehensive up-to-date traffic data is available
- updated information is available for network planning.

The central office maintenance system also enables the level of maintenance operations in conventional exchanges to be brought up to the standards of newer SPC systems, by providing both direct and indirect aid for the maintenance operations shown in Table 1.

The equipment installed in an exchange (Figure 1) consists of various functional hardware and program modules which can be combined to meet the varying requirements of different telephone administrations. In a typical application, the basic features that the system provides are:

- grade-of-service measurement using call simulation
- trunk testing
- ATME (automatic transmission measuring equipment) measurements
- fault recording and alarm indication
- traffic measurements and holding time supervision.

Some of these features, such as call simulation and traffic measurement, are, in principle, independent of the switching system and are specified by administrations. Other features may require adaptation to a particular exchange type.

The system also provides a number of optional features. Start up commands may be sent to existing test equipment over one or more control wires. Depending on the test equipment, results can be printed out on an existing test printer. Alarms can be supervised or results read over one or more wires. In addition, the system can provide a printed record of the duration of and charges for selected subscribers (e.g. following a subscriber complaint about billing). It also checks that the metering rate is correct for dialed calls. These operations are generally specific to each exchange type.

Table 2 summarizes the main technical characteristics of the central office maintenance system.

System Layout and Coverage

The total network structure of the central office maintenance system is shown in Figure 2. Starting at the lowest level it is as follows:

- Small (rural) local exchanges include devices for monitoring alarms and transmitting them to terminal systems in larger exchanges.
- Medium to large local and toll exchanges incorporate a terminal system which provides a selected set of maintenance facilities.

Figure 1
Structure of the central office maintenance system.

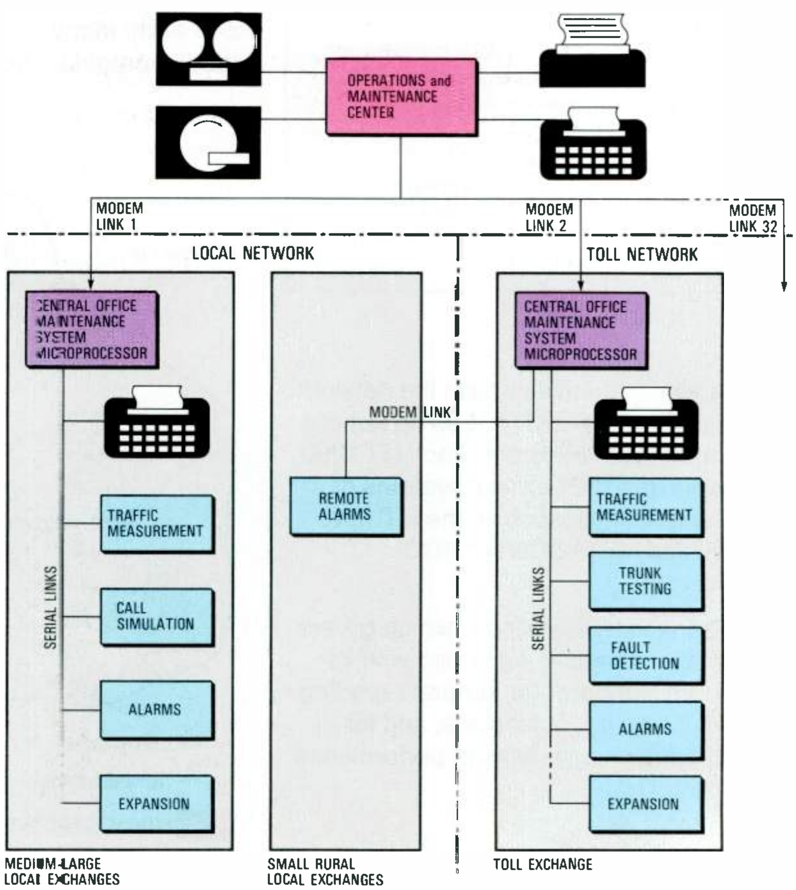


Table 2 — Main characteristics of the central office maintenance system

Terminal system

Call simulation
- 10 call generation-answering circuit pairs
- access to each 100-group

Trunk testing
- 3200 to 9600 test accesses depending on junctor type (i.e. 3 to 10 wires)

ATME
- CCITT ATME No 2 noise and attenuation measurements

Fault and alarm observation
- test point processors for 1024 points each

Traffic measurements
- 16000 test points in each terminal system
- measurements for routes and devices; intensity congestion and busy hour
- holding time supervision

Operations and Maintenance Center

Exchange links
- maximum 32
- terminal systems in conventional exchanges
- ITT 1240, Metaconta 10C
- digital and analog SPC exchanges

Workstations
- maximum 16 local or remote visual display units or teleprinters
- local or modem interface

Processor
- PDP 11 with 256 kbyte central processing unit
- console, 30 Mbyte disk, magnetic tape, line printer, and asynchronous multiplexer

Link specification
- CCITT V.24/V.28 interface operating at 1200 bauds
- ASC II format
- asynchronous, message type.

- Each administrative zone in the network is equipped with an ITT 1290 operations and maintenance center. Each ITT 1290 controls up to 32 terminal systems or SPC exchanges, such as the ITT 1240 digital and METACONTA* 10C exchanges.
- The administrative center, which covers all smaller zones, is equipped with its own minicomputer for general reporting of results on a regular basis, and for monitoring overall network performance.

