ELECTRICAL COMMUNICATION

ITT

VOLUME 38 • NUMBER 1 • 1963

ELECTRICAL COMMUNICATION

Technical Journal Published Quarterly by

INTERNATIONAL TELEPHONE and TELEGRAPH CORPORATION

320 Park Avenue, New York 22, New York

President: H. S. Geneen

Secretary: J. J. Navin

CONTENTS

Volume 38 **1963** **Number 1**

EDITOR, H. P. Westman **Subscription: $2.00 per year; 50¢ per copy**

This Issue in Brief

Application of Pulse-Code Modulation to an Integrated Telephone Network—Part 1 describes the advantages of applying pulse-code modulation to telephone networks. In pulse-code modulation, a speech wave is sampled periodically at a rate at least twice its highest frequency. The amplitude of each sample is defined by a number that is converted into a code group of pulses. After transmission as on-or-off pulses, they are reconverted to the corresponding amplitude values at the receiver.

Distortion in transmission occurs only if noise exceeds half the amplitude of a pulse in a position where no pulse should be or if more than half of the amplitude of a pulse is lost. Pulses are so brief that reflections from impedance mismatch are unimportant. Noise, crosstalk, and attenuation are independent of the number of repeaters and of the length of the transmission path.

Many trains of pulses, each for a different channel, can be interleaved to reduce the number of transmission paths. Transmission is practicable over ordinary telephone pairs in cable with repeaters at 2-kilometer (1.25-mile) intervals for 24 voice communication channels, using one pair for each direction in a 4-wire arrangement. The pairs for 2 conventional channels then accommodate 24 channels.

Initially, pulse-code modulation may be used to serve up to, say, 200 subscribers through a concentrator that may be connected to the exchange either by pulse-code or conventional connections. This will reduce greatly the number of space-path crosspoint switches as each crosspoint will be able to carry 24 channels.

A local exchange will serve a group of concentrators that need not be physically in the central office but each may be near its own group of subscribers. It will, of course, also provide paths to other exchanges. All connections will be over pulse-code multiplex highways. A transit exchange will operate on similar principles. There are only 2 basic switching stages in such a system, connections to the subscribers through concentrators and interconnections between pulse-code multiplex highways.

The timing of the interleaved series of pulses of each channel is critical for proper demodulation and switching. To reserve this particular time interval in all the exchange equipment traversed by that signal would require much duplication of equipment. Speech memories are therefore provided at the incoming and outgoing highways and are interconnected by a crosspoint, all being under control of switches that connect them at proper time intervals to permit the incoming signal to be stored in an incoming memory, transferred at a convenient time through the crosspoint to an outgoing memory, from which it is read at the proper time to transmit it to the next exchange or to the other subscriber.

Despite the flexibility of speech memories, blocking can still occur in the exchange and connections may be broken down and reconnected by a rearrangement method.

Pulse-code modulation uses signals similar to those for data transmission and will inherently provide wide frequency bands also suited to data transmission. Although signal distortion and losses of pulses are of no significance in telephony, some redundancy will be needed for adequate accuracy for data signals.

Part 2 is on transmission and encoding. Because in pulse-code modulation the amplitude, duration, and precise position of pulses are of no significance, the pulses can be regenerated at any point and transmission quality will be completely independent of transmission distance and of the number of repeaters. Consequently, the transmission-line equivalent remains absolutely constant regardless of variations in the transmission medium.

Use of digital techniques alone is adequate for all functions, such as modulation, demodulation, regeneration, control and supervisory operations, and speech transmission. Semiconductors fit readily into such a system. Speech transmission and switching information use the same kind of pulses and are handled over the same paths with about the same bandwidth requirements.

In coding, the amplitude of the sample is quantized to 1 of a prescribed number of levels. As the number of levels is increased, the uncertainty of exactly where the signal amplitude falls is decreased as each band represented by a code number becomes narrower.

To provide minimum and equal percentage widths for these levels as a function of the amplitude, nonlinear quantization is used. This is equivalent to amplitude compression followed by linear quantizing. The coder selected is completely digital in operation. It uses a compressor characteristic that can be reduced to a group of 5 segments of straight lines with different slopes producing 128 quantization steps.

In the decoder, the digits for each code group of pulses are accumulated in a digital memory and then decoded to give the resultant value of the sample.

Part 3 treats switching. There are only 2 switching stages required, one for connecting subscribers' lines to a pulse-code-modulation multiplex highway and the other for interconnecting such highways. The timing of the individual pulse trains to permit multiplex operation and the synchronization and coordination of these timing intervals for switching as a call passes through an exchange and from one exchange to another are important elements in the switching design.

For average traffic, a concentrator will serve 150 subscribers' lines and for light traffic 200 lines. It may be located near the subscribers and remote from the exchange to which it is connected. In an experimental system constructed to test the design, 7-digit code groups to designate the instantaneous amplitude of each sample of speech are transmitted at 10 kilohertz to provide 24 channels on a 4-wire basis using 2 telephone pairs for each conversation.

The subscribers' lines are connected to a concentrator, which assigns to each of 24 active lines a time interval for multiplexing and pulse-code modulates the speech samples during 4-microsecond intervals repeated every 100 microseconds. The pulse-code multiplex output of the concentrator goes to the exchange via 2 telephone pairs, 1 for each direction, called a highway.

At the exchange, the highway from a concentrator terminates in 2 speech memories, 1 for each direction. These 2 memories each have 24 lines in which speech code groups may be written for the 24 conversations it must handle. A free line, equivalent to a multiplexing time interval, is assigned to each new call by a time-switching memory.

As the outgoing highway must use for each new call a multiplexing time interval that is free to the concentrator or exchange to which the call is directed, a similar pair of memories under control of another time-switching memory is provided. These 2 memories will then be interconnected to transfer their pulse groups to each other through an electronic crosspoint switch under control of a space-switching memory.

All 3 switching memories record in identically numbered memory lines the addresses of the memory lines in the 2 speech memories so as to connect these 2 addresses together through the crosspoint for a 4-microsecond interval each multiplex cycle. Thus, through these memories, a speech sample arriving at the exchange at a randomly assigned interval in the multiplex cycle is transferred to another memory for transmission elsewhere at another randomly assigned multiplex time interval.

Multiplex switching is controlled by a clock in each exchange and variations among exchange clocks or variations in propagation time over lines or through repeaters are corrected by introducing under control of a synchronization circuit the required number of flip-flop stages to delay a "fast" pulse train.

If an incoming multiplex highway offers a call to an exchange at a time interval that is not available on the outgoing multiplex highway to which it is directed, internal blocking, which is a characteristic of all time-division-switching systems, occurs. A rearrangement of existing connections can then be made to accommodate the new call.

Experimental Pulse-Code-Modulation Transmission for Local-Area Telephony—A field experiment was made of pulse-code modulation for interexchange junctions in Madrid on 18 telephone pairs of 0·64-millimetre wire in existing 900- and 600-pair cables over a distance of 6·2 kilometres with unattended repeaters installed at 1-kilometre spacings in manholes.

The 24-channel system provided 23 speech paths and 1 synchronizing channel with signalling facilities suited to the rotary switching in the exchanges. Speech sampling was at 8 kilohertz with 70 quantization levels and about 25 decibels of signal compression. A unit-disparity unit-distance code requires 7 binary digits for speech and an 8th digit was added for signalling, making $1 \cdot 536 \times 10^6$ digits per second. By further time division, the 8th pulse provides 4 signalling conditions.

A unit-disparity code limiting to 1 the difference in the number of marks and spaces gives each code group an average density of pulses and transitions that aids in extracting the 1·536-megahertz timing wave from the received signals, reduces crosstalk, and permits codes for levels near 0, which are most frequently encountered, to avoid extreme numbers of transitions. A unit-distance code, in which codes for adjacent levels differ by only 1 digit, reduces errors that may occur in a parallel coder if several digits are simultaneously in a state of indecision.

Each speech input, after filtering and amplitude limiting, is sampled for 2·6 microseconds every 125 microseconds to produce amplitude-modulated pulses that are then multiplexed. After compression, the amplitude of each pulse is quantized by an array of comparators biased to operate at different levels. Each comparator is designed to define 2 levels. Through a diode matrix, the 7 digits of each code group are produced in parallel by the output of the comparators. The receiver is complementary to the transmitter. It uses magnetic cores to store the serially received digits for parallel readout. Transistor amplifiers and pulse-shape regenerators are used.

The compressor uses the approximately logarithmic characteristic of diodes, which are temperature controlled for stability. The expander is given the complementary characteristic by using a compressor network in a reverse feedback path.

Speech quality was good at optimum volume, tolerable for a 20-to-25-decibel range, and intelligible for a range of 40 to 45 decibels. A moderate extension in volume range over that provided by the test equipment would be desirable. Digit rates of 1·5 megahertz are feasible over paper-insulated multi-pair telephone cable and repeater spacings may exceed 1 kilometre.

United Kingdom–Faroes–Iceland (Scotice) Submarine-Cable Telephone System—The South section between Gairloch, Scotland, and Torshavn, Faroe Islands, is 290 nautical miles (537 kilometres) and includes 10 submerged repeaters. The North section linking Torshavn and Vestmannaeyjar, Iceland, is 408 nautical miles (756 kilometres) and is equipped with 15 repeaters and an equaliser. The polythene coaxial cable is 0·46 inch (11·7 millimetres) in diameter and has a characteristic impedance of 52 ohms. In one direction in the band between 24 and 96 and in the other direction between 120 and 192 kilocycles per second, there are 24 telephone channels, a third of which may be used for voice-frequency telegraphy. Engineering communication uses the bands 18–24 and 192–198 kilocycles per second and supervisory signals occupy bands from 96 to 100 and from 192 to 200 kilocycles per second.

Each repeater consists of a parallel pair of 3-stage vacuum-tube amplifiers whose gain rises from 22 decibels at 24 kilocycles per second to 65 decibels at 196 kilocycles per second. The amplifier being used for both directions of transmission, the required stability margin of attenuation is provided by filters. Power is supplied equally from both terminals over the cable with a drop of 120 volts for each repeater. Repeaters are mounted in rigid cylindrical steel housings and the electrical connections passing through the bulkheads are tested at 5 tons per square inch (787 kilogrammes per square centimetre) for a month before being accepted for actual use.

All the cable for a link and its repeaters were connected together and stowed aboard ship. As the cable is laid, insertion-loss-versus-frequency measurements were made to the shore terminal to predict the required characteristics of the equaliser, which was assembled to these specifications just before being jointed to the cable and laid.

Shipboard-Adjustable Submerged Equaliser— Equalisers are required on long submarine telephone cables to adjust for failure of repeater gain and cable loss to compensate, say, within 3 decibels, for each other over the entire frequency range of transmission. The attenuation characteristic of the cable determines the design of the repeaters. In addition to design limitations, manufacturing tolerances for both cable and repeaters, and variations in cable characteristic resulting from laying and sea-bottom conditions affect equalisation.

Part of the equalisation can be predicted but part must be designed, built, and inserted in the cable on shipboard as a result of measurements made during the laying operations and extrapolated to the completed length. The laying of the cable must not be interrupted while the equaliser is completed, installed in the pressure housing, and jointed to the cable, which requires about 16 hours.

That part of the equaliser to compensate for the predictable errors is built and hermetically sealed on shore in an environment of "dairy" cleanliness. To insure the same quality and reliability for the part that must be added on shipboard, several units of each component of resistance, inductance, and capacitance are also hermetically sealed in cans on shore for assembly on shipboard. A wide range of canned components is carried although only a few cans are used for each equaliser.

Equalisers are mounted in the same pressure housings as are used for repeaters. Made of high-tensile steel, the main body is a cylindrical tube or casing closed by a bulkhead at each end. Each bulkhead is sealed by two *O* seals carried in grooves round its periphery. Petroleum jelly is forced into the space between the seals to a pressure comparable to that at sea bottom and when submerged is raised slightly above sea pressure by a pressure-intensifying arrangement of pistons of different diameters. This pressure can be measured as frequently as desired without loss of pressure or jelly and provides a ready means for checking the sealing of the housing before laying.

A lead gasket between the inner face of the bulkhead and a retaining shoulder on the casing provides a diffusion seal against possible entry of water vapour passing the *O* seals. This lead seal is pre-stressed to insure that the lead has flowed even though in shallow water the sea pressure would not be sufficient.

A cable centre conductor is moulded in polythene through a threaded hole in the bulkhead and the polythene also covers a ridged stem that extends the hole on the high-pressure side. Water pressure seals the polythene to the stem, and the threading in the entry hole prevents the pressure from extruding the cable into the apparatus compartment.

The cable is firmly anchored within the outer end of the housing and is secured against rotating in the housing. The anchorage, bulkhead, and apparatus frame may be removed and replaced aboard ship with a few simple tools, and the sealing can be verified to be adequate to withstand sea pressure before the housing is layed.

Variable Reactance Frequency Multipliers—A nonlinear reactance, although free from dissipative losses, accepts real power from a source and releases it at other frequencies. It thus appears as a power-consuming device to the source but, since no power is retained, the over-all power input equals zero.

In the case of a lossless nonlinear resistance, the maximum efficiency as a frequency doubler would be 25 percent, the remaining 75 percent would be converted to direct current. The lossless nonlinear reactance does not rectify and so releases its power at one or more harmonic frequencies; the efficiency of conversion to a single frequency can reach the theoretical limit of 100 percent.

The characteristics of a voltage-sensitive semiconductor capacitor are given and some features affecting efficiency, power, circuit complexity, and tuning, especially for large frequency multiplication over wide bands, are treated. An example given starts with a band of frequencies in the 500-megacycle-per-second region and multiplies them to an 8000-megacycle-per-second band.

Supporting theoretical data appears in an appendix on the frequency doubler, the simplest and most efficient of the varactor frequency multipliers.

Contributions to Studies of Overflow Traffic—Consideration is given first to a simple primary group of trunks overflowing to a secondary group provided only for that purpose. Both rigorous and approximate equations are given for the occupation probabilities of the overflow group.

Next the case of 2 primary groups overflowing into a common secondary group is considered for various conditions of offered traffic and number of lines but with the same mean and variation of overflowing traffic. Approximate equations are given for the blocking probabilities or losses in the traffic offered to each primary group. These results are compared with those obtained using the method of Wilkinson; only slight discrepancies are evident, the latter being pessimistic for high-congestion conditions of overflow and optimistic for low congestion values. Finally, the results are generalized for any number of primary groups.

Interregister Multifrequency-Code Signalling for Telephone Switching in Europe—A description is given of a multifrequency-code signalling system that has been accepted by many of the leading telephone administrations in Europe. Based on the simultaneous transmission of 2 out of 6 voice frequencies between 1380 and 1980 hertz in the forward direction and between 500 to 1140 hertz in the backward direction, combinations are provided for 15 codes in each direction. By use of a "shift" signal, the number of useful codes is increased. Tables of these codes are given and their application in local, toll, national, and international networks is listed in detail.

The availability of an independent frequency band in each direction permits the use of compelled signalling. The duration of a signal is not important. The forward signal, for instance, is maintained until a proper reply signal is returned to the source to confirm its receipt and the performance of the required action.

End-to-end signalling is envisioned. In it a register in the office nearest the subscriber, or in a toll center acting as a pivotal point for a number of local areas, maintains control of the switching through all later offices, providing only the information necessary at each switching point to advance the connection to the next office and finally supplying the last digits of the called number to the exchange to which the called subscriber is connected.

Definitions and functions of line signals and definitions of general terminology are given in two appendixes.

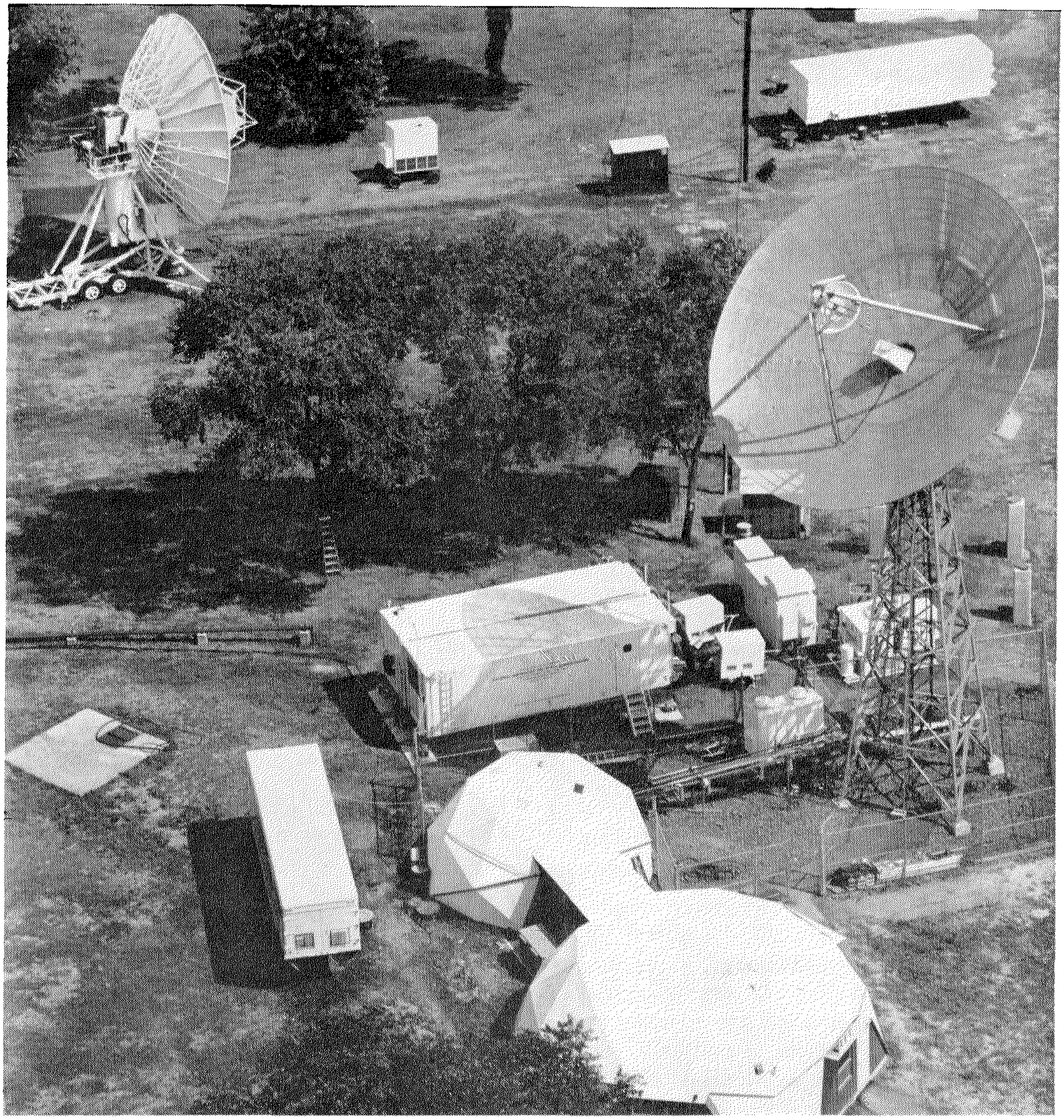


Figure 1—Project Relay satellite ground equipment. The permanent station in the foreground is at Nutley, New Jersey, and the transportable station in the background was sent to Rio de Janeiro for the linking of North and South America by radio via the satellite.

Project Relay—As part of its plans for Project Relay, the National Aeronautics and Space Administration provided for tests to link North and South America via the satellite over a 5000-mile (8000-kilometer) radio circuit.

In the foreground of Figure 1, is the permanent ground terminal near New York City that, built two years ago, is the first private station licensed by the Federal Communications Commission for earth-space communication experiments. It uses a 40-foot (12-meter) paraboloidal antenna. The terminal apparatus is housed in two igloo-type structures and includes a transmitter, tracking and communication receivers, antenna controls, monitoring consoles, and means of combining several voice channels into one signal for ease in handling.

In the background of the picture are the antenna and control van of a transportable station to be set up near Rio de Janeiro, Brazil, for operation by Companhia Radio Internacional do Brasil. This station travels in a van and three trailers and can be shipped by sea, air, or road. It can be erected by four men in as little as 16 hours. The 30-foot (9-meter) paraboloidal antenna can be dismantled into pie-shaped pieces. The station can handle 12 simultaneous two-way telephone conversations; 12 teleprinter or data circuits can replace each voice channel.

ITT Federal Laboratories
United States of America

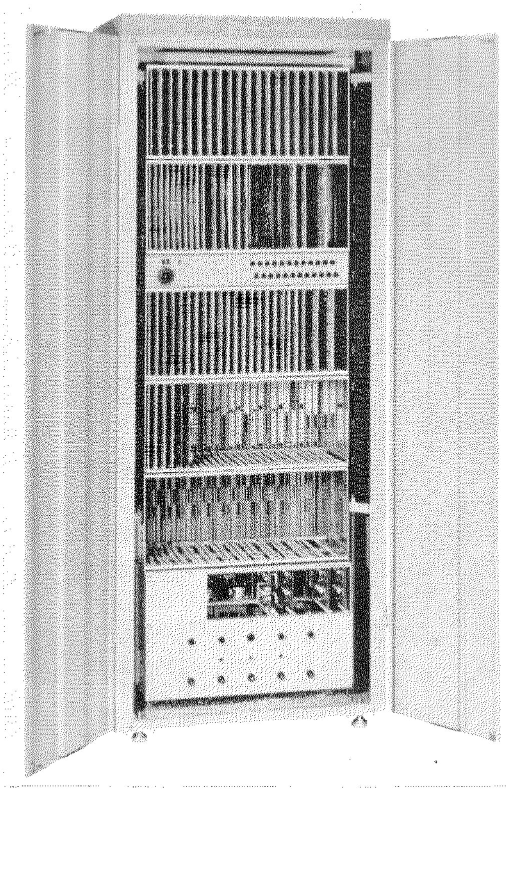
Electronic Telephone Switching with *PNPN* Diodes—A recently demonstrated 100-line private automatic telephone switching system provides a separate physical path for each two-way speech channel. The connections are established by a single equipment on a time-sharing basis.

When points on the input and output rows and columns of a 3-stage network of 4-layer *PNPN* semiconductor diodes are marked by potentials, a series of the diodes will fire sequentially across the network to form a conducting path between the marked points. A path will always be found between any two terminal points, but this will not always be the same path. Defective diodes will not disrupt traffic but will simply be avoided in setting up a path.

Lifting of the subscriber's handset will mark one side of the network and actuate a link allotter, causing a free link to mark the other side of the network. Diodes across the network then fire, connecting the subscriber to the link and initiating dial tone. Responding to the dialled digits, the link will mark the network input terminal of the called line. A second path will then fire through the network and connect the called line to the link and through it to the calling line. Ringing tone and then speech current are connected.

A printed-circuit board accommodates each 5 subscribers' line circuits, which give access to the link circuits. In the reverse direction, a link is connected to the line circuit of the called subscriber's line. The link circuit provides dial tone, stores the dialled digits, and establishes a route to the called line. It also provides ringing or busy tone. Fail-safe performance is provided by link circuits. The link allotter serves all calls on a one-at-a time basis. On failure, transfer to a stand-by unit is automatic.

Figure 2—A 100-line all-electronic private telephone exchange using *PNPN* semiconductor diodes for switching.



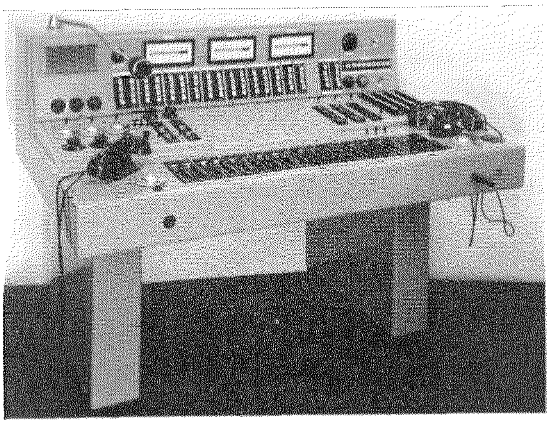
Maintenance may be by nontechnical personnel. An audible tone signals a malfunction and a lamp indicates the defective circuit board, which is simply withdrawn and a spare plugged in to replace it. A 100-line equipment is shown in Figure 2.

ITT Kellogg
United States of America

Stereophonic Studio Equipment—Two broadcast studios especially designed for stereophonic pickup have been installed for the Belgian Radio and Television service. They may be used for monaural operation also.

For stereophonic pickup, 6 input channels may be mixed and transmitted over the required 2 output channels. Each input channel has 2 connections for microphones and 7 for magnetophones, turntables, and other external sources. The attenuators for controlling and mixing inputs can be coupled in pairs. They are equipped with contacts for signalling purposes and to connect any of the additional inputs to monitoring loudspeakers via cue circuits. The output signals, at a level of 6 decibels referred to 1 milliwatt, are monitored with loudspeakers and light-spot modulation meters.

Figure 3—Belgium Radio and Television stereophonic broadcast studio equipment.



In addition to the required complement of amplifiers, power supplies, and control equipment, each studio is provided with 3 magnetophones and 3 turntables. One of the control consoles is shown in Figure 3.

Bell Telephone Manufacturing Company
Belgium

Telex Automatic Exchanges—Automatic teleprinter switching systems have been inaugurated in the New York and San Francisco offices of the American Cable & Radio Corporation. They permit its telex subscribers to set up international and intercontinental connections. All incoming traffic is handled on a fully automatic basis as is outgoing cable traffic. Outgoing radio traffic is on a semiautomatic basis.

Charges are recorded on perforated tape and reported to the calling subscriber by teleprinter after each call. The tape is also used for billing.

The New York exchange has a capacity of 1300 subscribers and 180 trunks connected to the most important switching centers of South America and Europe, to the domestic Western Union and telex systems, and to the Canadian telex network. The San Francisco installation will accommodate 200 subscribers and 37 trunks giving access to switching centers for Japan, Philippines, and Hawaii as well as to the domestic Western Union and telex systems. Both permit ordinary cablegrams to be exchanged via teleprinter.

Bell Telephone Manufacturing Company
Belgium

American Cable & Radio Corporation
United States of America

Telex Experiments Via Telstar—Following several successful exchanges between the United States and France via Telstar, the communication satellite of the American Telephone and Telegraph Company, the first direct connection between telex subscribers over this relay between London and New York occurred on 23 August 1962 and on the following day between Paris and New York.

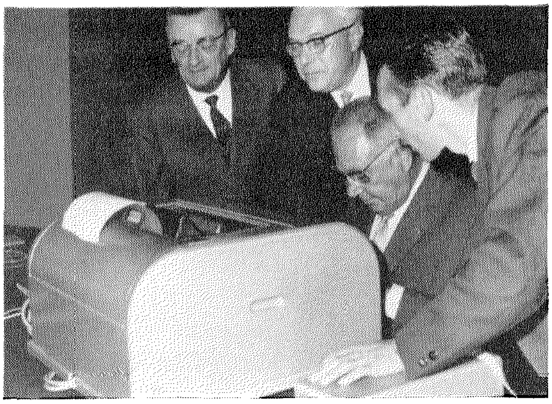


Figure 4—First experimental Telex connections between subscriber stations in New York and Europe via the Telstar communication satellite occurred on its 406th orbit of the earth. At left is the group in Croydon, England. From left to right are: Group Captain F. C. Daubney, Chairman of the Board and Managing Director of Creed & Company; Mr. Richard Thompson, Member of Parliament; Mr. D. L. B. Lithgow, General Commercial Manager of Creed; and Councillor John L. Aston, Mayor of Croydon. In Paris, at the right, also from left to right, are: Mr. E. Delpoux, Commercial Manager of Compagnie Générale de Constructions Téléphoniques; Mr. P. Lizon, General Manager of Le Matériel Téléphonique; and Mr. M. Pérès, President and Managing Director of Les Téléimprimeurs, at the keyboard.

Greetings were sent by Mr. H. S. Geneen, President of International Telephone and Telegraph Corporation, through the facilities of the American Cable & Radio Corporation in New York, and were received simultaneously in London by Standard Telephones and Cables and in Croydon by Creed & Company. On the following day, similar transmissions were made to Laboratoire Central de Télécommunications, Le Matériel Téléphonique, and Compagnie Générale de Constructions Téléphoniques in Paris. Suitable replies were returned and members of the international press also exchanged greetings via this new and promising radio relay.

In the United States, the Andover, Maine, Telstar terminal was used. The British General Post Office terminal was employed in England as was that of the Centre National d'Etudes des Télécommunications in France.

Northern Ireland Factory—Lord Brookeborough, prime minister of Northern Ireland, proclaimed "the beginnings of an important new industry" in Ulster when he welcomed the arrival of Standard Telephones and Cables (Northern Ireland) Limited to Monkstown with the opening of its first plant, shown in Figure 5. An additional main factory is under construction nearby and both will produce telephone exchange equipment.

Standard Telephones and Cables
United Kingdom

Pompes Salmson Joins Le Matériel Téléphonique—The Société des Pompes Salmson has recently merged with and will assume the name of Le Matériel Téléphonique.

Founded in 1917 as a family business, Société des Pompes Salmson has become a major supplier of centrifugal pumps in France. It manufactures a wide range of pumps for both industrial and domestic heating systems. A great variety of electric-driven pumps also go to the mechanical and chemical industries.

Its engineering, development, and test laboratories are at Argenteuil as are its service and repair shops. Manufacture is at a modern plant in Laval, where Le Matériel Téléphonique al-

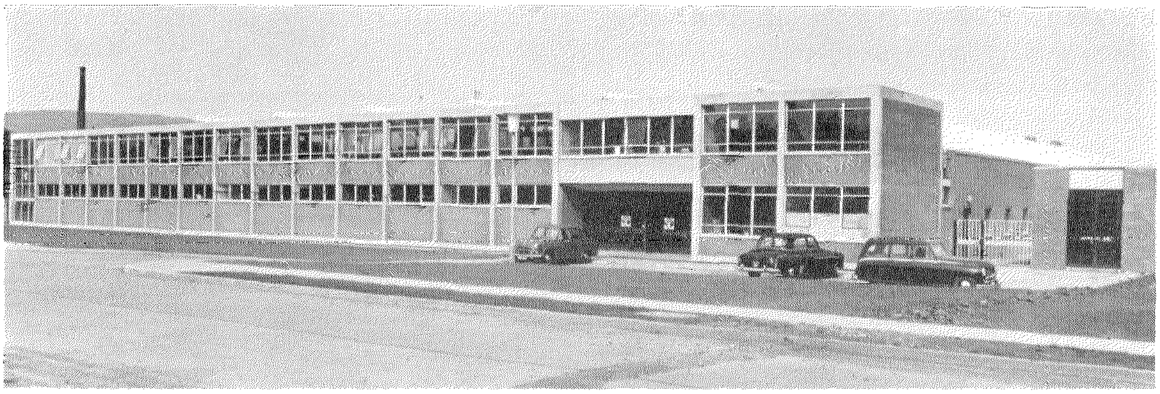


Figure 5—First factory of Standard Telephones and Cables (Northern Ireland) Limited in Monkstown, Newtownabbey, Northern Ireland.

ready has two factories. It employs over 600 persons.

Le Matériel Téléphonique
France

Australia Opens Its First Pentaconta Exchange—The first crossbar Pentaconta public telephone exchange in Australia was cut over in September 1962 at Melbourne. Present at the official acceptance of the Kew Exchange in July were: Messrs. Brian F. Jones, deputy director general of the Postmaster-General's Department; Glen N. Smith, director, and Arthur Stephens, assistant director of Posts and Telegraphs (Victoria); and T. H. Skelton, controller of stores and contracts of the Postmaster-General's Department.

With 3900 subscribers connected, 34 thousand calls were handled during the first 4 hours of service. Provision is made for 5200 lines, 5000 numbers, 238 incoming trunks, and 245 outgoing trunks. An important feature of the exchange is automatic alternative routing within the network to which it belongs. A second or a third alternative routing will be sought if the preferred route is busy. This may require a modified number of digits to be transmitted to a transit exchange.

Compagnie Générale de Constructions Téléphoniques
France

Standard Telephones and Cables Pty. Limited
Australia

Communication Between Austria and Italy—The map, Figure 6, shows the coaxial-cable and microwave radio network of Austria and its extension into Italy. The recently inaugurated coaxial system between Klagenfurt and Tarvisio will increase the 159 present channels between the two countries to an ultimate of 960 channels.

From Klagenfurt, two 4-gigacycle-per-second radio systems provide for multichannel telephone and a 625-line television signal to Graz and to Sonnwendstein, near Vienna. A 6-megacycle-per-second cable will connect the Vienna studios with the transmitter at Kahlenburg.

The coaxial cable for the Klagenfurt-Tarvisio system was manufactured in Austria. Part of the equipment at its Italian end and the coaxial system extending to Udine were provided by Fabbrica Apparicchiature per Comunicazione

Elettriche Standard of Italy. The remaining major part of the system was provided by Standard Telephones and Cables.

Standard Telephones and Cables
United Kingdom

Receiver-Indicator for Loran C—A new receiver-indicator for low-frequency long-range loran is shown in Figure 7. It is self tracking, once a signal has been acquired, and will provide continuous precise navigation data out to 1500 miles (2778 kilometers) from the transmitter. Previous systems were forced to sacrifice accuracy to obtain such range.

The equipment requires only 150 watts, permitting emergency operation from batteries, occupies 1.75 cubic feet (0.05 cubic meter) of space, and weighs 75 pounds (34 kilograms).

ITT Federal Laboratories
United States of America

Pneumatic Tube System for Toll Tickets—Despite the extensive automatization of the international toll system, with its attendant increase in traffic, many manual toll boards are still in service. A fully automatic toll-ticket pneumatic tube system has been installed to improve manual operations at the Zurich toll exchange.

One of two distributing centers is shown in Figure 8. The incoming tubes, which concentrate into a single tube, are above the outgoing

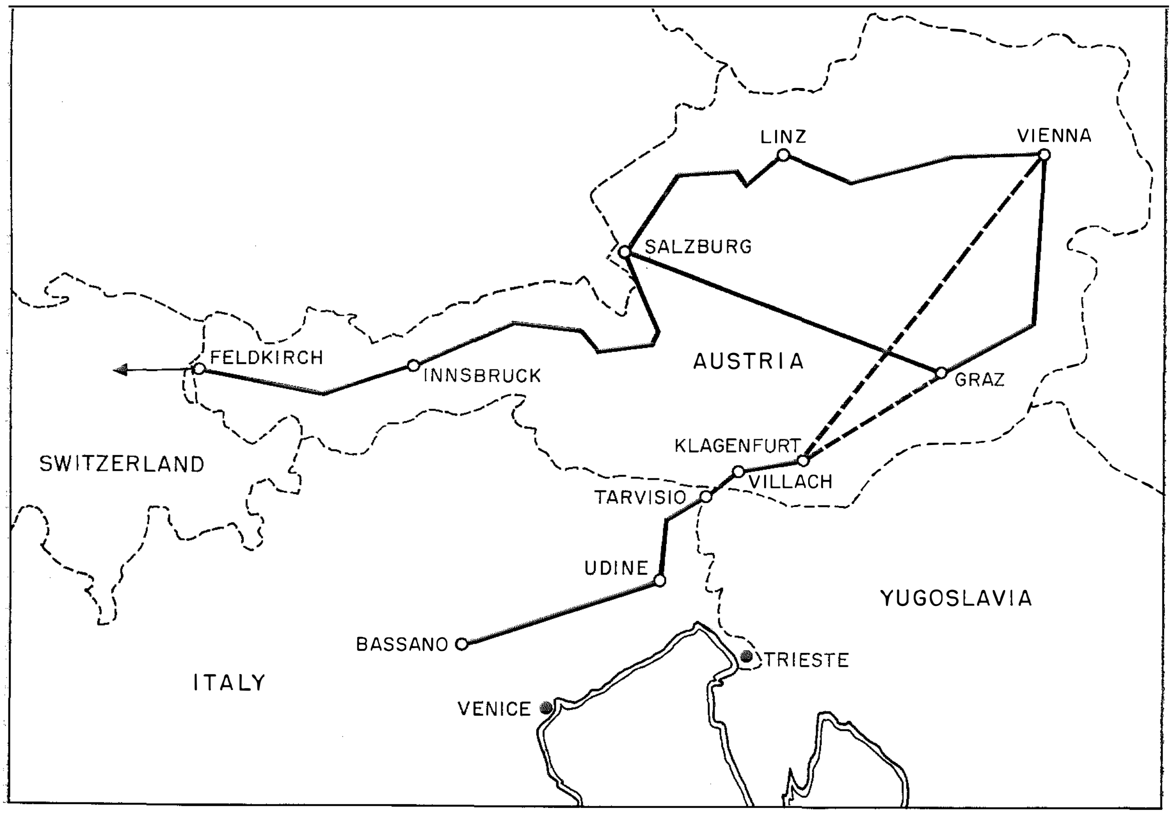


Figure 6—Austrian trunk system and its extension into Italy. Solid lines are coaxial cables and broken lines are microwave radio systems. Bassano and Udine are connected by small-diameter coaxial cable.

tubes with their remotely controlled routing valves. Supervisory photoelectric devices are not visible in the photograph. Capable of serving 208 stations, the tubes not yet needed have not been installed.