* A trademark of ITT System

Central Office Maintenance System Terminals

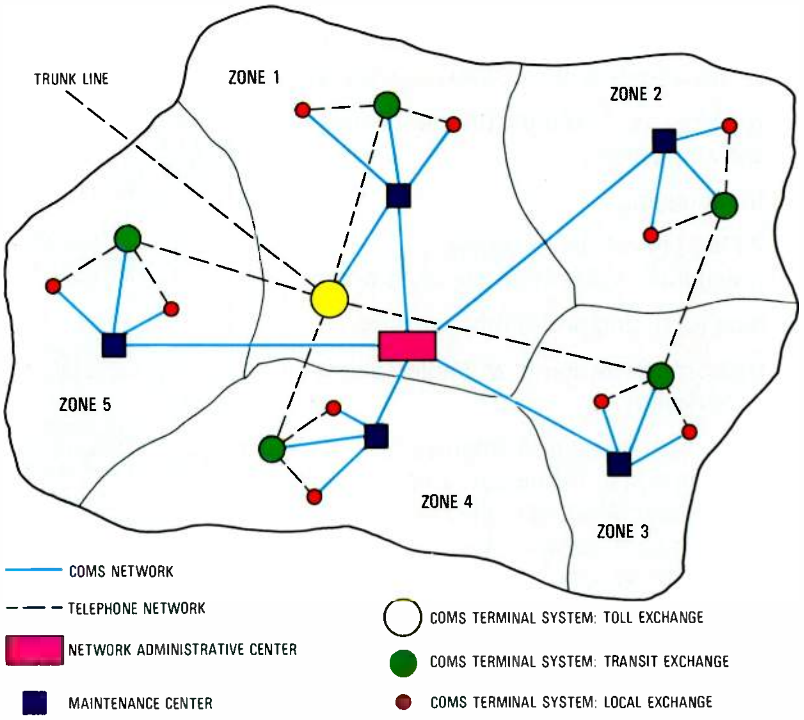
The terminal system consists of microprocessor controlled units which are configured to meet the operating company's requirements. The interfaces are designed for each type of telephone exchange in the network. These terminal systems are connected via a modem link to the ITT 1290 maintenance center. Maintenance operations can be initiated either locally from the terminal system's teleprinter or visual display unit, or remotely by the maintenance center minicomputer.

Each unit in the terminal system consists of a microprocessor part and an exchange interface. The exchange interface is used for input-output operations. A typical interface in the central office maintenance system is the test point scanning matrix; a microprocessor is installed together with the test point interfaces in a standard UNISWEP* subrack. The matrix can control 1024 test or measurement points. In some applications the same microprocessor is used to control as many as 4096 test points.

Additional terminal system units are a manual trunk testing controller, VDU terminal, and an integral modem.

As mentioned earlier, facilities provided by a terminal system include call simulation, trunk testing, ATME measurements, fault and alarm monitoring, and traffic measurements. The general tasks of call

Figure 2
Network structure of the central office maintenance system.



simulation, trunk testing, and monitoring are performed to detect and locate faults in the switching equipment. Traffic measurement results are used for network planning. ATME and other signal-level measurements provide information on the quality of transmission lines in the network.

Call Simulation
Call simulators are usually provided in terminal systems installed in local exchanges. Answering devices are equipped in all exchanges at which test calls will terminate. Tests are carried out by running preprogrammed test cases.

Trunk Testing
Trunk testing devices are connected to the outgoing relay sets through crossbar access switches. Access for up to 9600 trunk circuits can be controlled by the system. Interfaces and test programs are developed for each application. Trunks are tested by setting up test connections to answering devices in adjoining exchanges using the correct signaling sequences and signal levels. This allows the signaling sequence to be checked and the line to be tested to ensure that attenuation is within the specified limits. ATME tests of transmission characteristics are carried out on the outgoing side through the same access switches. On the incoming side, ATME interfaces directly with the selector stages. The ATME unit carries out noise and attenuation measurements in accordance with CCITT ATME No 2.

Fault and Alarm Observation
These supervisory operations are performed by the test point matrix microprocessor either in conjunction with traffic measurements or separately in dedicated test point units.

Functional supervision, which is performed during traffic measurements, detects abnormal holding times, observes constantly or never busied devices, and performs time congestion counts which show the percentage of time during which all devices within a bundle or group are seized.

Fault observation can, in principle, be divided between statistical counts of congestion and seizure, and the supervision of marker or register faults.

The equipment detects various alarms from exchange and transmission equipment, as well as from fire and open-door detectors.

Processor and peripheral devices in an ITT 1290 operations and maintenance center.

Traffic Measurements
A complete set of traffic measurement programs is available, allowing system operation to be configured to meet the operating company's requirements.

Measurements are carried out by the test point processor units, each of which deals with 4096 test points. The scanning sequence for each test point takes 3.6 s. The system provides a group measurement report on busy hour traffic, including the total traffic intensity for each group; average traffic, holding time, and time congestion per device; and the number of constantly or never busied devices in a group. Printouts of unsymmetrical loading and abnormal holding times are also available. The former report on devices within a group that have an abnormal traffic level (e.g. a route); the latter indicate that there are deviations in the holding times compared with other devices in the group.

All microprocessor units in the terminal system communicate over serial links with the central office maintenance system master unit which thus acts as a concentrator for all system functions. Remote alarms from smaller exchanges may be centralized at this master unit. In a typical application the master unit has four main tasks: loading semipermanent data into other microprocessors, man-machine communication, automatic supervision of the functions performed by other microprocessors, and alarm supervision.

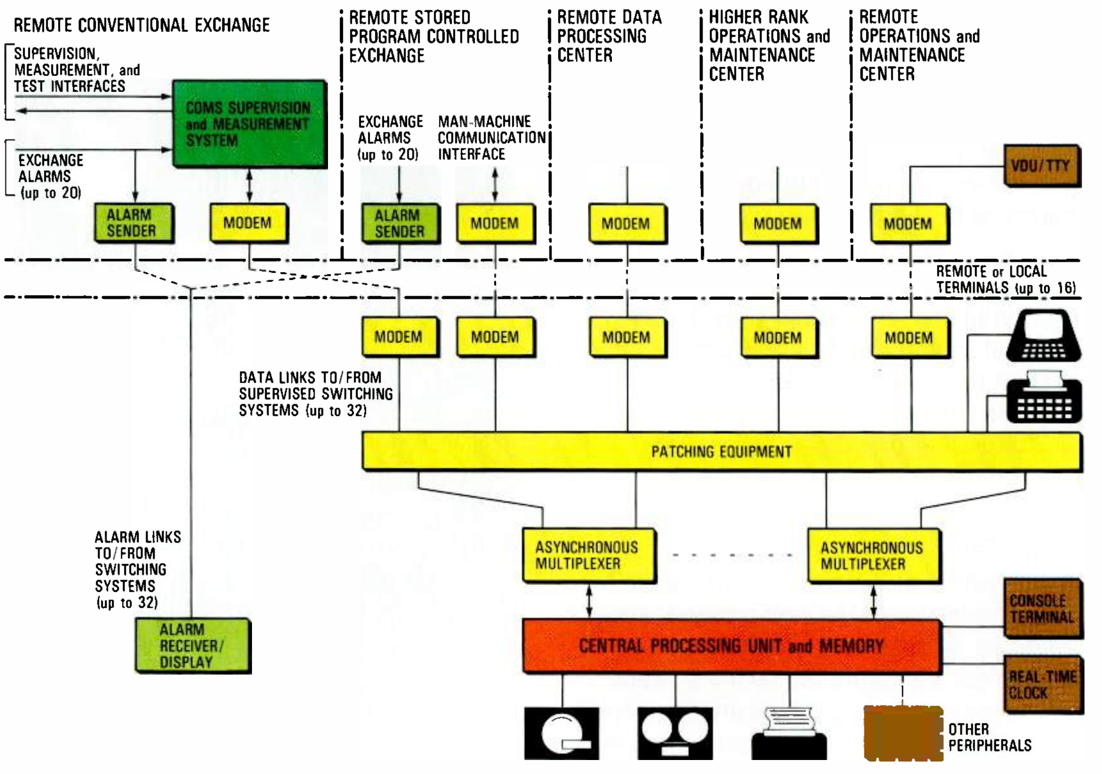


Figure 3
Structure of an ITT 1290 operations and maintenance center.

Operations and Maintenance Center

Terminal systems can communicate with an ITT 1290 operations and maintenance center (see Figure 3). This processor-based system, which was designed for the ITT 1240 digital and Metaconta exchanges, can control up to 32 links connected to either COMS terminal systems or SPC telephone exchanges.

For applications involving the central office maintenance system, the ITT 1290 processor performs database management functions and controls operations both within the center and in the terminal systems.

Database Management

System operation requires certain data to be kept in the ITT 1290 database, including the terminal system configuration, interface tables for exchange connections, and network data (e.g. numbering plan, routing tables).

Management of this database requires facilities for inputting and making changes to data, outputting existing data, securing data and carrying out reloads, and transferring new configuration and test data to the terminal systems.

Operations Control

The ITT 1290 has built-in test facilities for the detection of equipment, program, and functional faults. All operational data is secured to ensure high system reliability.

The man-machine language uses abbreviated "English-like" commands. Using this language, it is possible to specify how different messages are treated by the ITT 1290 in order to schedule the sending of messages, to transfer common messages to some or all system ports, and to specify the ports used by the ITT 1290. The central office maintenance system also requires statistical application programs for the different maintenance function reports.

Terminal System Operational Control

The ITT 1290 operations and maintenance center monitors faults in the central office maintenance terminal systems in the same way as it detects faults in the center itself.

The ITT 1290 processor controls many functions for the wide range of operations offered by the system. It sends test start-up times to the terminal system units in accordance with specified schedules or provides an immediate command to start testing. In the reload state, the ITT 1290 is able to restart the terminal system units, then run an initialization test and reload the semipermanent data.

Messages from terminal systems to the ITT 1290 operations and maintenance center are treated in one of three ways:

Scheduled message output: messages can be specified as belonging to a certain type which are printed out either hourly, daily, or weekly. The corresponding files are a main message file and a long-term retention file.

It is possible to print out the contents of a file should it overflow as a result of excessive numbers of messages being received from the terminals.

Selected message output: on demand, messages recorded during a particular period (half hour of the previous day or a certain day of the previous week) can be printed out. Messages stored on magnetic tape can also be printed out in this way.

Direct message output: messages may be directed to the required workstation or printer.

The treatment of different types of message can be specified so that messages are either sent to the printer, recorded on tape, or forwarded to the workstations. Information recorded on magnetic tape may also be output to workstations.

Use of Personnel

Together, the central office maintenance system and the ITT 1290 operations and maintenance center allow a telephone administration to make effective use of maintenance staff. Trained engineering staff can be located at the operations and maintenance center; repair teams can then be directed to an exchange where a fault has been detected.

Conclusions

The first central office maintenance systems have been installed in A-204 crossbar exchanges in Finland. During this period of operation in the field the systems have proved that they can meet the reliability requirements that were set during the design phase. Improved use of all switching devices has been achieved, and correct dimensioning of the network ensures savings for the administration.

Pekka V. Heikkinen was born in 1947 in Finland. He graduated in electronics from Helsinki University of Technology in 1972. Subsequently he was employed by the Finnish PTT, and in 1973 joined SEP. From 1974 to 1976 Mr Heikkinen worked as a design engineer on the Metaconta 11A project at CGCT in Paris. On returning to SEP he was appointed technical manager, and in 1981 became deputy technical director.