Control is from a central register utilizing conventional telephone components including a Pentaconta selector. The tickets are routed by transmission of digits from the key set at the toll position.

Standard Telephone et Radio
Switzerland

Gyroscope Within a Gyroscope—The precision to which a gyroscope will maintain its orientation in space is importantly related to the amount and variation of friction between the rotating mass and its support. Figure 9 shows a gyroscope consisting of an optically smooth fused-quartz sphere mounted within a metal sphere that rotates on a shaft.

Initially, both spheres are rotated as a unit by an air-driven turbine. When operating speed is achieved, several high-pressure air jets float the quartz sphere within its enclosure, providing a nearly frictionless bearing and a highly flexible coupling between the two spheres that rotate at the same speed.

Deviations of the axis of rotation of the outer sphere with respect to that of the inner sphere

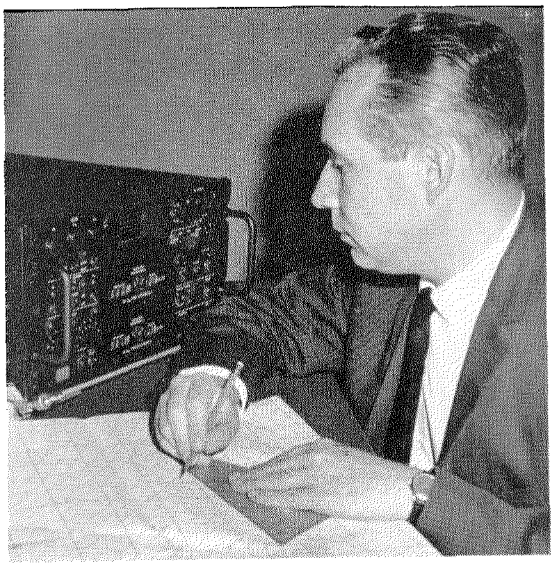


Figure 7—Self-tracking low-frequency loran receiver providing continuous precise position data out to 1500 miles (2778 kilometers).

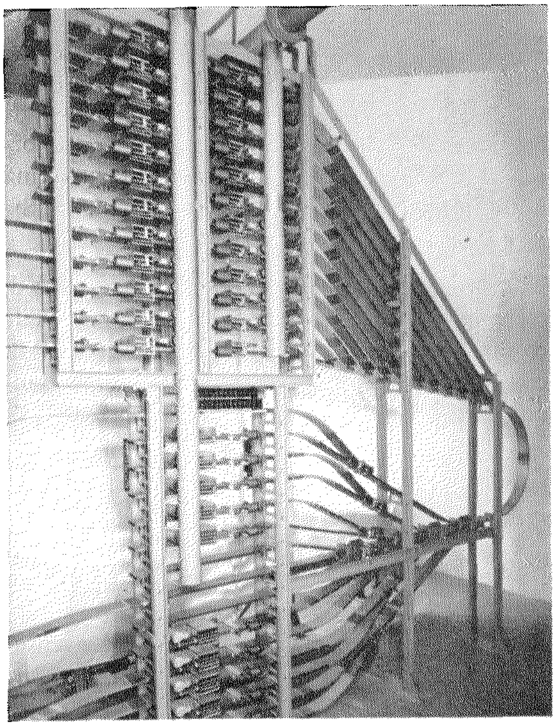


Figure 8—Distributing center for toll-ticket pneumatic tube system in the Zurich toll exchange.

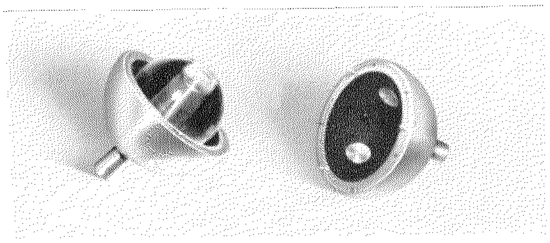


Figure 9—The rotating parts of the gyroscope within a gyroscope consist of a fused-quartz sphere that floats within an outer turbine-driven metal sphere.

are sensed by a beam of light reflected from a mirror on the inner sphere and cause the platform supporting the outer sphere to be adjusted to return the system to its proper orientation.

ITT Federal Laboratories
United States

Data-Processor Control of Switching—A stored-program data processor has been combined with a switching center to supervise its operation. Messages may be switched automatically to alternative routes in event of blockage or loss of the preferred path, transmission accuracy is continuously checked and retransmission ordered to correct errors, and message accounting is performed to insure receipt of all data at the proper destination.

The programming may be changed as needed, providing great flexibility of operation. The equipment shown in Figure 10 has been in continuous operation as part of a major communication complex since 1960.

ITT Federal Laboratories
United States of America

Destination Area Recording in Long-Distance Dialling—Information on traffic flow over telephone toll-exchange trunk groups is most useful for network design if it is in terms of the areas to which the calls are directed. The recorder, shown in Figure 11, to do this is connected to an originating register in the toll exchange to punch on tape the destination area code for each call.

The tape is scanned by automatic equipment that may extract any desired information such as the proportion of total calls to each area, their value in traffic units, or the sum of events recorded on several tapes.

Standard Elektrik Lorenz
Germany

Deep-Water Acoustic Monitor—Capable of probing to an ocean depth of 7000 feet (2129 meters), a new hydrophone, low-noise solid-state amplifier, high-speed constant-tension winch, and specially developed communication cable make up a new light-weight acoustic monitoring system. The new cable consists of two silver-plated copperweld conductors covered with an extremely tough insulating material that is completely free of pin holes that usually degrade the performance of cables operating at deep-sea pressures. Less than 0.1 inch (2.5 millimeters) thick and weighing about 25 pounds per mile (6.1 kilograms per kilometer), the cable is capable of carrying wide-band television signals.

ITT Federal Laboratories
Surprenant Manufacturing Company
United States of America

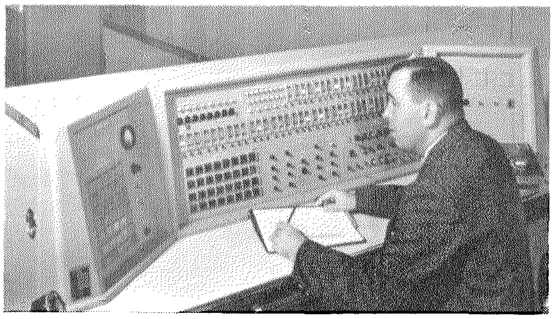


Figure 10—A combined switching center, computer, and data processor that can handle 75 000 words per minute.

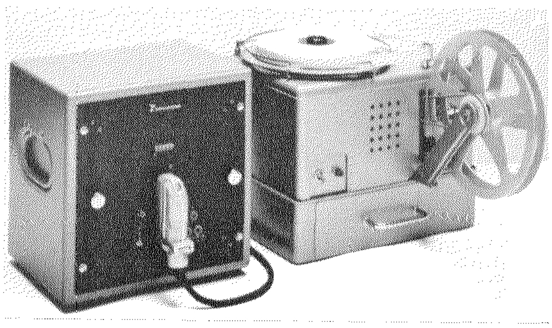


Figure 11—Equipment for recording, for toll traffic calculations, the area codes to which long-distance calls are directed.

Single-Sideboard Radio Equipment—Transmitters and receivers for single-sideband radiocommunication have been developed for use in uninhabited tropical regions. Simplicity, low cost, and ease of maintenance and operation have been primary features of the design. Vacuum tubes have been used throughout.

The fixed equipment includes a receiver, a 300-watt transmitter, and a 75-watt transmitter-receiver. The latter, shown in Figure 12, is available in both simplex and duplex versions.

The mobile transmitter-receiver units are of 30 and 75 watts capacity. A separate control unit mounts on the vehicle dashboard. The antenna unit permits efficient loading over the band from 3 to 15 megacycles per second of antennas designed for passenger cars and for trucks.

Accessory units permit remote control, use of teleprinter circuits, and interconnection with telephone exchanges.

Standard Electrica
Portugal

Marine Radio Receiver—The triple-conversion radio receiver shown in Figure 13 covers from 70 kilocycles per second to 30 megacycles per second. It has the exceptional stability, sensitivity, and selectivity required for operation aboard ship.

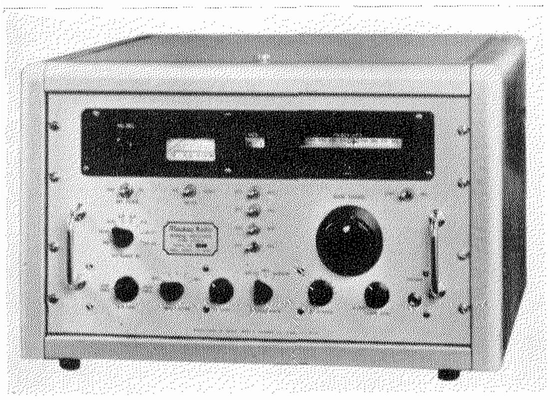


Figure 12—Single-sideband 75-watt transmitter-receiver for fixed station.

Image rejection is better than 80 decibels over the entire tuning range. The bandwidths at 6 and 60 decibels down are 6 and 20 kilocycles per second, respectively, for the broad conditions and 3.1 and 6.5 for the medium selectivity condition. Narrow-band selectivity is provided by a mechanical filter. The inclusion of a 100-kilocycle-per-second crystal calibrator assures dial settings to within 2 kilocycles per second of any specified frequency.

Mackay Radio and Telegraph Company
United States of America

Radar Indicator with Voltage Readout of Marker Positions—A 9-inch (23-centimeter) plan polar indicator for a short-range precision radar is shown in Figure 14. The console includes deflection and video amplifiers and the extra-high-voltage generator. The accompanying power supply also contains the main sweep generator and circuits for producing a marker

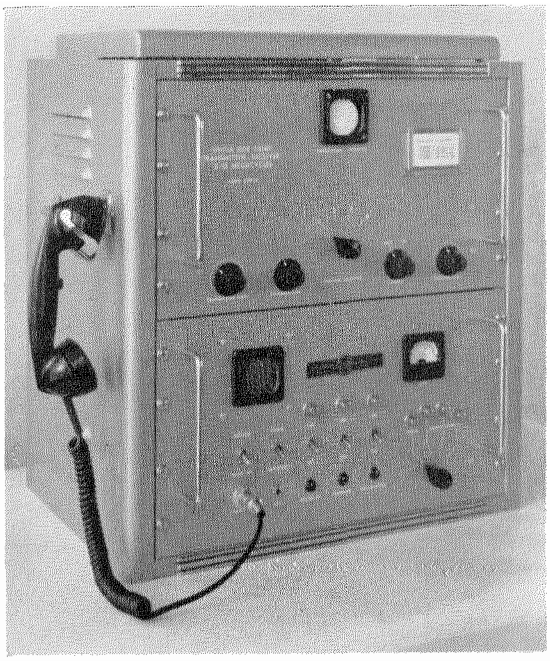


Figure 13—Marine radio receiver, which covers from 70 kilocycles per second to 30 megacycles per second.

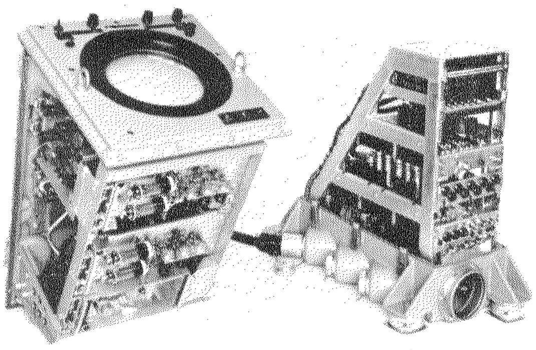


Figure 14—Plan polar radar indicator. The position of a marker is under the control of the operator. Coordinate voltages corresponding to the marker position are produced for use in other equipment.

and controlling its position on the screen of the cathode-ray tube. The position of the marker is converted to voltages corresponding to x and y coordinates originating at the center of the polar plan indicator. The coordinate voltages are available for use in other equipment.

The equipment will operate over a range from −40 to +65 degrees centigrade and will withstand mechanical shocks up to 50 gravitational units. Each weighs about 40 kilograms (88 pounds).

Standard Radio & Telefon
Sweden

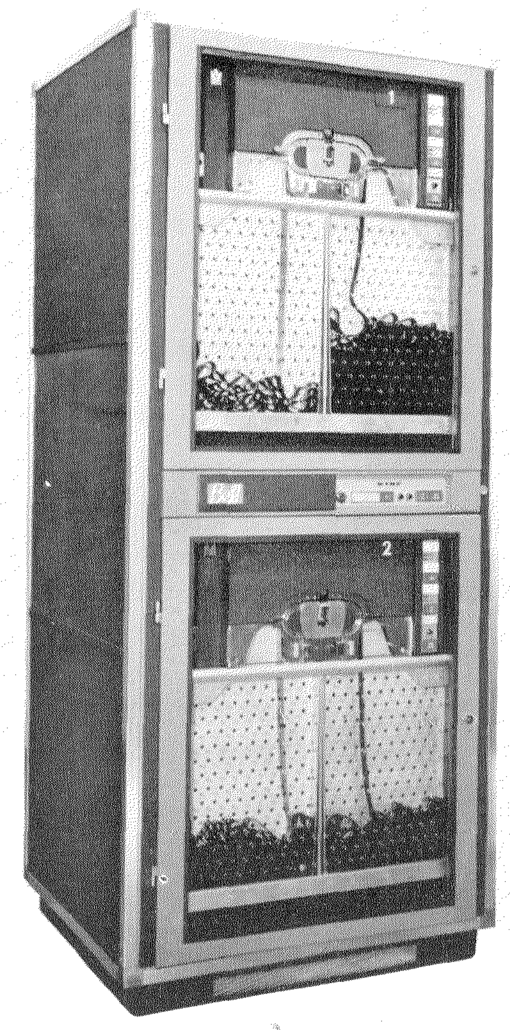


Figure 15—Magnetic tape unit with 5 transports. Two are visible at the front, 2 are at the rear side, and the remaining transport is within the framework.

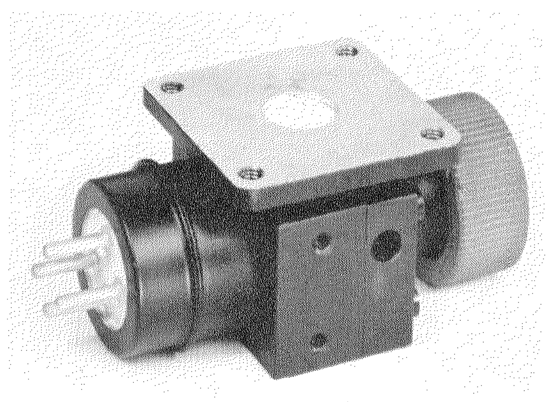


Figure 16—Reflex klystron for the 11-gigahertz region producing 0.5 watt with 600 volts on the accelerator electrode.

Compact Precision Timer—For demolition work, arming programs, and similar functions requiring extreme accuracy, a compact timer has been developed. A quartz crystal, transistor driven, provides the time base. The timing interval may be preset to several hours in increments of minutes and seconds. A 100-millisecond output pulse of 2 amperes is produced. The timing accuracy is within 0.1 percent for a supply voltage that varies between 4 and 8; it can be improved by providing a constant supply voltage. A typical unit occupies 15 cubic inches (246 cubic centimeters).

ITT Federal Laboratories
United States of America

Precision High-Resolution Oscilloscope—A 14-inch rectangular-screen cathode-ray tube provides resolution of 25 lines per centimeter. Deflection linearity is 1 percent of full-scale value along its major axes. The maximum plotting error anywhere on the screen is 0.6 centimeter. Drift is less than 0.5 centimeter per hour after warmup.

This high-resolution tube employs electrostatic focusing and magnetic deflection. It is operated in a circuit that will accept either single-ended or balanced inputs. For the latter, the sensitivity is 10 millivolts per centimeter. Response is flat to within 3 decibels from direct current to 50 kilocycles per second for full deflections, decreasing linearly at higher frequencies.

ITT Industrial Products Division
United States of America

High-speed Magnetic Tape Units—Displayed at exhibitions in Brussels, Munich, and Paris, were magnetic tape units of a new type shown in Figure 15. The tape is driven by pneumatic capstans and stored loosely in a bin of modular design to meet various dimensional requirements.

The tape speed may be between 1 and 5 meters per second (40 and 200 inches per second), its width either 12.55 or 25 millimeters (0.5 or 1 inch) and the starting and stopping times are 3 milliseconds. The recording density is 20 bits per millimeter (500 bits per inch) and 100 000 alphanumeric characters can be transferred per second on 8 adjacent tracks at this density and the highest speed. Also at this density, the capacity is 6 million 7-bit characters. Only 1 nonpermanent reading error will occur in 100 million bits.