Markku A. Nikkola was born in 1949 in Finland. He received an MSc in communication engineering at Helsinki University of Technology in 1972. After graduating he worked in Finland for Siemens, the Helsinki Telephone Company, and Burroughs. In 1977 he joined Plessey Telecommunications in the UK where he worked as a team leader in the systems group of British Telecom's System X project. Since 1980 Mr Nikkola has been with SEP in Finland, first as export project manager and recently as sales manager for public switching products.

Standard Electrik Hellas SAIC

The company's presence in Greece dates back to 1948 when it was a supplier of telecommunication equipment to both the public and private sectors. Initially this function was performed by a private agency which represented all ITT System houses in Greece. Subsequently, the agency was taken over by ITT in 1965 when it decided to establish ITT Hellas both to represent ITT companies and to manufacture industrial and consumer telecommunication equipment for the Greek market and export.

Up to 1972 the unit produced telephone exchanges, telephone subsets, and private communication equipment. However, the economic recession of 1973 to 1977 made it necessary for ITT Hellas to diversify its activities, find new export markets, and introduce new products.

In 1977 the new name Standard Electrik Hellas (SEH) was adopted under the auspices and with the close cooperation of Standard Elektrik Lorenz, ITT's associate in the Federal Republic of Germany. As a result, SEH gained assistance and know-how that enabled the company to expand its manufacturing capabilities and to increase its exports to West Germany.

SEH intensified its efforts to identify new market opportunities. As a result of studies conducted by the marketing department, SEH added new products to its production programme, including color television sets, television games, loudspeaker systems, deflection coils, contact spring sets for switching equipment, and cable forms. In 1980, SEH entered the military business by supplying the Greek Ministry of Defense with VHF/FM transceivers. The company also produces military products to NATO's Allied Quality Assurance Publication standards.

Parallel to its marketing activities and efforts to increase local production and the volume of resales of ITT System products, the technical department was active in developing and designing new products to meet specific local market needs and requirements. One result was the development of a small, low cost electronic PABX suitable for smaller businesses. Furthermore, SEH is currently developing a new telephone subset which is scheduled to be introduced in 1984; this is being designed to meet the needs of the Greek market.

Today SEH is one of the leading companies in Greece, manufacturing and supplying telecommunication and electronic systems equipment and components to the telecommunication administration, commercial and industrial business, as well as to government organizations and export markets. Over the past 10 years, the company's domestic and export sales have exceeded 3 billion drachmas. Between 1980 and 1982, the company's sales increased by nearly 250%.

SEH's success in the telecommunication market is based on the experience gained from its association with other ITT companies. This know-how benefits SEH and ensures that modern technology is used in the company's products, which is an advantage for Greek industry in general. Indeed, it is essential to the application of electronic telecommunication technology in Greece.

Today SEH employs over 300 people at its Amaroussion premises in Greece.

Production

As one of the main suppliers to the Greek telephone administration, SEH has manufactured over 300000 switching lines (both PENTACONTA* and crossbar systems), over 1.2 million telephone subsets, considerable quantities of key systems, and military radio sets both in manpack and mobile versions. A significant part of the company's business is in the export market, including the production of telecommunication equipment for western European countries (Belgium, France, Switzerland, Germany) and the Middle East.

Quality and Training

To be in line with the ITT high quality product specifications, SEH has invested considerably in training programmes. Almost all personnel have undertaken training of some kind, and are continuing to do so as sophisticated new technologies are introduced.

A zero defect programme exists in SEH where all products are strictly under the control of the ITT quality standards.

Future Prospects

SEH is entering into new fields of advanced technologies, and introducing new products such as transmission equipment, electronic line concentrators, facsimile equipment, data terminals, and printers. Above all, the ITT 1240 Digital Exchange – the world's most sophisticated digital switching system which has been ordered by progressive telephone administrations worldwide – is at present being evaluated in Greece.

J. S. Papantoniou
General Manager
Standard Electrik Hellas
Athens, Greece

* A trademark of ITT System

Small Electronic PABX

In the past, only large PABXs have offered a comprehensive range of subscriber features. A new PABX for smaller businesses utilizes modular programs to provide a wide range of facilities which can be expanded by introducing new modules.

A. B. Papadopoulos
Standard Electrik Hellas SAIC, Athens, Greece

Introduction

Since microprocessors were first designed and made available just a few years ago, business communication has entered a new era with continuing advances in component technology making possible the development of sophisticated PABXs that offer a wide range of features to meet ever increasing user requirements. The rapid evolution of such systems has created a more selective and competitive market.

To meet these needs, SEH (Standard Electrik Hellas) has introduced a low cost electronic PABX (private automatic branch exchange) based on microprocessor technology which offers subscribers excellent performance, is simple to operate, and easy to install and maintain.

Applied Technology and Main Components

The major impact of LSI (large scale integration) technology over the past few years has resulted from the introduction of low cost MOS devices which now dominate the microcomputer market. MOS LSI was therefore chosen as the technology for SEH's low cost PABX, the main components of which are:

- Z80 microprocessor; this compares favorably in both program and hardware capabilities to any other 8-bit microprocessor and is fully compatible with the popular 8080A CPU.
- Memory. Three different types of memory are used: EPROM for storing the control

Low cost PABX and operator console.

program; RAM for storage of variable data; and PROM for the address decoder. Provision has also been made for the use of optional CMOS RAM with battery back up.

- Switching matrix of the single 8-channel analog multiplexer, space division type. This has three binary control inputs which select one of eight channels to be turned *ON* and connect the input to the output.
- Other components include standard integrated circuits and discrete components including transistors, diodes, and optocouplers.

The hardware components are mounted on plug-in printed boards of double eurocard size fitted with 64-pin euroconnectors.