Bell Telephone Manufacturing Company
Belgium

Reflex Klystron—A reflex klystron, VLR 222, is built of metal and ceramic for operation between 10.7 and 11.7 gigahertz. It is suitable for transmission by radio of 300 or 600 telephone channels or a television signal.

The reentrant resonant cavity that determines the operating frequency is tuned mechanically at a rate of 340 megahertz per turn of a screw acting through a diaphragm.

A modified Heil type electron gun is used. Perveance is 4.4 microamperes per volt$^{3/2}$ at an accelerator voltage of 600. The cathode current is 65 milliamperes.

Operated in the 2¾ mode with 600 volts on the resonator, an output of at least 0.5 watt is delivered to a matched load. Dependent on the mechanically set resonator frequency, the electronic tuning range varies between 60 and 110 megahertz with a modulation sensitivity between 0.5 and 1.4 megahertz per volt. The modulation characteristic is linear within 2 percent for frequency deviations of ± 5 megahertz.

As a local oscillator in the 3¾ mode at 400 volts a power of 80 milliwatts is delivered to a matched load. As evident in Figure 16, output is delivered through a circular ceramic window in a *UG-39U* flange for *R100* waveguide.

Standard Elektrik Lorenz
Germany

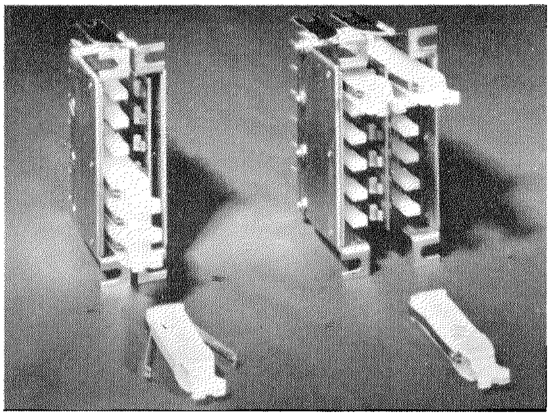
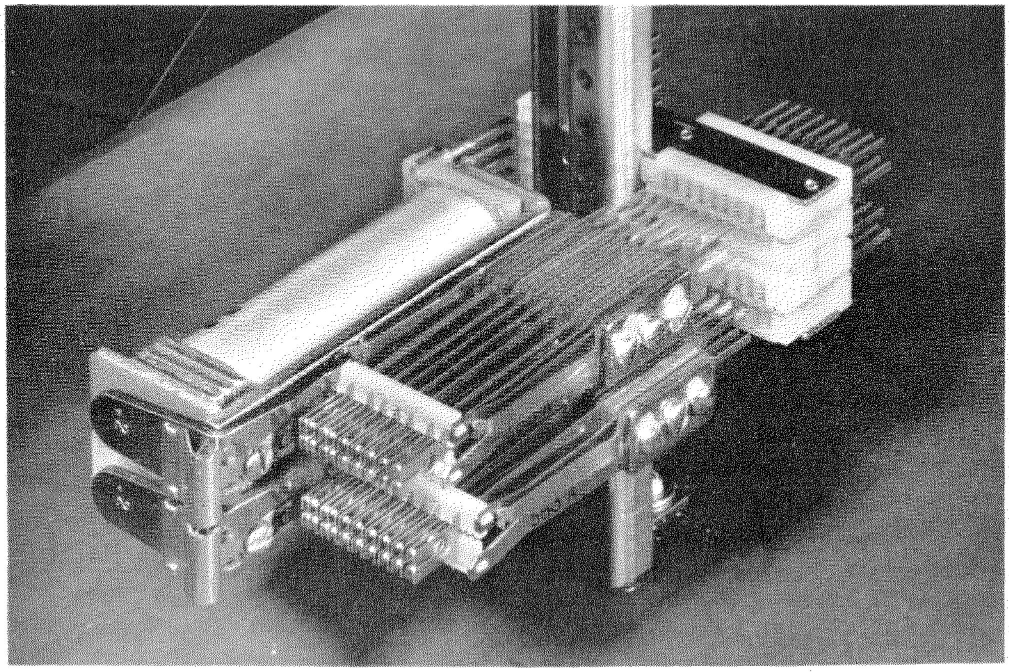
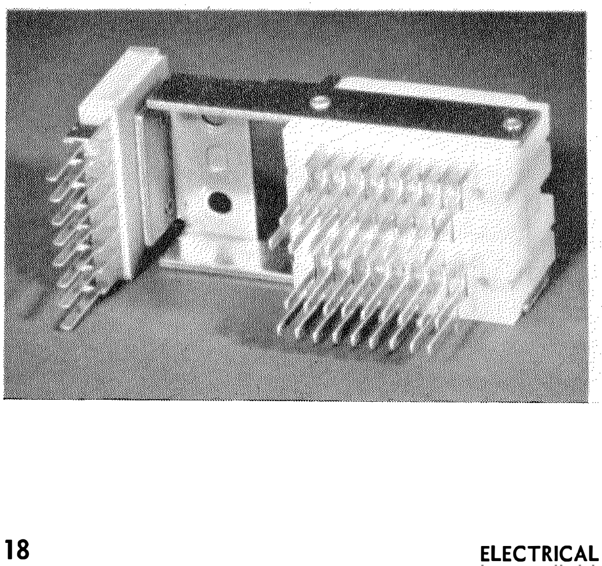
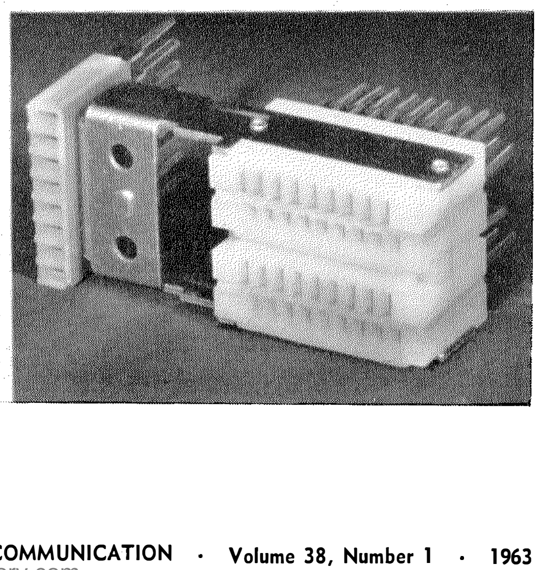


Figure 17—Cases for 6 and 12 telephone fuses. The fuse wire is soldered between the ends of the flat springs, thus holding them close to the molded frame, as in the fuse at the right.

Figure 18—Contact base into which the standard Pentaconta relay may be plugged.







Protection for Silicon Rectifiers—Silicon rectifiers, including controlled types, have limited thermal capacity and can be destroyed by rapid overheating due to voltage transients. Selenium rectifiers will absorb brief overloads and special units called Safe-T-staCs have been developed to protect silicon units. Their use generally allows employment of silicon rectifiers with reduced ratings.

Standard Telephones and Cables
United Kingdom

Telephone Fuses and Panels—For use in telephone exchange power supply leads, the compact fuses and panels shown in Figure 17 have been developed. The fuse element is soldered between the ends of 2 nickel-silver springs that also provide electrical contact to the panel. They are mounted in 2 polyamide molded pieces.

The maximum current rating is 2 amperes and the tension exerted on the fuse wire is low enough to permit a minimum rating of 0.2 ampere.

The fuse receptacles provide signalling contacts insulated from the fuses. The dimensions of the panel mounting areas are $3\frac{5}{32}$ inches (80 millimeters) by $\frac{31}{32}$ inch (25 millimeters) for the 6-fuse unit and $1\frac{33}{64}$ inches (39 millimeters) for the 12-fuse model.

Compagnie Générale de Constructions Téléphoniques
France

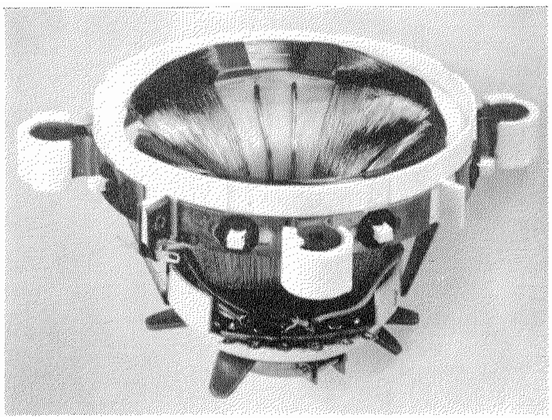


Figure 19—Deflection system incorporating spot wobble to produce a line-free television picture.

Plug-in Connectors for Pentaconta Relays—A recently developed connector, which may be fastened to a support by 2 screws, permits the standard Pentaconta relay to be plugged into a circuit. As evident in Figure 18, a maximum of 18 connecting points are provided for each of 2 spring pile-ups and 8 more are available for the operating coils.

The connector springs are of nickel-silver with 2 contact points in parallel, each exerting a pressure of 300 grams. Solderless wrapped wiring connections may be used.

Compagnie Générale de Constructions Téléphoniques
France

Television Receiver for 405 and 625 Lines—Production has been started on a broadcast television receiver designed for operation on the British 405-line very-high-frequency transmissions and on the 625-line ultra-high-frequency bands that will also be used in the United Kingdom. Only one additional valve is required over the conventional 405-line design.

Kolster Brandes
United Kingdom

Deflection Yoke for Line-Free Television—The television picture-tube deflection assembly shown in Figure 19 has a winding, in addition to the horizontal and vertical windings, that when activated by a crystal-controlled 13.56-megacycle-per-second current wobbles the spot to produce a line-free picture. To correct for both cushion and keystone distortions, four adjustable magnetic shoes are arranged at 90-degree intervals around the yoke. A negative-temperature-coefficient resistor compensates for changes in the resistance of the vertical deflection coils.

Standard Elektric Lorenz
Germany

Local Interexchange Telephone Carrier—The use of carrier systems to increase the capacity of existing multipair cables linking telephone central offices is primarily an economic matter; whether it is cheaper to lay new cable or provide carrier equipment to work with existing cable. In a test arranged with the General Post Office in England, carrier equipment has been

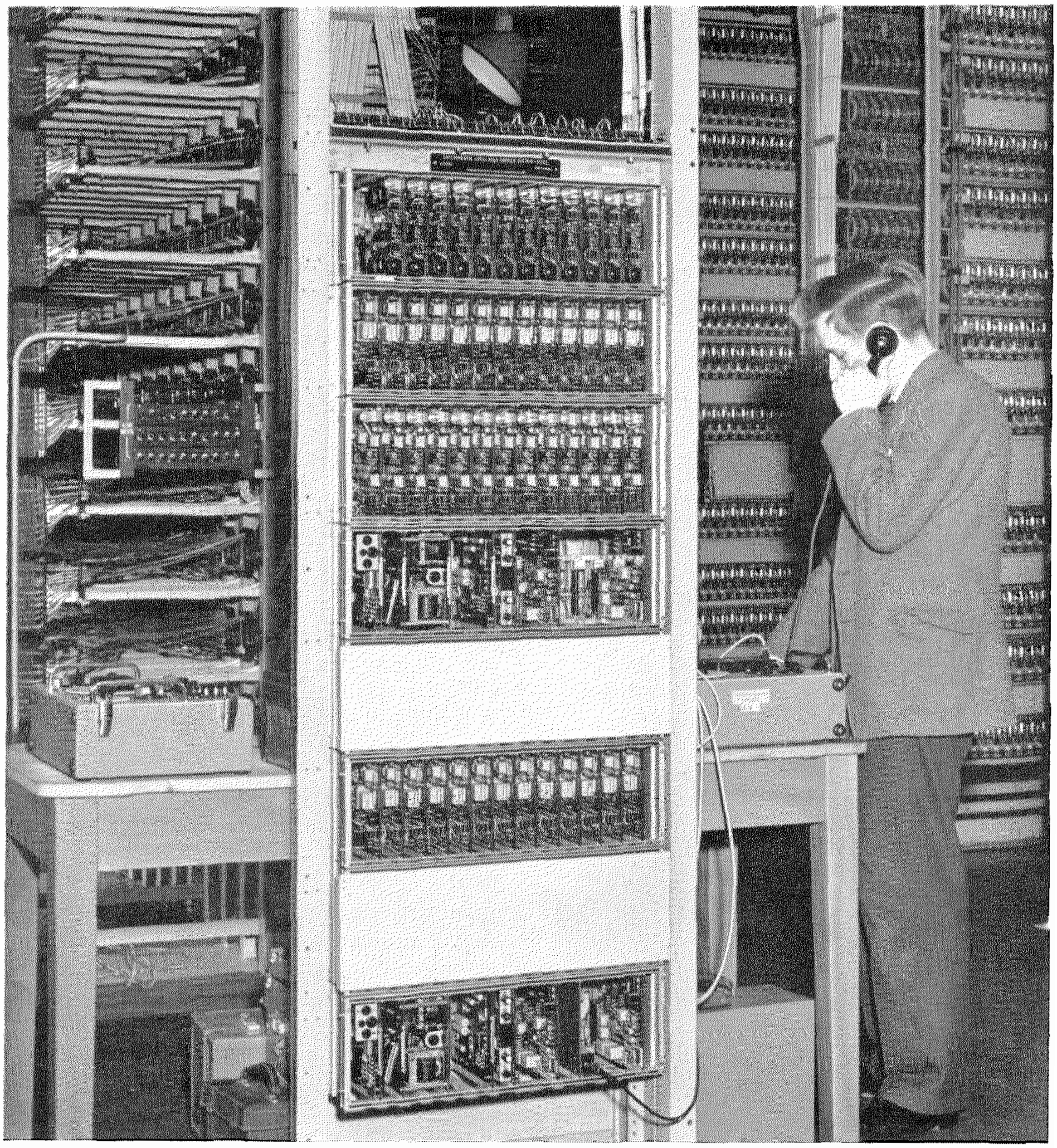


Figure 20—Carrier and double sideband is being tested for interexchange telephone carrier over existing multipair cable.

installed at London and Kent and at 9 points along this cable route.

Using new developments in transistors and ferrites and avoiding expensive filters by employing carrier and both sidebands, two pairs in existing cables carry 12 both-way voice channels and a 13th channel that is utilized for dialling and other signalling. Some of the terminal equipment in the Crayford exchange in Kent is shown in Figure 20.

Standard Telephones and Cables
United Kingdom

Barrier-Grid Storage Cathode-Ray Tube—In Figure 21 is a barrier-grid storage cathode-ray tube usable as a moving-target-radar indicator, as a random-access binary store, and as an image-storage device. The 5-inch (127-millimeter) tube contains a precision electron gun and a sandwiched target made up of a metal backplate, a dielectric layer, and a barrier grid having square apertures. The scanning electron beam passes through the grid to the dielectric layer. Secondary electrons are emitted from the dielectric and pass through the barrier grid to a collector, producing an electric signal.

Depending on the charges left on the elemental dielectric areas and the potential on the barrier grid, a state of equilibrium can be reached. No secondary electrons then pass through the grid to the collector and there is no signal output.

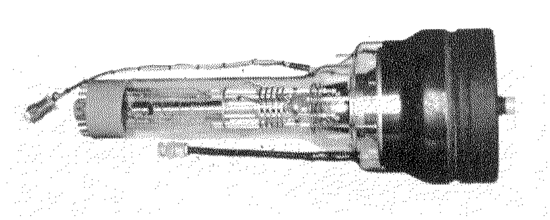


Figure 21—A barrier-grid storage cathode-ray tube usable as a moving-target-radar indicator, a random-access binary store, and as an image-storage device.

As a radar moving-target indicator, the radar video signal is applied to the backplate in synchronism with the scanning beam. After a few scanning cycles, the repetitive input signal from a stationary or clutter target causes the tube to reach equilibrium and output is produced only for moving targets. A cancellation rate of 26 decibels is achieved for clutter.

As a random-access binary store, the tube can store about 58 000 bits in a 240-by-240-bit raster. The resolution is 1200 lines per diameter and 15 writing levels are distinguishable in the output.

ITT Industrial Laboratories
United States of America

Reciprocal-Type Ferrite Phase Shifter—A microwave phase shifter of the reciprocal type, useful for phase modulating microwave signals and for simulating Doppler signals, consists of a ferrite body in a short rectangular plastic X-band waveguide. An exciting current at 10 kilocycles per second produces a phase shift of 1.3 degrees per milliampere. The insertion loss is less than 1 decibel and the voltage standing-wave ratio is less than 1.2. A power supply and meter calibrated in degrees of shift from 0 to 450 form a convenient operating aid.

ITT Federal Laboratories
United States of America

Surbond and Surock Wire Insulation—Surbond is a Teflon insulated wire with a coating that assures adequate adhesion in modern potting compounds, such as polysulfide, polyurethane, epoxy, and silicone types as well as improved surface resistance and aging over uncoated Teflon.

Surock is a specially coated thin-wall Teflon insulated wire that is two to three times as good as the uncoated product for abrasion resistance, cut-through resistance, and hot-soldering-iron endurance. Electrical properties and resistance

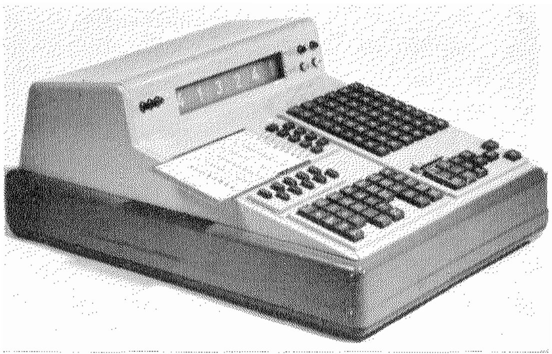


Figure 22—Load-control set used in the weight and balance system installed at Copenhagen-Kastrup airport for Scandinavian Airlines.

to solvents are unaffected by the coating. Adhesion to potting compounds is increased.

Surprenant Manufacturing Company
United States of America

Aircraft Weight and Balance System—More than a year ago, there was installed at the Copenhagen-Kastrup airport for Scandinavian Airlines a system for reporting the weight of passengers, luggage, and freight to be put aboard each aircraft and its distribution as an aid in balancing the load.