Technical Characteristics

The new PABX utilizes an expandable modular construction, providing from one to six public exchange lines and from two to 18 extensions. The hardware modularity allows the ratio between the number of internal and external lines to be varied as required.

Space division multiplex switching is used; each public exchange line and extension board has its own dedicated switching matrix which operates under the control of a microprocessor-based controller.

The PABX is very compact, with a width of 290 mm, a depth of 210 mm, and a height of 220 mm.

System Configuration

Hardware

The hardware configuration (Figure 1) is based on three types of printed board:

- central control unit (microprocessor and memory)
- public exchange line interface circuit board
- extension line interface circuit board.

Communication between the different system boards is carried out via common address and data buses, and also via various control lines on a fourth board (the mother board) which incorporates euroconnectors which the system boards plug into. The communication lines on the mother board are divided into the following groups:

Analog signal lines. These carry the audio signals from one board to the other, as well as the 400 Hz tone signal which is carried on a separate line reserved for this purpose.

Digital signal lines. These are divided into data lines, address lines, and control lines. They are used by the central processor unit to obtain information from and transmit instructions to the peripheral boards.

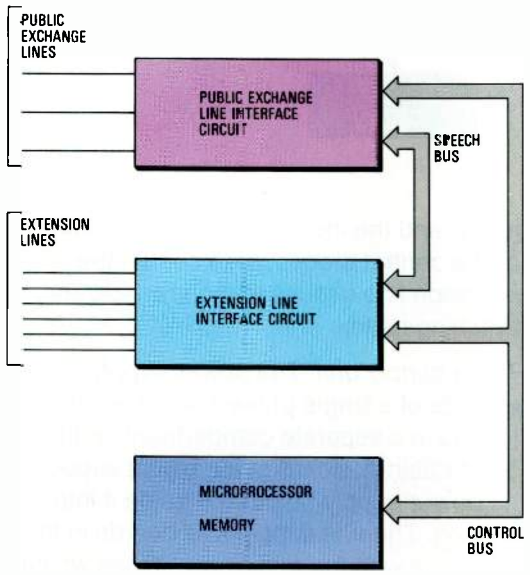


Figure 1
Hardware configuration of SEH's small electronic PABX.

Power supply lines. These are connected to the main power unit and supply the required system voltages to all boards.

Central control unit. The control unit consists of the Z 80 microprocessor, various memory chips, and ancillary components, as well as an output for driving the loudspeaker which is used as a central tone caller.

Public exchange line interface circuit board. This board incorporates all the necessary hardware for the connection of one to three external lines under the control of the central processor unit. It includes the public exchange line connecting circuits, the corresponding section of the switching matrix, and the input/output ports required for the central processor to control the public exchange line circuits and the associated switching matrix. A relay is provided on this board to connect the exchange lines to corresponding extension lines in the event of a mains power cut.

Extension line interface circuit board. This board incorporates all the hardware for the connection of six internal lines under the control of the central processor. It includes the extension line connecting circuits, the corresponding section of the switching

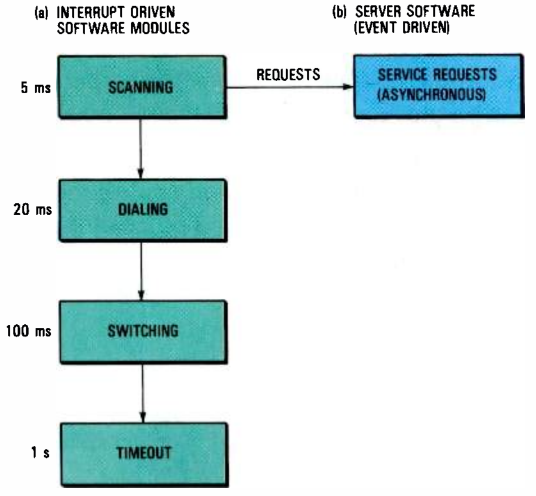


Figure 2 Program configuration.

matrix, and the input/output ports required for the central processor to control the extension line circuits and corresponding switching matrix.

Power supply unit. The power supply consists of a single printed board which is housed in a separate compartment in the PABX cabinet. Board guide plates support the power supply board and guide it into position. The unit supplies all boards in the exchange with the required voltages which are generated directly from the mains supply through positive and negative integrated voltage regulators.

Subset line connections. All subset lines are carefully segregated and are terminated on printed circuit connections mounted on a board which again is housed in a separate compartment in the exchange cabinet.

Programs

Modular programs have been used (Figure 2) as this approach allows new features to be added without affecting the program structure. The implementation of new or revised features is effected by replacing the program (EPROM chip). The final program instructions are written in assembler language, but the operations are defined in a high level language. The basic program modules are:

Scanner which scans all input device lines to determine their status and compares present line status with previous line status. If the status has changed, the program requests the service flag and writes the new status in the input device record.

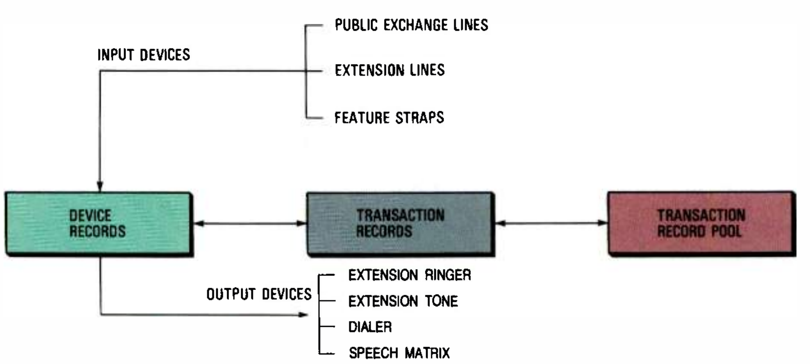
Server, which is activated by requests for service, checks the validity and feasibility of requests, services the requests, and sends all control outputs to the output queue.