A Stantec Zebra computer is associated through a central electronics system with 40 weight-input and 3 load-control sets. The weight-input sets are used in all bookings of passengers and luggage to record flight class, passenger category, reservation status, number of passengers, and destination airport. The central equipment acknowledges an acceptable booking operation with a green light. A lack of seats or weight overload for a particular aircraft will cause a booking to be refused, which is signalled by a red light and the pertinent information is recorded on a page printer.

The load-control sets, shown in Figure 22, are in the central office and permit the limits within which each particular aircraft may be loaded to be recorded in the store of the computer. A visual display indicates the bookings already completed and permits changes to be made to accommodate different types of loads. Shortly before departure, a final statement of loading is provided by the load-control set. It includes the addresses of both senders and receivers of all freight, which are transmitted by page printer to the destination airport.

The system handles about 15 thousand input items per day in serving 90 to 100 flights. Engineers of Standard Telephones and Cables, United Kingdom, collaborated in the design.

Standard Elektrik Lorenz
Germany

Clavier Receives Award

The Achievement Award, the highest issued by the Professional Group on Communication Systems of the Institute of Radio Engineers, was presented on 3 October 1962 to Andre G. Clavier "for his outstanding contributions to the science and technique of communication systems."

A member of the research staff of the International System from 1925 to his retirement in 1962, he published extensively on his continuous activities on microwave equipment design and propagation studies.

Application of Pulse-Code Modulation to an Integrated Telephone Network

Part 1—Advantages of Pulse-Code Modulation

P. MORNET
Laboratoire Central de Télécommunications; Paris, France

1. Introduction of Electronics into Telephone Switching

The essential advantages expected from the introduction of electronics into telephone switching are a reduction in the volume of equipment, greater reliability, and increased flexibility of operation.

The first two advantages reduce housing and installation costs and maintenance. The third can increase operating profits by offering additional services to subscribers.

Electronic switching methods differ in the way the principles of space or time division are applied to the speech circuit and to the control circuit.

The distinction between these two types of circuits—the former actually providing the speech signals between the subscribers, and the latter controlling the speech connections to be established or released—appeared more and more clearly as the evolution of electromechanical exchanges progressed and, of course, remains valid for exchanges introducing electronic elements of any type whatever.

Both space and time division of switching functions have developed progressively in the past.

In time division, the frequency of use of any component or element is a direct function of its operating speed. If its response time is very short compared to the time allotted to perform its particular operation, the whole exchange may need only one such element, which will handle all calls and will usually be installed in a central position.

Several experimental exchanges have been made having a control circuit operating on this principle and a speech circuit of the space-division type in which the crosspoint, which actually provides the continuity of the speech path, consists either of metallic contacts or electronic elements.

Among these experimental exchanges, two may be mentioned; one was installed and run by Bell Telephone Laboratories at Morris, Illinois, in the United States, and the other is a private 240-line electronic exchange built by Laboratoire Central de Télécommunications, which has been in operation in Paris since January 1960.

However, as none of the solutions for the problem of the electronic crosspoint has been considered completely satisfactory, attention has been focussed on semielectronic systems as the immediate successor to electromechanical systems.

2. Time-Division Speech Network

While such semielectronic systems are of great interest, there seems to be a controversy over whether they are only a stage toward techniques in which the speed of the electronic elements can be used to better advantage.

In fact, in a space-division speech network, the number of crosspoints is always very large, even using the most elaborate methods for their number. Moreover, the time during which a given crosspoint is used always is short compared to the time during which it is waiting.

As has long been known, it is not necessary to transmit speech as a continuous signal, but it is sufficient to transmit brief samples at a rate that is at least twice the highest frequency of the speech spectrum.

Several transmission systems based on this principle have been tested or operated, all using of course the fundamental advantage of interleaving the samples from several conversations to obtain multiplex operation.

With fast electronic elements, the principle of space division can also be applied to the speech circuit. Crosspoints may still be required but through multiplexing their operating rate is much higher than in space division; as a result their number may be considerably reduced and thus affect the volume of the exchange equipment.

3. Time-Division Transmission Systems

Transmission systems based on time division differ essentially from one another in the method used for the transmission of the information contained in the sample of the speech wave. The following 3 types of pulse modulation have been developed.

A. Pulse-amplitude modulation in which the amplitude of the pulse itself varies with that of the sample.

B. Pulse-width modulation in which the amplitude remains constant but the width of the pulse is a function of the modulation.

C. Pulse-time modulation in which the amplitude and width of the pulse remain constant but the time of the pulse with respect to a reference time varies as a function of the modulation.

4. Pulse-Code Modulation

These various types of modulation are all more-or-less vulnerable to noise and crosstalk. Much more effective protection can be obtained with the use of pulse-code modulation disclosed in French patent 852 183 issued to Reeves in 1938.

According to this process, the amplitude of each sample is measured and its value is transmitted in the form of a series of pulses. Only the presence or the absence of a pulse is significant.

Thus, only the disappearance of a pulse from the signal or the appearance in it of an additional pulse can cause a deterioration of the information transmitted.

In fact, if the maximum amplitude of an extraneous pulse is less than half of that of the coded signal pulses, the original signal can be reconstituted without error at the receiver. A mere alteration of the characteristics of one or several pulses caused by crosstalk, distortion, or noise has no effect provided the above limit is not exceeded.

Generally speaking, pulse-code modulation provides great reliability of transmission, and noise, crosstalk, and attenuation of the speech are independent of the transmission length and of the number of repeater points.

4.1 Increase in Switching Reliability

The increase in reliability of pulse-code modulation is however not limited to the transmission itself. It is a well-known and evident fact that, a priori, elements operating under ON and OFF conditions guarantee good operation superior to that from elements that are required to have precise characteristics independent of time and the conditions of use, such as temperature, power supply, et cetera.

Moreover, the ON-or-OFF conditions are particularly convenient for the operation of transistors, which are thus especially suited to the field of switching. In both these conditions, the power dissipated is, in fact, negligible. In the ON condition, the difference of potential between emitter and collector is very low, of the order of a tenth of a volt for germanium and of the order of a volt for silicon, whereas in the OFF condition, the current flowing through the transistor is only a leakage current, with an intensity of a few microamperes.

The adoption of this type of modulation permits applying such operating conditions to all the circuits dealing with speech signals. Such is the case for transmission equipment and for the speech circuit in switching.

4.2 Problem of the Crosspoint

For the speech circuit in particular, the problem of the crosspoint is greatly simplified. In space switching, besides a low price, an impedance ratio of at least 10 000 between the closed and open (off) conditions is required, this ratio determining the crosstalk. In time-division switching, such as in pulse-amplitude modulation, one imperative requirement of the crosspoint is that it forgets quickly the information that it has just transmitted so as not to disturb the following information or, in other words, not to introduce crosstalk. This means that all transistors used as gates must possess extremely short switching times.

Pulse-code modulation is, on the contrary, perfectly compatible with closed-to-open impedance ratios of the order of 10 and requires no particular speed of operation. These features together with the greatly reduced number of crosspoints for time division make this type of modulation ideal for switching.

For an objective assessment of the merits of pulse-code modulation, however, it is necessary to consider also the conversion of analog signals to digital signals and vice versa. The coder and decoder units that perform these operations are the only elements of the speech circuit that do not work completely in this ON-or-OFF mode. Thus, their quality of communication depends only on these units provided the minimum conditions required of transmission and of the switching circuits are fulfilled.

5. Problem of The Telephone Network

The advantages of pulse-code modulation can best be used in an integrated network, that is, one in which both transmission and switching are done with the same type of modulation.

Experience has shown that pulse-code modulation, because of its resistance to interference, can be transmitted on ordinary telephone pairs whether or not they form part of ordinary telephone cable. The attenuation of the pulse is of course great, but it is sufficient to introduce a simple repeater at about every 2 kilometers, to restore the whole of the information without loss. In practice, these repeaters could occupy the place of the Pupin coils used at present, thus involving only small installation expenses.

In actual practice, a 24-channel pulse-code-modulation system requires two telephone pairs, one for each transmission direction. Nevertheless, 24 communications are handled by 2 ordinary pairs which, in conventional systems, carry only 2 communications.

Moreover, the principle of the 4-wire connections imposed by pulse-code modulation might perhaps be advantageously extended to the subscribers' lines. The subscriber's set *and line circuit,* elements whose price has a large influence on the total cost of a system, can indeed be simplified under certain conditions.

The possibility of multiplying transmission capacity by a large factor can be exploited in different ways, but it appears, especially if one considers the local telephone network where connections are short, that the switching circuits also should be adapted to pulse-code modulation to avoid costly equipment for changing the type of modulation. It is also preferable to extend the multiplex transmission as close as possible to the subscribers and this leads to considering the generalized use of a structure of the "expanded" type for the local network. Figure 1 gives an example of this.

The subscribers in such a network are connected to concentrators. Each concentrator is served by a pulse-code-modulation highway of, for example, 24 channels. The local exchange itself then essentially comprises only the switching circuits between pulse-code-modulation multiplex highways.

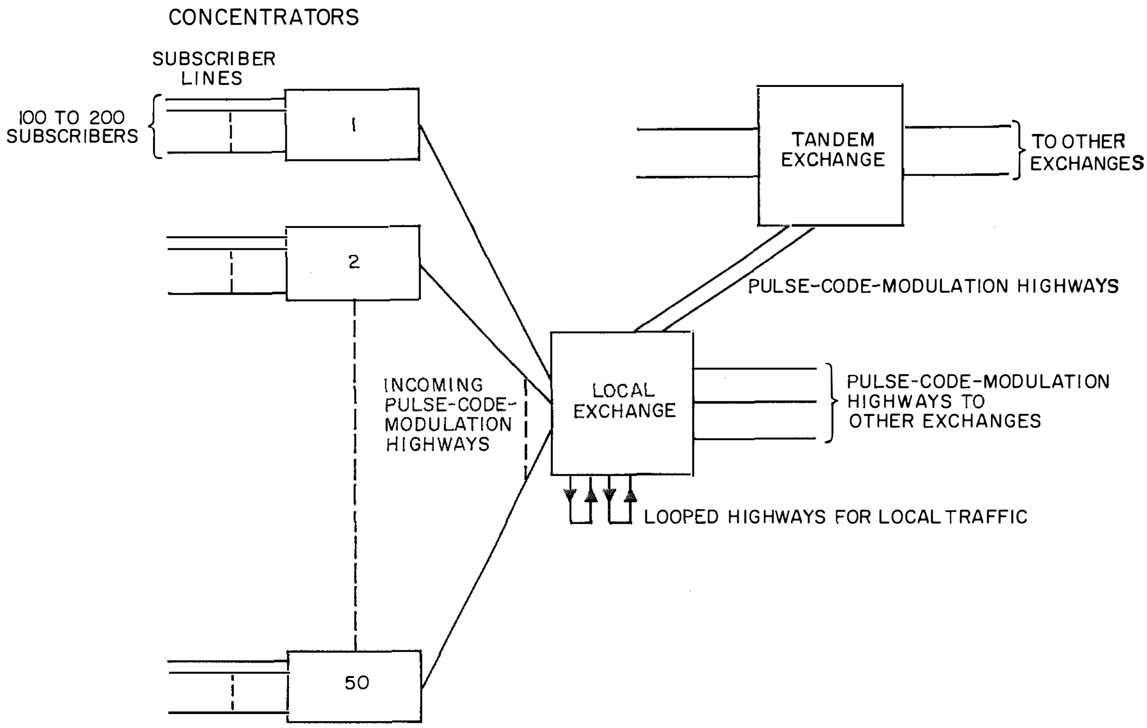
These highways are of 3 types: those providing connection to concentrators, those allowing interconnection with other exchanges, and a few to handle local traffic.

Lastly, the transit exchanges ensure interconnections between local exchanges and only need perform switching operations between pulse-code-modulation multiplex highways.

5.1 Concentrator

A concentrator, shown in Figure 2, consists essentially of the switching stage between the subscribers' lines and the pulse-code-modulation highway allotted to this concentrator. It can be

Figure 1—Local telephone network using pulse-code modulation.



placed in the same building as the local exchange but, to make the best possible use of the expanded type of exchange, it will preferably be put near the subscribers who are connected to it, and thus minimize the length of their lines. It includes the sampling circuits as well as the coding and decoding devices. Its very small size permits easy installation.

5.2 Local Exchange

Consider as a typical example a local network of 8000 subscribers having individually a traffic of 0.1 erlang at the busy hour, a local traffic of 20 percent of the total traffic, and a distribution of the lines over 50 concentrators with each one serving 160 subscribers.

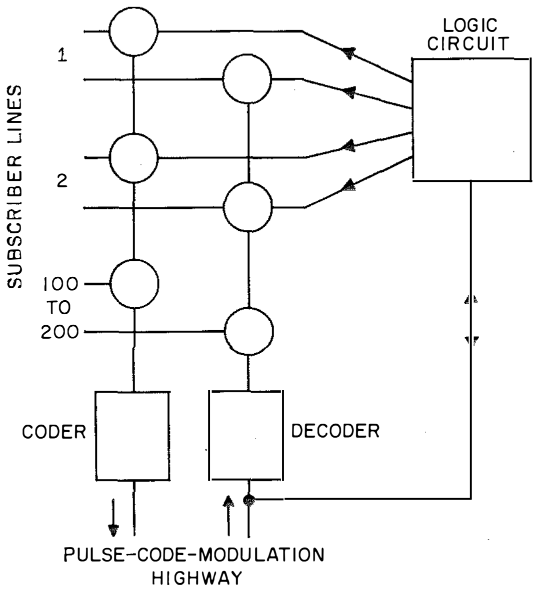
The local exchange must thus ensure both the interconnections between the 50 pulse-code-modulation multiplex highways connecting it to the concentrators and the highways connecting it to other local or transit exchanges.

In principle, a transit exchange will then be no different from a local exchange since, in the two cases, the switching operations to be carried out are made only on pulse-code-modulation multiplex highways.

There are therefore, in such a network, only two basic switching stages, the first assuring connections between subscribers' lines and a pulse-code-modulation junction, which are the concentrators, and the second, which is the basic element of the local and transit exchanges, assures interconnections between the various pulse-code-modulation multiplex highways.

Both types of switching stages have been constructed and tested experimentally.

Figure 2—Concentrator.



A preliminary study of the speech network of a complete exchange has furthermore shown that, in general, one switching stage could be sufficient. The adoption of a more-complex connection network with several stages does not seem justifiable. Even if it permits some savings in the number of crosspoints, this number is already much reduced by multiplex operation and the gain in absolute value can only be small. The network with one stage therefore seems to be the best solution, even for very large exchanges.

5.3 Network Operation in Time Division

The digital nature of the pulse-code-modulation signals makes it possible to record them in a binary-cell memory. This possibility is made use of as soon as the signals arrive at the exchange. Without this input memory, the establishment of a connection between two subscribers would present the problem of finding a time position simultaneously free on all the multiplex highways involved in the connection between the concentrator of the calling subscriber and the concentrator of the called subscriber. The choice of the time position would then necessitate the passing of information among all the interested exchanges. Moreover, an alignment of the time positions would have to be made on a variable number of multiplex highways according to the geographic positions of the two subscribers. The traffic possible on these highways would then be very small and this would lead to a poor use of the transmission and switching circuits.

5.3.1. *Alignment of Time Positions in Each Exchange*

The situation described above can fortunately be remedied by recording the signals on their arrival in the exchange in what is called a speech memory.

With this speech memory is associated another, the so-called time-switching memory, which contains the identity of the various lines of the speech memory. A cyclical reading of the time-switching memory allows the reading or writing of the speech-memory lines in any order.

This order is chosen by the control circuit of the exchange so as to satisfy the alignment of the time positions inside the exchange.

The unit formed by these two memories, of speech and time switching, makes it possible to carry out a switching in time of the channels of an incoming highway. They serve therefore to make the time at which a signal is handled in the exchange independent of its arrival time.

An example of this is given in Figure 3. A speech signal arrives on channel 12 of highway J_i and is written on line 12 of *MP1*, under control of the synchronization circuits. The identity of this line 12 is written on line 15 of *MCT1* under the control of the logic circuit. At time interval 15, line 15 of *MCT1* is read, line 12 of *MP1* is read and its contents transferred to line 17 of *MP2* connected to highway J_o. The identity of this line is supplied by *MCT2* and read at time 15. The transfer from *MP1* to *MP2* is made by the crosspoint connecting J_i to J_o, which is conductive at the time of the reading of line 15 of *MCS* at time 15. The speech signal will be transmitted to J_o at the reading of line 17 of *MP2* at time 17.

5.3.2 *Rearrangement*

Despite the great flexibility provided by speech memories associated with switching memories, blocking can occur inside an exchange, which reduces efficiency in the order of 50 percent for 24-channel highways. A novel method of connection break-down and rerouting, referred to as the "rearrangement" method, overcomes this difficulty and eliminates blocking due to the alignment of the time positions by taking advantage of the flexibility provided by the speech memory.

The principle of the method is explained in part 3 of this series of articles. At this point it suffices to mention that this method greatly increases the highway efficiency at the cost of only a slight increase in the control equipment of the exchange, and that the rearrangement operations are performed with negligible or zero loss of information.

In the application of this method, it is feasible either to limit the transpositions to those that can be made within an exchange or to make transpositions also on the time positions in the transmission circuits between exchanges.

The complexity of the signalling problems involved in the latter solution leads us to think that at present the solution that limits the rearrangements to the inside of each exchange is preferable, despite its smaller savings in equipment.

The flexibility provided by the purely digital nature of all the signals makes it possible to realize exchanges of absolute operational reliability that have a speech circuit of very-reduced dimensions and that permit best utilization of the transmission circuits resulting from increased capacity and very-high efficiency.