Switching checks the output queue and directs the output to the appropriate device.

Timeout handler scans all active transaction records and decrements the timeout counters of those for which the timeout flag is set. If the counter is decremented to zero, the request for timeout flag in the input record is set.

The above routines operate independently of one another and are implemented concurrently. The data upon which the routines operate is of two basic categories, fixed and variable. Device records are fixed and are linked with transaction records from a free transaction record pool list, as required. Records which are not needed are returned to the pool. The flow of information is illustrated in Figure 3.

Figure 3 Flow of information.



Operator Console

The system is supplied with a cost-effective operator console. An LED display shows the basic status of the public exchange and extension lines.

System Features

The system offers a full range of standard features, including the following:

- use of any type of standard subset – rotary or quickstep – with two-wire connection to each subset
- call transfer with or without announcement
- internal or external enquiry
- conference calls
- call queuing
- automatic redial of last number
- call barring
- selective access to trunks

PABX with the exchange line, extension line, and central processor boards removed.

– programmable night service
– call return to operator
– seizure of incoming call by dialing one digit
– power failure transfer.

Additional optional features may also be implemented by adding further program modules.

Conclusions

The new electronic PABX developed by SEH uses solid state technology to provide users in smaller businesses with the wide range of features generally associated with large PABXs. Additional features can be provided simply by adding new program modules. The simplicity of the design ensures that installation is straightforward and maintenance is minimized.

A. B. Papadopoulos was born in Greece in 1930. In 1958 he was awarded a diploma of advanced technology in electrical engineering by Northampton College of Advanced Technology, London. In 1963 he received a BSc in electrical engineering at London University. After several years R&D in automatic control of cable factories, Mr Papadopoulos joined STC to work on the development of subscriber apparatus. In 1973 he transferred to SEH Athens where he is now technical manager.

Electronic Tone Caller

Identification of a ringing subset can be difficult in environments where several are sited close together. A new electronic tone caller enables the ringing tone to be adjusted by the subscriber to provide a pleasant, recognizable sound.

A. B. Papadopoulos
Standard Electrik Hellas SAIC, Athens, Greece

Introduction

The extensive use of telephone sets, particularly in open spaces or large offices where several sets are installed close together, has created the need for a distinguishable calling sound, and a more pleasant one than that of a conventional magneto bell operated directly by the exchange current. To meet this need, an electronic tone caller has been developed by Standard Electrik Hellas for use in standard telephone subsets.

Technical Description

The SEH 41981 electronic tone caller is based on an integrated circuit that operates from low frequency ringing currents and produces variable higher frequency dual-tone signals from an electro-acoustic transducer. The frequency of these signals can be adjusted by the subscriber using a variable RC network, external to the integrated circuit, to produce individual calling tones. This is achieved by a rotary switch; the slotted end of the shaft can be turned by a coin to select the preferred setting. A volume control allows the volume to be adjusted.

The electrical circuit of the tone caller may utilize the telephone handset receiver as the output transducer to reduce costs (Figure 1). The remote possibility of acoustic shock to the subscriber should a calling tone be emitted at the moment the subscriber depresses the hook switch while holding the handset near to his ear, can be avoided by incorporating a thermally sensitive resistor with a negative temperature coefficient of resistance, or alternatively by applying feedback from the output to a control RC circuit of suitable time constant at the input. The thermal characteristic of the thermistor or the time constant of the control circuit are chosen so that the acoustic output of the tone caller increases gradually from an initial low value to its full volume, thus providing a controlled acoustic build up.

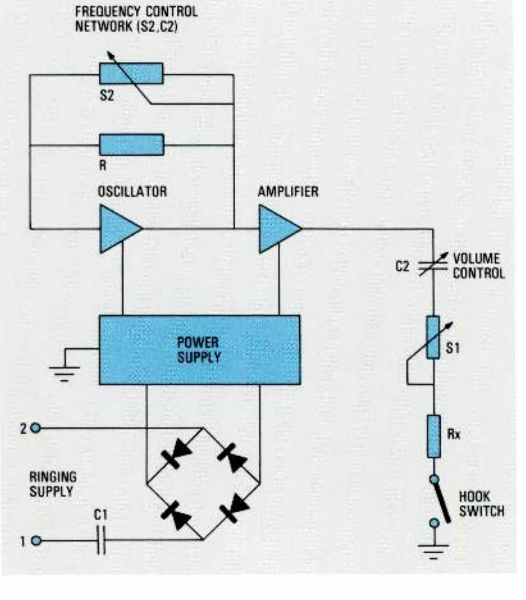
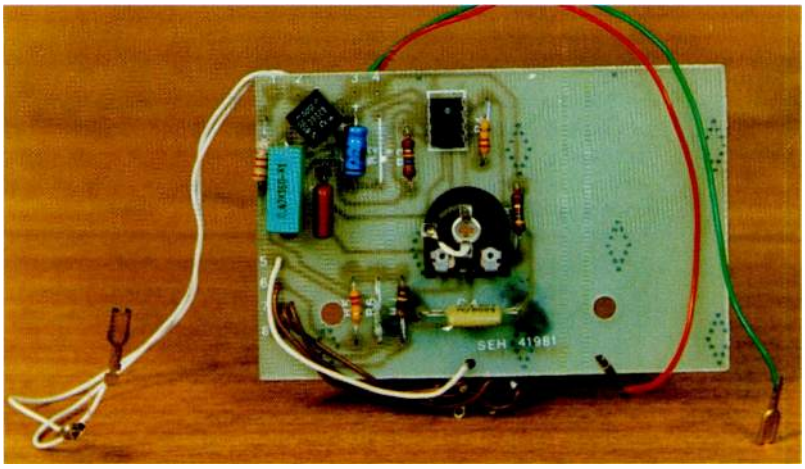


Figure 1
Block schematic of the SEH 41981 tone caller circuit.