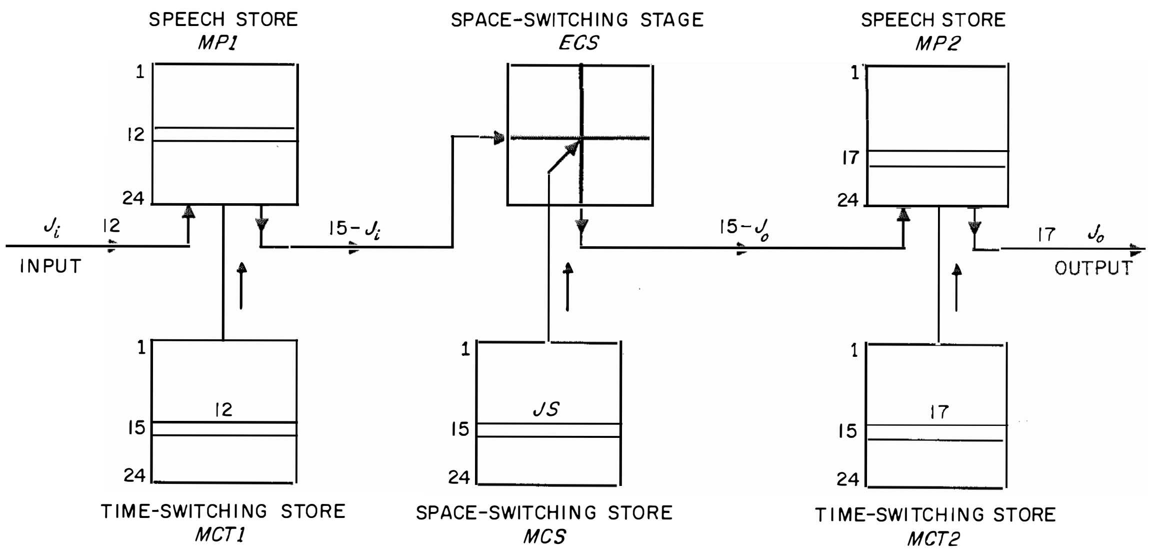


Figure 3—Switching stage between pulse-code-modulation highways.

5.4 Synchronization

A very-great problem is that signals arriving over the highways with any phase whatever must, under control of a local clock at the exchange, be identified, separated, and correctly routed.

This problem is easily resolved for the concentrator and would not be very difficult for a star-shaped network, but it becomes more difficult in a mesh network because priorities must be assigned to the different exchanges to properly control the whole network and because the transmission parameters are variable.

There is, however, a very sure solution through use of the digital memory that records the signals from each highway on their arrival in the exchange.

Writing in this memory is controlled by a synchronization circuit that introduces time delays that may be as large as the time between adjacent bits to compensate for slow shifts or rapid jitter in propagation time. It places all bits in positions corresponding to their original periodicity.

The synchronization circuit also identifies each channel by its code and routes its incoming bits to a designated channel line in a memory. Because the timing of the bits has already been corrected, this memory can operate normally and the reading can then take place at any time determined by the local clock of the exchange. This memory thus plays the role of a buffer and allows asynchronous operation by eliminating those long slow variations of propagation time that are too large to be handled by the vernier time-delay circuit.

Obviously, this process involves in certain cases a loss of information, which is manifested only by a noise that is absolutely negligible in telephony. A relative variation of frequency of 10^{-6} between the incoming information and the local time causes a loss of the order of 1 pulse out of 10^6.

6. Introduction of Pulse-Code Modulation into Existing Networks

The question arises of course as to how such a network could gradually replace the existing network or at least be introduced into it.

It is first necessary to mention that there will probably be an intermediate stage during which the telephone exchanges will have centralized electronic control circuits with a programming flexibility such that the introduction of new network operating techniques will be possible. This intermediate stage should permit the later introduction of the pulse-code-modulation integrated network.

But, independent of this consideration, a certain number of suppositions can be made. Pulse-code-modulation might first be introduced into connections between exchanges to avoid installing new cables. Some pairs in saturated cables could thus be used to constitute pulse-code-modulation highways. Satisfactory tests have already been made along this line.

Another possibility could be that in areas of rapid population expansion existing subscribers' pairs could be grouped in two's to constitute pulse-code-modulation highways serving concentrators.

The efficiency of the lines would thus be multiplied by a factor of the order of 100 since only 2 pairs would permit connecting up to 200 subscribers to the network.

Equipment for changing the type of modulation as well as the type of synchronization would then have to be provided at the points of connection with the existing local exchanges while waiting for the number of concentrators to become high enough to justify transformation of the exchange itself.

It is, of course, possible to imagine other processes, but only a methodical study of the extension of the network and of its economy will show how the problem may best be approached.

In the case of a network that is to be built completely, only the interconnections with other networks would present problems. But these being few in number, the solutions, even costly ones, can then be considered without difficulty.

7. Problem of The Military Network

One particular case, however, the military network, merits special attention.

One of the most important considerations in this case is the setting up of the network. The network can cover a rather large area and have connections of several hundreds of kilometers established by varied means, some of which do not always present great uniformity of transmission qualities. These means can include, in fact, multiconductor cables, coaxial cables, radio links, and others that must in certain cases be connected end to end.

Questions of impedance and level matching then arise, which, with conventional types of modulation, can be resolved satisfactorily only by using a 4-wire connection from one end to the other of the transmission. It would even be preferable to handle the switching on a 4-wire basis to avoid many conversions between 2- and 4-wire circuits.

With pulse-code modulation, these problems remain but are simplified. As has been stated above, this modulation is effective with a useful signal-to-stray-signal ratio of 6 decibels and makes it possible to neglect the reflections that would be inadmissible with voice-frequency signals.

The switching problems are particularly easy to handle with pulse-code modulation and it is no longer necessary to consider conversions between 2- and 4-wire systems, which are always delicate.

Pulse-code modulation not only offers the possibility of an integrated network, but solves the complex problem with respect to conventional systems assuring a quality of speech practically independent of the length and quality of the connections.

8. Transmission of Digital Information

It is now certain that in the future telephone network, signals to transmit information other than speech will take on more and more importance. Already devices such as the Dataphone, put on the American market by Western Electric, permit such exchanges at low speed. Pulse-code modulation itself consists of information of this type transmitted at a very-high speed, 70 kilobauds for a telephone channel sampled at a rate of 10 000 times per second and for a 128-level quantization corresponding to a 7-digit binary code.

An integrated telephone network using pulse-code modulation over a 24-channel multiplex highway must provide for the transmission of digital data at a rate of 2 megabauds.

As was mentioned previously, the necessity of synchronizing the signals on arrival in the exchange and of making rearrangements from time to time may possibly involve loss of information. These losses are absolutely negligible for telephony but require, nevertheless, the application of a certain redundancy for the transmission of digital information.

The required minimum redundancy will be determined by a systematic study of the information losses in the transmission circuits as well as in the switching circuits.

Nevertheless, without appearing to be too optimistic, rates of the order of 35 kilobauds for a single channel or of 1 megabaud for the whole of a highway can be envisaged. These may appear larger than needed at the moment but they can be completely used for example in the case of long-distance centralized handling of numerical data in a computer. This possibility could be offered to subscribers.

9. Conclusion

Pulse-code modulation makes it possible to draw the maximum benefit from the introduction of electronic elements into telephone systems because it gives a homogeneous solution to the problem of the telephone network taken as a whole and not under only one particular aspect. Application of this type of modulation to telephone switching requires new techniques that have many points in common with those of digital computers, but it also presents problems that are peculiar to telephony, such as signalling, operation, and in a more-general manner those of the organization of a network.

Our theoretical and experimental work has illustrated the possibility of a practical application of this modulation to the public telephone network in a not-too-distant future.

The following articles dealing in more detail with two aspects of this problem, transmission and switching, will give a more-precise idea of the present state of our developments.

Pierre Mornet was born in Châteauroux, France, on 21 March 1919. He graduated as a radio engineer from the Ecole Supérieure d'Electricité of Paris.

He was a mechanical engineer in the Air Force before joining Laboratoire Central de Télécommunications in 1943. In 1945 he was recalled by the Air Force Army, Transmission Service. Mr. Mornet returned to Laboratoire Central de Télécommunications in 1952 and since 1956 has been in charge of the switching and transmission laboratory.

Application of Pulse-Code Modulation to an Integrated Telephone Network

Part 2—Transmission and Encoding

A. CHATELON
Laboratoire Central de Télécommunications; Paris, France

1. Introduction

Scarcely 15 years elapsed between the famous invention of Alexander Graham Bell in 1876 and the description of the first multiplex telephone system by Maurice Hutin and Maurice Leblance in French patent 215 091, filed in 1891. Nevertheless, the first commercial equipment, Western Electric, type *A,* was not built and cut over before 1918. No telephone multiplex system was in operation in France before 1932, when the *D* system of the French Post Office was inaugurated.

These first multiplex telephone systems, using aerial lines or underground cables, as well as the first multiplex radio link tested in 1931 by A. G. Clavier and L. Gallant between Saint-Inglevert, France, and Lympne, England, were based on the frequency-division multiplex principle.

At the present time, the great majority of multiplex telephone systems operated in the world are of this type. There are tens of thousands of them and the whole length of their circuits amounts to tens of millions of kilometers.

In a frequency-division multiplex system, a given frequency range corresponds to each telephone channel. The electric filters necessary for separating each voice-frequency band from 300 to 3400 hertz must differ according to the exact position in the transmission band to which that speech channel was transposed. Moreover, these filters are delicate and expensive elements.

These reasons are sufficient to explain why, very soon after the first frequency-multiplex systems were put into operation, many researchers tried to develop multiplex systems that were completely different.

It is odd to note that time-division multiplexing was not only discovered but used before the invention of the telephone: in fact, the Baudot telegraph multiplex system was put in operation in 1874.

The idea of replacing the low-frequency electric wave from the microphone by pulse signals dates from 1924; it is due to R. A. Heising. The considerations that led Heising to modulate the width of the pulses were related to improving the efficiency of a radio transmitter.

The idea of multiplexing telephone channels by interleaving pulses obtained by the cyclic sampling of the various speech signals corresponding to these channels, was conceived after 1930.

The first experimental transmissions utilizing this type of modulation were conducted in France, Great Britain, and the United States from 1937 to 1941, but they did not result in commercial production of equipment.

Other systems of pulse modulation perfectly adapted to time-division multiplexing have been proposed in French patent 846 849 issued to E. M. Deloraine and A. H. Reeves and have even led to factory production of devices, the first model of which was the multiplex type 10 of the British army, manufactured as far back as 1942.

However, in all these systems, the information transmitted was directly related to a characteristic of the electric signal that is continuously variable such as amplitude, width, or position of the pulse. The sources of distortion, crosstalk, and noise inherent in these various systems of transmission, could still considerably affect the received signal.

The credit of having invented a system of pulse-code modulation that makes it possible, theoretically, to get completely rid of the disturbances usually introduced during transmission, is attributed to A. H. Reeves, to whom French patent 852 183 was issued in 1938.

In pulse-code modulation, numbers are transmitted, represented by signals similar to telegraph signals but at a speed much higher than that usually utilized in telegraphy.

These numbers are the amplitude measurements of samples taken regularly from the speech signals. The information then lies only in the presence or absence of the code pulses and is to a certain extent independent of their amplitude, width, and phase.

The principle is simple; however, the necessity of multiplying the number of the transmitted pulses by at least 5 per sample to obtain an acceptable definition increases the bandwidth for transmission in corresponding proportion. Besides, the complexity of the encoding and decoding elements common to all channels of a multiplex pulse-code-modulation telephone system sets a minimum for the number of channels that may be advantageously multiplexed by this process.

In 1938, it was not easy to measure accurately the pulse amplitudes at the rate of several tens of thousands per second. The number of active elements used, instead of the numerous filters utilized in the frequency multiplex systems, was such that the use of vacuum tubes involved considerable difficulties.

As in all kinds of digital techniques, the introduction of semiconductors has simplified and considerably hastened the resolution of the problems raised.

2. Advantages of Pulse-Code Modulation in Transmission

It is of interest to note some of the most-important advantages inherent in the use of pulse-code modulation.

2.1. Transmission Quality Independent of Distance

For transmission, the most-spectacular advantage of pulse-code modulation is certainly the possibility of regenerating signals in an ideal manner and thus avoid, theoretically, any loss of information, so as to obtain a quality of transmission that is independent of distance. Experiments have shown that this is indeed the case, indicating that the decrease of transmission quality with distance is absolutely negligible, which is contrary to what happens in all other known systems.

2.2 Absolute Stability of Transmission-Line-Equivalent

It is also obvious that the transmission-line equivalent remains absolutely constant, however the quality of the transmission medium may vary. The variations of this transmission-line equivalent in the numerous systems using amplitude modulation, particularly in the great majority of the carrier systems on cables, are well enough known to make further explanation unnecessary.

2.3 Homogeneity

The use of a single telecommunication technique and of a coded form for the information being transmitted makes possible a more-homogeneous transmission system than was previously available.

2.3.1. *Homogeneity of Techniques*

In whatever is concerned: modulators or demodulators, control and supervision devices, regenerators, or channel-dropping equipment, to mention just transmission elements, only the digital technique is used.

In pulse-code multiplex transmission, it is possible to dispense with the use of channel filters, which are as varied as they are fragile and which are used in such great numbers in frequency-division multiplex telephone equipments.

Moreover, semiconductor logic elements are still in a state of rapid development as distinct from filters, which it seems now improve very slowly. The time is probably not far off when telecommunication engineers will tend to ignore the flip-flop mode of operation and will be concerned with only the bistable logic element in the form of a hermetically sealed multipole.

2.3.2. *Homogeneity of Transmitted Information*

The homogeneity of the pulse-code-modulation information becomes clearly apparent in the increasing resemblance between transmission and switching. The fundamental advantage of this is that information can be switched and transmitted in the same form. This homogeneity appears also in various points of the transmission system.

Thus, the speech information that appears as a series of binary digits can take advantage of all the facilities offered by the digital-information-handling techniques: analysis, memorizing, ciphering, correlation, et cetera.

The choice of bandwidth to provide for a given type of information, such as, telephony, high-quality radio, television, high-speed data transmission, or radar display transmission, can be made automatically by digital means.

The dropping, inserting, and combining of channels are also carried out in a very flexible manner.

3. Transmission in an Integrated Telephone Network

The main purpose of multiplexing is to reduce the number of transmission paths, permitting saving on part of the price of cables to be laid, or avoiding the necessity of laying new cables if those already in service are overloaded.

In the system to be described, conversion from voice-frequency signals to pulse-code-modulated multiplex signals is performed through pulse-amplitude modulation. This solution seems to be the most economical.

It should be noted in Figure 1 that the signals are in the form of pulse-amplitude modulation only as an intermediate modulation inside the concentrator. This avoids any deterioration of the information. Besides, as subscribers' lines are generally very short, variations of the transmission-line equivalent between subscribers will be minimized. It may also make possible the use of a 4-wire subset. Otherwise, the conversion from 2-wire to 4-wire operation is made in the concentrator. The pulse-code modulation in the network is handled entirely on a 4-wire basis.

At the concentration stage, the subscribers' lines are cyclically scanned by pulses supplied

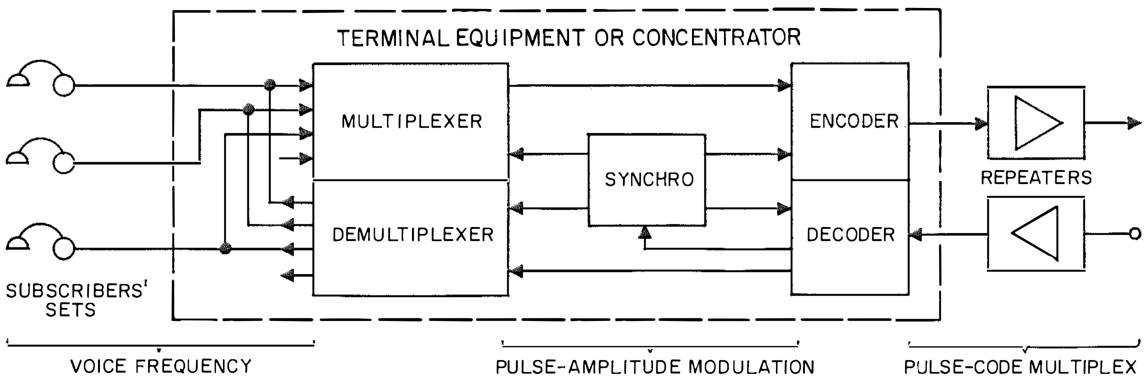


Figure 1—The voice-frequency signals from the subscribers sets are sampled in the multiplexer to produce pulse-amplitude modulation that is then converted to pulse-code modulation in the encoder.

from a timing device, controlled by signals from the exchange to which the concentrator is connected. This control uses the same path over which speech information is sent by the exchange.

The multiplex signal made up of interleaved amplitude-modulated pulses obtained by scanning various subscribers' lines goes to a coder. The coder converts from pulse-amplitude to pulse-code modulation and the multiplex signal is sent to the exchange via a junction equipped with regenerative repeaters. The spacing between repeaters is determined by the quality of the transmission path. It is these pulse-code-modulation junctions and channels that are switched at the exchange. Moreover, the exchange adds to the speech signals other information that enables the concentrator, firstly, to receive the information from the exchange with perfect synchronization, and, secondly, to send the speech signals that are continually coming from the exchange to only the subscribers' lines for which they are intended after they have been demodulated.

The timing device performs certain other operations which will be described in the following article.

4. Multiplexing and Sampling

In an integrated telephone network, the role of the multiplexing element is to concentrate the calls in process at any one time and to sample the speech signals of these active subscribers.

At the present time, multiplexing by interleaving amplitude-modulated samples of the signals to be transmitted, is certainly the simplest and the least-expensive method.

It is sufficient to sample a signal at a rate that is at least twice the highest fundamental frequency contained in the spectrum of the signal. Demodulation consists in sending the pulse-amplitude-modulated samples through a low-pass filter having a cut-off frequency equal to the maximum frequency in the original signal.

For the frequency band from 300 to 3400 hertz, it is theoretically sufficient to sample at 6800 times per second. As low-pass filters become proportionately more costly as they approach the ideal, the sampling frequencies usually chosen for time-division multiplex telephone systems are from 8000 to 10 000 hertz.