Basic Circuit

The basic circuit of the tone caller is shown in Figure 1 which illustrates the frequency control of the tone caller output by a variable network (C2, S2) of resistors, capacitors, or a combination of both. As indicated earlier, this network can be switched selectively by the subscriber to vary the tone and frequency of the signal. Rx is the telephone handset receiver which is connected to the output of the tone caller when the handset rests on the hook switch; operation of the hook switch when the handset is lifted



Adjustable tone caller developed by SEH for use in telephone subsets.

disconnects the telephone receiver from the tone caller and connects it to the telephone speech circuit.

Conclusions

A cost effective electronic tone caller for use in telephone subsets has been developed by Standard Electrik Hellas. The tone caller allows the subscriber to alter at will the calling tones of his subset, thus providing a preferred and identifiable calling sound where a number of subsets are installed close together.

A. B. Papadopoulos was born in Greece in 1930. In 1958 he was awarded a diploma of advanced technology in electrical engineering by Northampton College of Advanced Technology, London. In 1963 he received a BSc in electrical engineering at London University. After several years R&D in automatic control of cable factories, Mr Papadopoulos joined STC to work on the development of subscriber apparatus. In 1973 he transferred to SEH Athens where he is now technical manager.

This Issue in Brief

Bjørløw-Larsen, K.
Submarine Power Cables
Electrical Communication (1983), volume 58, no 2, pp 150–154

The generation of electric power in Norway has grown considerably over the past 40 years. As a result, and because of Norway's very long coastline with a vast number of fjords and islands along the coast, a great demand has arisen for submarine power cables. To date, about 1100 submarine power cables had been commissioned, several of which are lying at considerable depths – the maximum so far being 670 m. The use of submarine power cables in Norway is unique on a world scale and has given STK a position as one of the leading manufacturers in this field.

Sletten, E.
ITT 5500 Business Communication System
Electrical Communication (1983), volume 58, no 2, pp 155–159

The ITT 5500 business communication system is based on digital switching and CEPT standard digital transmission. Modularity is the most salient feature of the system concept – a feature which opens up new approaches to solving problems of business communication. The author describes the basic system concepts and shows how system modules can be combined to build traditional centralized PABXs, distributed PABXs, and private communication networks. Other examples indicate how non-voice traffic is handled in the system, thereby providing users with integrated services communication.

Dietschi, R.; Gessler, Ch.; Staber, E.
Network Quality Tester
Electrical Communication (1983), volume 58, no 2, pp 162–168

Automatic telephone networks, in common with other types of equipment, require regular maintenance to ensure a high quality of service to the subscriber. Thus the quality of service must be continuously monitored using suitable test equipment that supplies reliable data about the network. The authors describe a network quality tester which has been specifically designed to perform this task rapidly and effectively. It can arrange test reports according to various criteria so that weak points in the telephone network can be detected and remedied early. The authors also describe how the test equipment can be used for troubleshooting, and the steps taken to ensure that testing cannot interfere with subscriber traffic.

Maurer, Ch. A.
Videotex System for Trial Service in Switzerland
Electrical Communication (1983), volume 58, no 2, pp 169–173

A pilot Videotex service has already been evaluated in Switzerland, but with the advent of a number of new service requirements it was decided to develop a much larger scale trial system, both to evaluate the equipment and to judge the reaction of information users and providers. The author describes the Videotex system developed by STR and the facilities available to users of the service. The trial will be based on two Videotex centers which are interconnected via the Swiss PTT's Telepac data network. Initially it will offer a telephone network interface for information users, as well as two interfaces for information providers; one for inputting information to the centers via the public telephone network and one for suppliers with external databanks accessing the system via the Telepac network.

Hedberg, T.
Flexible Data Circuit Terminating Equipment for Circuit-Switched Data Networks
Electrical Communication (1983), volume 58, no 2, pp 176–180

In 1979 SRT started the development and manufacture of a flexible DCE for the Nordic public data network. The requirements for a DCE for circuit-switched data networks differ from those for a conventional modem. Basic differences are the need for envelope formatting and simple yet complete integrated network supervision to achieve high reliability. The author describes the new DCE – a flexible modular equipment that can work with practically all subscriber lines and terminals. Three types of logic unit can be equipped to adapt the DCE to various terminals, and five types of modems/adapters are used for different lines and transmission speeds.

Jonsson, R. G.
Surveillance and Communication Receiver CR 91
Electrical Communication (1983), volume 58, no 2, pp 181–185

The CR 91 receiver has been designed to meet all the requirements for a versatile communication and surveillance receiver. The author describes how frequency generation is performed using a single loop synthesizer, and the way control signals are distributed over a serial data bus. Human factors played an important part in the front panel design, and resulted in the controls for the sweep and scan functions being located on a separate pad. Another important aspect discussed by the author is the facility for tailoring the sweep and scan functions to meet the user's requirements simply by utilizing different program modules.

Brood, R. J. A.; Buijs, F. M.
Unilink Signaling Converter
Electrical Communication (1983), volume 58, no 2, pp 188–192

Telephone networks in most countries are a mix of different exchange types as a result of the gradual evolution in exchange technology. As a result, many different types of interexchange signaling are used making it impossible for some exchanges to communicate directly. The authors describe a telecommunications service system – the Unilink signaling converter – which has been introduced to overcome this problem. When a Unilink converter is fitted in an exchange, both that exchange and the one with which it is communicating see mirror images of themselves via the connection with the converter. Although Unilink has been designed for the Netherlands network, it can easily be adapted for operation in other telephone networks worldwide.