The choice of the sampling frequency is governed by considerations of the total cost of the system.

In ordinary multiplex transmission, the number of filters depends only on the number of channels being multiplexed. The tendency is to choose a sampling frequency approaching twice the maximum frequency in the signal to be transmitted, since the bandwidth of the multiplex transmission is proportional to the sampling frequency.

For an integrated telephone network, the number of filters is related to the number of subscribers connected to the concentrator. This number can be 10 times higher than the number of channels being multiplexed. In this latter case, the cost of the low-frequency filtering is an important element in the price of the whole equipment, and the tendency will be toward a high sampling frequency to reduce the cost of the filters.

It is important to note that the cost of the devices that are common to all subscribers in the concentrators or terminals, such as, the coder, decoder, and power supply, is practically independent of sampling frequencies between 8 and 10 kilohertz.

5. Coding and Decoding

The conversion from pulse-amplitude modulation to pulse-code modulation is done in an element called the coder. The convertor for the reverse operation is called the decoder.

The fundamental operation of coding is the quantization of the amplitude-modulated pulses derived from the speech wave. This quantizing operation is fundamentally the measurement of the amplitude of each sample taken from the speech channels. For economy, this measurement must be made during the time between 2 consecutive samplings, which requires that one coder must be able to handle all the samples coming from all the channels that are being multiplexed. For example, if the sampling frequency is 10 kilohertz and 25 channels are to be multiplexed, the coder has 4 microseconds to carry out this operation. It is necessary that this operation be sufficiently precise to permit the faithful reconstitution at the receiving end from the pulse-code-modulated signal of, firstly, a pulse-amplitude-modulated signal and then the original speech signal.

Nevertheless, it must not be too precise, for this will increase the bandwidth and the equipment will be delicate or costly. In fact, the numbers provided by the measurement require more digits, binary or other, as the measurement becomes more precise.

There is no structural difference between a coder for a single channel and one for a multiplexed group of speech channels. The latter coder should simply operate p times faster than the former, p being the number of channels being multiplexed.

It may also be pointed out that the coding-decoding operation produces an inherent type of disturbance in the signal. This results simply from the uncertainty that is inherent in every measuring operation, no matter how precise it may be. The fact that this disturbance is called "quantization noise," "quantization distortion," or even "quantization error" shows clearly that it concerns a phenomenon new to telephony. Fortunately this type of disturbance is normally the only one that affects the system. Moreover, it can be reduced as much as desired, but at the price of increasing the complexity of the equipment. The essential problem is to compromise between an inexpensive transmission system and giving the highest possible quality of service.

The coding of the signals is called linear when all the quantization steps are equal. Let

U = maximum amplitude of the unclipped speech signals

u = amplitude of the weakest speech signal to be transmitted

q = quantization step.

To clarify this idea, assume that the maximum level of the signals U exceeds by 40 decibels that of the minimum transmitted signals u. Assume, also, that the signal-to-noise ratio of quantization must remain higher than 20 decibels. The quantization step q must be smaller than $u/10$ and the necessity of coding the signals of amplitude U requires that the total number of steps be higher than 2000.

For linear coding, the coding precision of weak signals is just sufficient, whereas that of the strong signals is largely redundant. The equipment used is exaggeratedly complicated and very bad use is made of transmission bandwidth.

The advantage of nonlinear coding is that the amplitude of quantization steps may be made to increase with the amplitude of the signal, thus maintaining a constant precision of coding for all amplitudes. This results in a compression of the signal.

5.1 Structure of Coders

The various speech coding devices, proposed and experimented with, can be divided into two categories, those in which the actual coding elements are completely distinct from the compressing elements and those in which these elements are inseparable.

5.1.1 *Compressor Plus Linear Coder*

In Figure 2, the compressor is of the instantaneous type. It looks like a quadripole, the transfer function of which is a decreasing function of the input signal amplitude. The output voltage must depend only on the input voltage at the same moment, otherwise crosstalk will be introduced between the samples that come from different channels and are applied successively to the compressor.

At the receiving end, an expander having a transfer function the inverse of the compressor must, of course, restore the relative amplitudes of the original signals.

The major difficulty of the compression-expansion operation is to be able to reproduce in practice and maintain for a long time the transfer functions of the compressors and the expanders, otherwise distortion will be introduced in these processes.

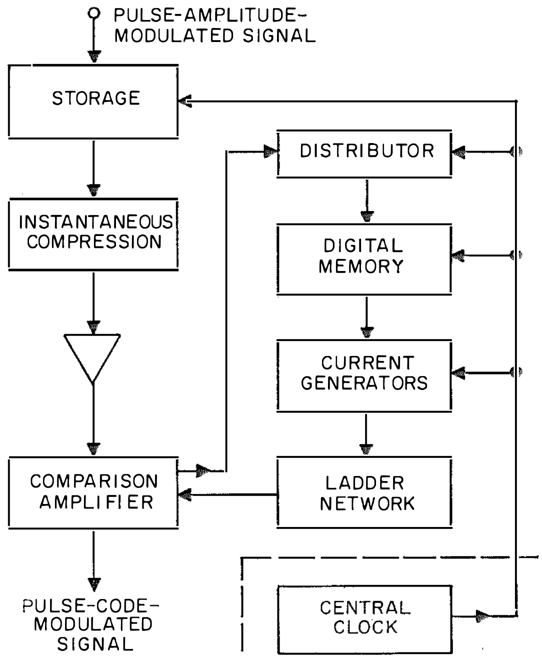
The important role played by compression is already known. The problem arises of determining what transfer function of the compressor will limit the number of digits, which are generally binary, that define the quantization value of a given sample and will also maintain a signal-to-noise ratio that is acceptable over the whole dynamic range. The complexity of the equipment will usually affect its cost and reliability and must, of course, not be neglected in the choice of the transfer function.

Numerous functions: logarithmic, hyperbolic, and exponential, either continuous or discontinuous, have been proposed for compression. The most attractive is the logarithmic function which may be readily obtained from the curvature of the characteristic $v = f(i)$ of a diode. Nevertheless, these devices, which theoretically present the advantage of great simplicity, are very handicapped by variations in the characteristics of individual diodes and by further variations with changes of temperature and finally by their aging.

Another type of compressor uses the abrupt threshold variations in the gain of an amplifier when the amplitude of the input signal exceeds certain predetermined values.

The linear coders associated with such compressors are of the parallel type when the measurement of the pulse-amplitude-modulated sample coming from the compressor is made in one single step. When this measurement is made by successive approximations the coder is called the series type. The number of successive tests in such a method is obviously related to the precision of the measurement. The time allotted to each measurement is fixed: it is the interval that separates 2 successive samplings.

Figure 2—Means of producing pulse-code modulation by using a signal compressor and linear encoding.



The series-type coders thus have elements that work at higher speed than those of the parallel-type coders, but they are fewer in number.

In the linear coder of the series type shown in Figure 2, the signals from the compressor are quantized in the following manner. The signal to be measured is compared successively with the reference signals derived by the coder itself from the results of the preceding comparisons. The results of the successive comparisons appear at the output of the comparator in the form of electric signals, which can take only 1 of 2 values. These results are transmitted in pulse-code modulation and also stored in a digital memory after appropriate switching by a distributor that is controlled by a local clock. The p binary digit obtained at the end of the p first comparison of a measurement cycle initiates a change in the reference signal to make it suitable for the $(p+1)$th comparison. There are 2^p possible values.

The reference signals are obtained by addition in a ladder network of the currents supplied by the generators controlled by the cells of the digital memory.

The average difference between the signal to be measured and the reference signal is halved for each successive comparison. The accuracy of the measurement is thus defined by the number of successive comparisons, which are limited by the number of binary digits that may be transmitted in each code group.

All these operations are synchronized by the local clock, which also defines the cadence of the sampling and multiplexing operations.

5.1.2 *Coder-Compressor*

Only the coder-compressors that use a special cathode-ray tube in which the path of the electron beam is intercepted by a specially cut mask will be cited. Even though this solution is very flexible as far as the type of code is concerned, the disadvantages, due to the use of nonhomogeneous techniques and to the fragility of their electron optics, become all the more prohibitive as the required fineness of the coding of weak signals increases.

Coder-compressors may also use a voltage divider in which the cells are increased with the number of quantization steps. However, this design is quickly limited by the quantity of material to be used. It should probably be reserved for special applications where the norms imposed on the transmission are less severe.

The coder-compressor developed for our use is of a series type. Its structure resembles that of the linear coder described previously. However, its compression characteristic can be reduced to a group of segments of straight lines with different slopes. Such a characteristic can be obtained by purely digital means.

Before describing the coder-compressor, we shall see how it is possible to determine the optimum compression characteristic. In our particular case, this is closely related to the structure of the coder-compressor, which is why this subject has not been discussed previously.

To simplify this study, we shall start from the following hypotheses.

A. The code used is a pure binary code with $n+1$ binary digits and 2^{n+1} steps.

B. Only a certain range of signals will be considered, those that have the same polarity with respect to the potential corresponding to the absence of speech.

C. The compression characteristic will be represented by using, as abscissa, the value x of the signals to be compressed and encoded, the unit chosen being the maximum value of the useful signal, and as ordinate, the ratio y of the measurement given by the coder, to the maximum value.

The signal-to-quantization-noise ratio remains appreciably constant for any signal level, if the ratio of the amplitude of the largest step used to measure a sample, to that of the sample itself, is constant.

For this, it is necessary that the increase of signal Δx corresponding to the increase of a step Δy be proportional to the signal x, that is

$$\frac{\Delta y}{\Delta x} = \frac{a}{x}.$$

The points that correspond to the passage from one quantization step to the following step must then be found on the curve of equation

$$y = 1 + \alpha \log x.$$

The essential differences between such a compression characteristic and the one proposed by H. Holzwarth and studied in detail by Bernard Smith are as follows.

A. The curve described by Holzwarth, which has the following equation in our system of reference,

$$y = \frac{\log(1+\mu x)}{\log(1+\mu)}$$

passes through the origin ($x = 0$, $y = 0$). This is not necessary if the dynamic range of the useful signals is taken into account. This range is obviously limited: in particular, the signals of amplitude lower than the first quantum cannot be transmitted.

B. The characteristic of $y = 1 + \alpha \log x$ presents the advantage over the preceding equation of corresponding to a signal-to-quantization-noise ratio that is constant over the entire useful dynamic range, whereas this ratio deteriorates about 6 decibels for weak signals by the second type of logarithmic compression.

If we now consider the ratio of the slopes of the chords that subtend 2 adjacent arcs having respectively, as ends, the points of ordinates

$$y_i + (k/2^n), \quad y_i, \quad \text{and} \quad y_i - (k/2^n)$$

that is, corresponding to successive quantization step groups that have the same number of steps, it is seen that

$$\rho = \frac{x_{i+k} - x_i}{x_i - x_{i-k}} = 10^{k/2^n\alpha}.$$

This ratio is independent of i, that is, if we choose a compression characteristic formed of segments of a straight line subtending, as previously, the logarithmic characteristic of equation

$$y = 1 + \alpha \log x$$

the signal-to-quantization-noise ratio at the corresponding points of these segments does not vary over the entire useful dynamic range. It is nevertheless obvious that this ratio varies along each segment and increasingly so as the number of segments becomes smaller.

We shall now establish the relations that exist between the factor α, which determines the rate of the logarithmic compression, and the parameters of the pulse-code-modulation transmission, that is:

A. The number of digits used for the coding, $n + 1$.

B. The ratio of the amplitude of the signal to that of the largest step used to make the measurement p.

C. Finally, the useful dynamic range, which is defined as the ratio A of the amplitude of the unclipped maximum signal to that of the smallest signal judged useful for coding with the precision that corresponds to the ratio p.

The ratio p must remain constant over the entire useful dynamic range.

The smallest signal of this useful dynamic range, which must not be confused with the smallest useful signal, has in our system the amplitude

$$x = 1/A.$$

It is natural to consider that there is no compression for the signals that are weaker than those of the useful dynamic range. This is another way of saying that the coding of the very weak signals is linear.

Actually, this is the case with most of the existing systems. The ordinate that corresponds to the abscissa point $x = 1/A$ is then $y = p/2^n$.

Hence the first relation

$$p/2^n = 1 - \alpha \log A.$$

The first step of the useful dynamic range is defined by

$$\Delta x_1 = x_2 - x_1 = x_1 \quad (10^{1/\alpha 2^n} - 1)$$

$$\Delta y_1 = 1/2^n$$

the ratio $x_1/\Delta x_1$ being equal to p, we obtain the second relation

$$1/p = 10^{1/\alpha 2^n} - 1$$

or even

$$\frac{1}{\alpha} = 2^n \log \left(1 + \frac{1}{p}\right).$$

Finally, we can deduce the relation between the useful dynamic range A, the number of digits $n + 1$, and the ratio of signal-to-quantization noise p

$$2^n = p + \frac{\log A}{\log (1 + 1/p)}$$

which can also be written

$$2^n \approx p(1 + 2.3 \log A). \qquad (1)$$

It is seen for example that the use of natural binary codes with 7 digits ($n = 6$) makes it possible to obtain, if logarithmic compression is used, a signal-to-quantization-noise ratio that is constant and equal to about 22 decibels inside a useful dynamic range of 36 decibels. Under these conditions, the ratio of the strongest unclipped signal to the weakest coded signal is about 60 decibels.

We shall now examine the deterioration in signal-to-quantization-noise ratio when the logarithmic characteristic is replaced by a variable number of chords that subtend its arcs.

The deterioration observed over the range of a knee of this new characteristic is given by the value of the ratio of the slopes of the segments that form the knee. This ratio is, as we have seen,

$$\rho = 10^{k/2n\alpha}$$

or even

$$\rho = 10^{k \log (1+(1/p))}.$$

The deterioration of the signal-to-quantization-noise ratio that occurs over a knee is then

$$20 \log \rho \approx 8.7\ (k/p), \text{ decibels.} \qquad (2)$$

The order of magnitude of this deterioration is therefore $k/2$ decibels. This signifies that the use, for a given polarity, of a characteristic having 4 straight-line segments subtending a logarithmic characteristic leads to a slightly greater deterioration than that which corresponds to the loss of the smallest binary digit. On the other hand, the use of a characteristic having 8 segments leads to a lesser deterioration than that which corresponds to the loss of this digit.

The coder-compressor that was used in our first experiments gives a compression characteristic that is comprised of 5 segments in all.

The central segment has 64 quantization steps and corresponds to an improvement of 25 decibels of the signal-to-quantization-noise ratio for weak signals. The 4 other segments each have 16 quantization steps.

A schematic diagram of the device used is shown in Figure 3. It does not include an analog-type of instantaneous compressor ahead of the linear coder. The operation is completely digital, and is in all ways like that of an ordinary linear coder. The current generators are naturally more numerous since it must be possible to perform the coding of the last digits according to 3 different laws depending on the amplitude of the sample measured.

The greatest difference results from the introduction of a decoding matrix between the code store and the current generators. This matrix makes it possible at the time of each comparison and according to the results of the previous comparisons to determine which reference signal range, supplied by the current generators, should be used for the rest of the coding.

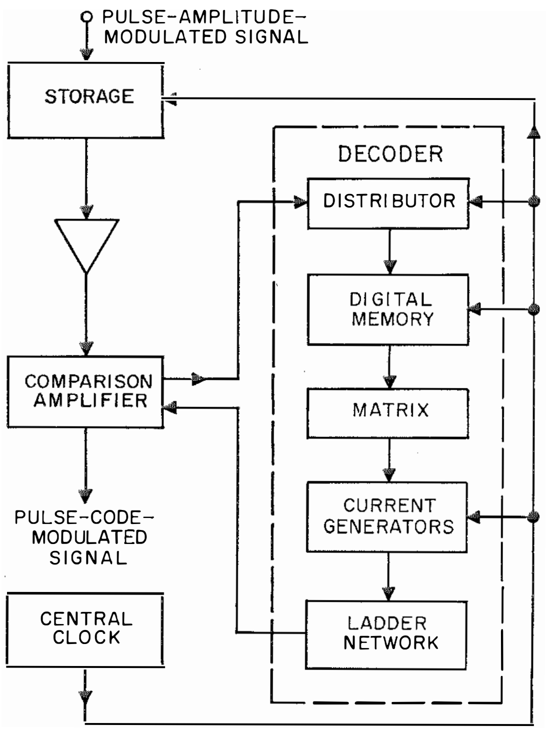


Figure 3—Digital method of producing pulse-code modulation with nonlinear quantization.

From the technological point of view, the only new difficulty concerns increasing the sensitivity of the comparison amplifier. This increase in sensitivity is, moreover, equal to the improvement in the signal-to-quantization-noise ratio for the weak signals.

5.2 Decoding

The decoder is the element that permits the reconstitution of multiplex signals from the codes.

A noticeable symmetry exists between the decoding and coding elements. In fact, the very device that permits the derivation of the reference signals for the comparisons, constitutes a decoder. This device is represented in Figure 3.

The decoding proper does not present any particular difficulty.

The information is almost always transmitted in series; in other words, the different digits of each code appear successively at the input of the decoder. The highest-valued digits generally precede the lowest and, in linear coding using the pure binary code, each digit has a value that is double that of the following digit and half that of the preceding digit.

There are two different ways of using the information that arrives continually at the input of the coder. Either accumulate the information in a digital memory and do the decoding all in one step or translate separately each one of the elements of information as it arrives. In the latter case, the accumulation or integration of the complete code information is made in an analog memory. This analog memory can either be at the level of the common pulse-amplitude-modulation amplifier that precedes the demultiplexer, or it can belong to each one of the channels. The filter that permits the reconstitution of the low-frequency information from the demultiplexed amplitude-modulated pulses, can be used for this.

There is a major disadvantage to using an analog memory for decoding. The energy sent into this analog memory is proportional to the time during which the memory is connected to the decoder; that is, any modulation of the width of the control pulses of each partial integration results in a stray modulation of the demodulated signal.