Hoefsloot, J. J. C. M.; Steinberg, R. A.
Pentaphone II Private Automatic Branch Exchange
Electrical Communication (1983), volume 58, no 2, pp 193–197

Many private automatic branch exchanges offer too many lines or are too large for use in small businesses. The authors describe the Pentaphone II system which has been designed specifically for smaller business applications. This very compact private automatic branch exchange offers five internal telephone circuits and one or two public exchange lines. The use of a microprocessor controlled electronic crosspoint matrix means that it is reliable, silent in operation, and has a low power consumption. An optional DTMF receiver is available if pushbutton dialing is required.

Andersen, D.; Stridbaek, E.
Digital Subset for the ITT 5300 Business Communication System
Electrical Communication (1983), volume 58, no 2, pp 200–205

Based on the successful field trial of a digital subset in the town of Horsens, SEK decided to go ahead with further development of such subsets, initially for use with the company's ITT 5300 business communication system – an advanced digital PABX. The authors describe the main features of the DT 80 digital subset, its use with the ITT 5300, and the optional modules that can be used to support the increasing need for integrated data transmission. Finally the authors outline a further extension to the subset's data transmission facilities by integrating the ITT 5300 with a 16-bit microprocessor, offering users access to word processing, teletex, and mailbox facilities.

Lukaschek, K.
Videopult System for the Management of Railway Stations
Electrical Communication (1983), volume 58, no 2, pp 208–212

The Videopult system was developed as an alternative to the conventional command keyboard for railway signaling. It was first installed for field trials at Wolfurt station in Vorarlberg to test the concept of an integrated management system for railway stations. The author describes the design of the Videopult system, and shows how it has been integrated with existing railway system control equipment. A major feature of the system, interactive inputting of control functions using a light pen, proved to be an excellent technique. The field trial, which was successfully concluded after one year, has shown several ways in which the Videopult system can be enhanced to provide additional features, such as computer assisted reporting.

Theuretzbacher, N.
Enhanced CHILL Tasking Concept and Language for a Business Communication System
Electrical Communication (1983), volume 58, no 2, pp 213–217

The increasing complexity of third generation communication systems called for a new programming technology. This paper presents the results of a development project with the objective of introducing the CHILL programming language as a basis for such a new technology. To cover the specific requirements of this application area, the real-time multitasking capabilities of CHILL had to be enhanced by introducing means for supporting timeout handling, access to a database system, specific input/output operations, etc. The new features have been added by using the concept of operating system primitives, thereby avoiding extensions or changes to the basic CHILL syntax. The author also describes an optimizing CHILL compiler that supports the enhanced tasking concept, and is independent of the target computer.

Garcia Semov, J. L.; Diez Kowalski, E.
Mobile UHF Transceiver RF Unit
Electrical Communication (1983), volume 58, no 2, pp 220–224

The principal design objective for the transceiver RF unit was to develop a compact unit which would meet all the relevant CEPT recommendations. The authors discuss how this was achieved and describe the unit which utilizes modern analog and digital integrated circuits. Frequency generation and control are achieved with a combination of a wideband voltage controlled oscillator and a fully digital synthesizer. Data stored in programmable read-only memory is used for channel selection and to change the operation mode. All circuits are wideband and cover the entire 0.7 m band without the need for realignment after a frequency change, thus ensuring low servicing and maintenance costs.

Lafuente, L. M.
Adaptive Differential Pulse Code Modulation Coder for Low Bit Rate Transmission of Speech Signals
Electrical Communication (1983), volume 58, no 2, pp 225–229

Marconi Española has considerable experience in high security cryptophony for military applications. In this article the author describes an adaptive voice encoder with prediction which enables the signal-to-quantization-noise ratio to be markedly improved for a given transmission rate. This technique does not require especially complex circuits.

Heikkinen, P. V.; Nikkola, M. A.
Central Office Maintenance System
Electrical Communication (1983), volume 58, no 2, pp 232–237

Continuous expansion and modernization of telephone networks must be accompanied by a reorganization of operations and maintenance methods if the most effective use is to be made of maintenance staff. The authors describe a new central office maintenance system which can be used either independently in an exchange, or as a terminal system in conjunction with an ITT 1290 operations and maintenance center. Advantages of the system are automatic testing of many functions, remote supervision, fast and accurate fault detection, and the availability of comprehensive traffic data for such purposes as network planning.

Papadopoulos, A. B.
Small Electronic PABX
Electrical Communication (1983), volume 58, no 2, pp 240–243

In the past, only larger PABXs have been able to provide a full range of subscriber features. Now, however, the availability of inexpensive MOS LSI circuits means that it is feasible to include such features economically in much smaller PABXs. A new electronic PABX developed by Standard Electrik Hellas serves from two to 18 extensions and up to six exchange lines. A modular program structure has been used to provide a comprehensive range of facilities which can easily be extended by the introduction of new modules. Simplicity of the overall design ensures reliability and ease of maintenance.

Papadopoulos, A. B.
Electronic Tone Caller
Electrical Communication (1983), volume 58, no 2, pp 244–245

In open spaces or large offices where numerous telephone subsets are located close together, it can be difficult to distinguish which of them is actually ringing. A new adjustable tone caller overcomes this problem by allowing the subscriber to vary the ringing tone so that it is clearly distinguishable and pleasant to hear. If required, costs can be minimized by utilizing the handset receiver, in which case protection circuits are employed to prevent the user being subjected to acoustic shock in the unlikely case of an incoming call being received as the hook switch is depressed.

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