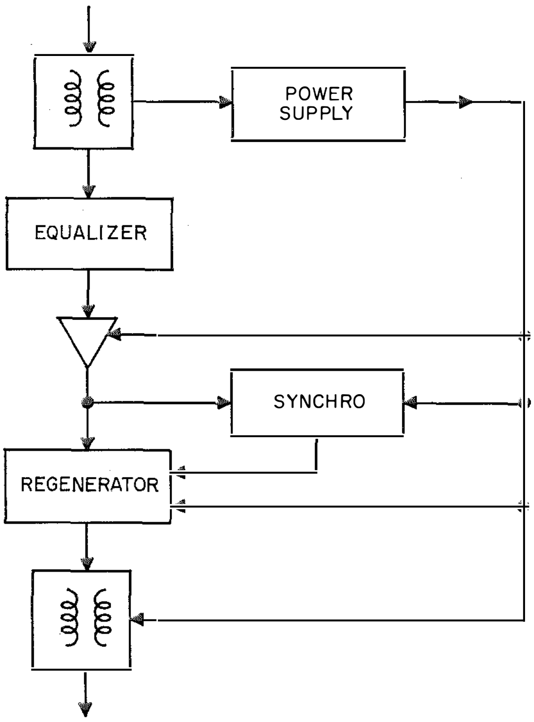
The situation is the reverse if decoding is effected as a whole. Then the energy transferred depends on the width of a signal control pulse and it is even possible to conceive of devices where to a certain extent the width and the shape of this control pulse have only a secondary influence on the demultiplexing.

In our decoding equipment, accumulation is in a digital memory identical to the one in the coder.

6. Amplification and Demultiplexing

In a time-division multiplex telephone equipment that is an integral part of a network, it is not possible to consider separately the different operations that follow the decoding, namely: pulse-amplitude-modulation, demultiplexing, and filtering, combined or not with a low-frequency amplification. In fact, more than anywhere else, the economic problems are closely connected to the technical problems. The fluctuations of the prices of the different electronic components are such that any solution can rapidly become obsolete. It must not be forgotten that in an integrated telephone network a single coder can serve more than 200 subscribers. The total amplification necessary for reception can be obtained through an amplifier common to 200 subscribers. Considerations of the over-all price of demultiplexing gates and filters suitable for each subscriber may make desirable the addition of a rudimentary low-frequency amplifier to each line equipment. Moreover, protection may be thus provided to the speech signal against crosstalk introduced into the part of the equipment that handles pulse-amplitude modulation.

Figure 4—Regenerative repeaters for pulse-code-modulation signals.



Each one of the line equipments used in our system has a low-frequency amplifier. A regenerated repeater for pulse-code-modulation signals is shown in Figure 4.

7. Signalling

The signalling problems in an integrated telephone network using pulse-code modulation are closely related to the switching problems. Neglecting speech signals, the information that passes between the subscriber and the exchange through the concentrator, such as: dialling, supervision, ringing, and of course various other tones, is routed in the same way as the speech signals. As far as the coding, decoding, multiplexing, and demultiplexing are concerned, these signals are not separated from the speech signals. Let us recall that the subset used is a 4-wire set with a transistor amplifier in the ringer.

8. Synchronization

To recognize the beginning of each group of pulses, it is necessary to transmit synchronization codes of the same nature as the speech codes but which have a constant structure and are repeated cyclically. For example, the repetition cadence of these codes can be equal to the sampling frequency of the speech signal. At the receiving end, a digital discriminator makes it possible to distinguish between the synchronization code and other codes, and a correlating device, also digital, assures that it is indeed the synchronizing signal and not a signal formed, for example, by the end of one speech code and the beginning of the following code. The separation of the synchronization code from the range of speech codes is not an a priori requirement.

The synchronization makes it possible to recognize the beginning of each code group, that is, the rank of each unit of binary information received, as well as to determine the number of the multiplexed channel corresponding to this code. The synchronization device possesses a certain inertia so that the accidental destruction of a single unit of information of a synchronization code would not cause the loss of useful information transmitted during the following cycles. This inertia, supplied by a digital memory, is closely associated with the correlation used in searching for synchronization at the start of operation or after accidental loss of synchronization.

It is also to be noted that, on the scale of binary information, a finer synchronization is necessary. The propagation-time variations, the crosstalk between the different multiplex highways or between these and other transmission systems, the fluctuations of the supply sources, and noises of various origins, have repercussions on the phase of the pulses at the input of the exchanges or the concentrators, as well as at the input of the repeaters. The problem of synchronization in an integrated and mesh telephone network using pulse-code modulation has been solved by using digital memories. This solution will be developed in the paper dealing with switching.

9. Conclusion

The use of pulse-code modulation in an integrated telephone network offers the considerable advantage of making the transmission quality independent of the distance and of the network configuration. The transmission proper introduces practically no additional crosstalk, distortion, or noise. The transmission-line equivalent remains absolutely stable.

The introduction of pulse-code-modulation into telephony offers, moreover, other advantages that, even though unknown to the subscriber, are none the less real. Thus, the elimination of the numerous, costly, and delicate filters used in frequency-division multiplex telephone systems involves substantial savings in volume, weight and price of equipment. Moreover, the use of a single technique for both transmission and switching should lead to the disappearance of the radical differences that exist at present between these two fields of telecommunications.

Andre Chatelon was born in Versailles, France, on 2 November 1928. He received the license ès Sciences from the University of Paris and graduated as an engineer from the Ecole Polytechnique of Paris in 1951.

In 1953, he joined Laboratoire Central de Télécommunications and was in charge of research and development in the field of electronic digital computers until 1958. He is now head of the transmission department.

Mr. Chatelon is a member of the Société Française des Electroniciens and of the Institute of Radio Engineers.

Application of Pulse-Code Modulation to an Integrated Telephone Network

Part 3—Switching

J. LE CORRE

Laboratoire Central de Télécommunications; Paris, France

1. General

The application of pulse-code modulation to telephone switching leads to an examination of the 3 main points.

A. Design of a switching stage connecting voice-frequency subscribers' lines to a pulse-code-modulation multiplex highway.

B. Design of a switching stage for interconnecting pulse-code-modulation multiplex highways.

C. Synchronization of the coded and multiplexed speech signals at the input of each exchange to which they are transmitted.

When these 3 problems are solved, the development of the speech circuits in the various types of exchanges through which a call may be routed is straightforward.

The control circuit of a pulse-code-modulation exchange does not raise any particular problem. The interconnections with the speech circuit will be facilitated by the multiplex operation of the latter.

The experimental system, developed to test pulse-code multiplex operation of a telephone system that will be described, used a 7-digit code with a sampling rate of 10 kilocycles per second to provide 24 channels on a 4-wire basis.

2. Speech Circuit

Two types of switching stages and the synchronization circuits will be examined.

2.1 Subscribers' Lines and Concentrator

The switching stage connecting the subscribers' lines to a pulse-code multiplex highway may be in a concentrator near the subscribers and at a distance from the exchange. The 24 channels of the multiplex highway allow 24 simultaneous conversations. The number of lines that can be served depends on their traffic; for average traffic, about 150 subscribers' lines can be connected to a concentrator, 200 or more for light traffic.

2.1.1 *Speech Circuit*

Figure 1 is a simplified representation of the concentrator, which is connected to the ex-

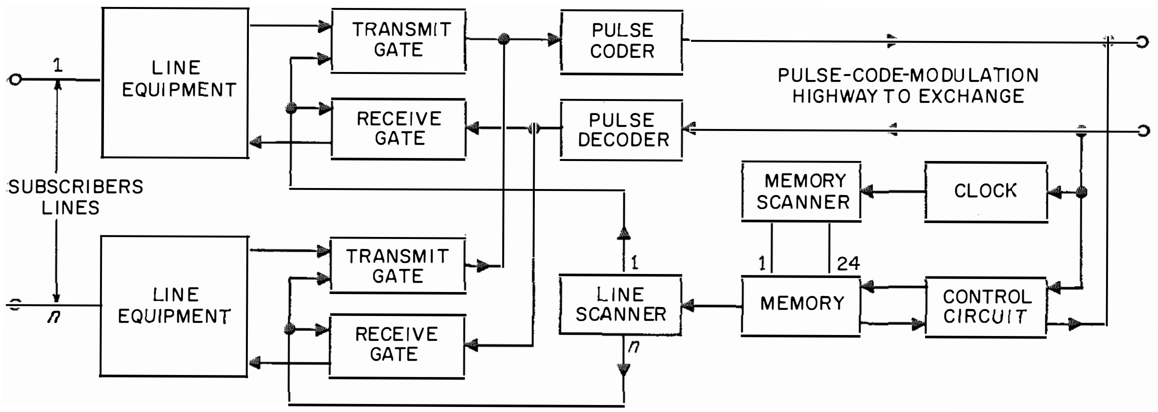


Figure 1—Concentrator for subscribers' lines.

change on a 4-wire basis over 2 telephone pairs. These pairs terminate in a coder and a decoder.

Under control of the line scanner, each active subscribers' line, up to 24 in number, will be connected for 4-microsecond time intervals to the coder and the decoder, giving that subscriber access to the exchange.

The identities of the subscribers in conversation are recorded in a memory having a line for each of the 24 possible connections. Each memory line will record 8 digits to identify in coded form any one of 255 subscribers.

Every 100 microseconds, at time t_n corresponding to connection n, row n of the memory is read. The identity of the subscriber to be connected is transmitted to a line scanner, called the decoding matrix, each output of which controls the 2 gates associated with a subscriber.

The continuous signals corresponding to the state of the line (receiver on hook or unhooked), as well as the dialling signals, are transmitted through the same path as the speech signals.

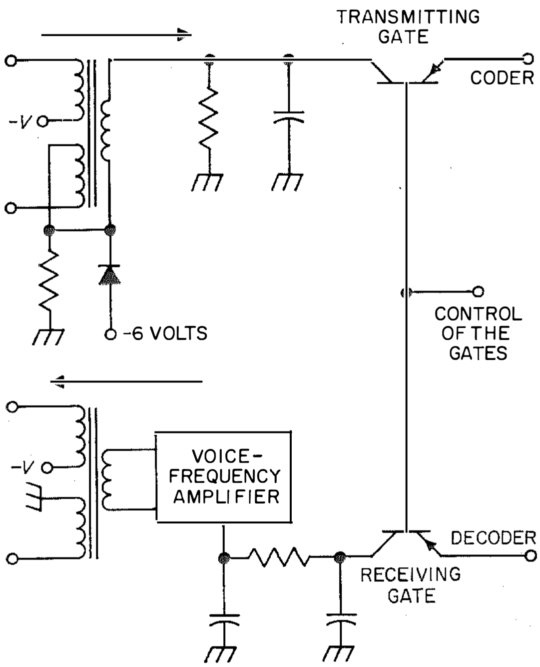


Figure 2—The 4-wire solution of the subscribers' line equipment.

2.1.2 *Control Circuit*

It is economical to perform in the concentrator only the minimum of logic functions, since they can be fulfilled in the exchange by circuits common to several concentrators.

The only functions required from the concentrator are the establishment and release of a connection between a subscriber and a given channel. These controls are accomplished in the concentrator by writing or erasing information on a memory line.

For the passage of information between the concentrator control circuit and the exchange, use is made of the same channel as that for speech.

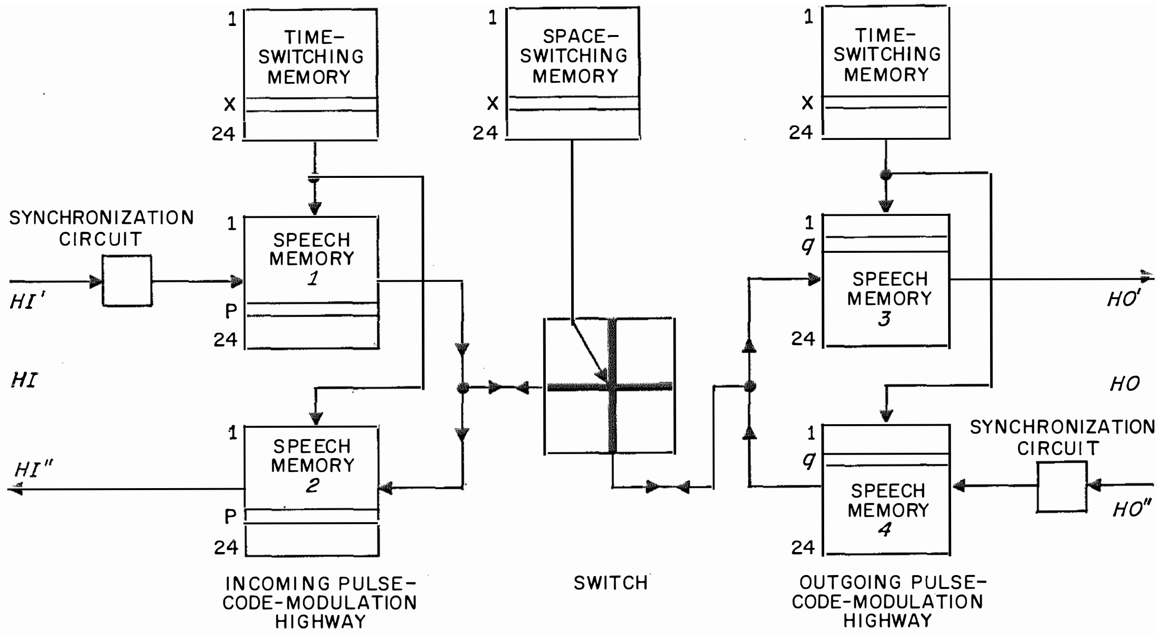
To connect a subscriber to a channel, the exchange sends the code of this subscriber to the concentrator, at the scanning times assigned to the channel. The logic circuit of the concentrator tests the subscriber's line (free or busy) by ensuring that the code of the subscriber is not already written on one of the memory lines, sends the corresponding information to the exchange, and establishes the connection if the called subscriber is free.

For the release of a connection, the exchange sends a release code to the concentrator at the scanning time assigned to the connection to be released.

As digital information can be transmitted between concentrator and exchange at a rate of 70 000 binary digits per second for each channel, the transmission of orders can be performed with great redundancy ensuring very reliable remote control of the concentrator.

The detection of a calling subscriber is controlled from the exchange. Through the control circuit and memory of the concentrator, the line scanner is made to connect each subscriber to the exchange through a free channel, one after

Figure 3—Switching stage between incoming highways through coupled outgoing highways.



the other, thus assuring the connection of all the subscribers to a call-detection circuit, which then knows the state of each line and can recognize a new calling subscriber. All lines are tested about 4 times per second.

2.1.3 *Line Equipment*

The equipment proper to each subscriber's line deserves particular attention. Its influence on both cost and volume of the whole equipment is very important; the other devices of the concentrator are common to all subscribers. One solution, a 4-wire arrangement of line equipment, is given in Figure 2.

Since the use of concentrators reduces considerably the length of the subscribers' lines, the utilization of 4-wire lines may become possible, thus permitting a simplification of the subscriber's set (antisidetone circuit) and of the subscriber's line equipment.

As the sampling is made through 2 different gates, 1 for each direction, these gates are of a simple design employing only 1 transistor of alloyed type per gate.

The transmitting gate operates at a low level compatible with a correct level of crosstalk. A single-transistor voice-frequency amplifier following the receiving gate allows it to operate at a level equivalent to that of the transmitting gate. The control of these gates is thus simplified.

The use of a relatively high sampling frequency of 10 kilohertz permits a simple design for the filtering circuits.

The 2 pairs of the subscriber's line terminate in 2 transformers permitting the transfer from 2-wire circuits on the subset side to 1-wire circuits on the concentrator side.

The average voltage of the signals sent to the coder is −6 volts when the receiver of the subset is unhooked. In this condition, the diode connected to −6 volts is conductive. The voltage drops to about 0 when the receiver is hooked.

Numerous other solutions are possible for the type of line equipment described. In particular, the replacement of each transformer by a symmetrical circuit having 2 voice-frequency transistors may be considered. These transistors, which are already inexpensive, make it possible to reduce the volume of the equipment.

In the case of 2-wire subscribers' lines, the use of a differential transformer for the transfer from 2 to 4 wires seems to be the most economical solution.

The sampling of the conversation is then made by 2 different gates as in the case of a 4-wire line.

A voice-frequency ringer with amplification in the subscribers' set is required so that ringing signals can be transmitted through the speech circuit.

2.2 Switching Stage Between Pulse-Code Multiplex Highways

Any pulse-code-modulation exchange will be connected to other exchanges or concentrators by multiplex highways. The connections between the various channels of these highways are made by a switching stage connecting the so-called incoming highways to the so-called outgoing highways. A description of this switching stage was given in Part 1 of this article.

Figure 3 shows, in a simplified manner, the elements acting in a connection between an incoming multiplex highway HI and an outgoing highway HO.

2.2.1 *Space Switching*

A space-switching stage having as many inputs and outputs as there are, respectively, incoming and outgoing highways, operates in time division. During any one of the 24 time intervals of 4 microseconds each repeated every 100 microseconds, it can connect any input to any output and the connections made can have any configuration whatever.

In the 4 microseconds during which an input is connected to an output, 2 coded speech samples relating to one conversation are transmitted, 1 in each direction.

The 7 digits of a sample are transmitted successively in time. The speed of the switch makes it possible, during the 0.5 microsecond allotted to a digit, to transmit 1 bit of information in each direction. The control of the connections in the switch is made from a space-switching memory. One of these memories having 24 rows is allotted to each incoming highway. During 1 of the 24 time intervals, t_x for example, row x of this memory is read. On this line is written the coded identity of the output to be connected to the incoming highway; a decoding matrix controls the crosspoint corresponding to the output.

2.2.2 *Time Switching*

Each highway, incoming HI and outgoing HO, is equipped with 2 speech memories, one for transmit and one for receive, and with a time-switching memory.

For example, the coded speech signals arriving at the exchange on HI' are written on the speech memory 1 through the synchronization circuit. The 7 digits of channel 1 are written on line 1 of the memory, those of channel 2 on line 2, et cetera.

The synchronization circuits needed for this operation will be described later.

The coded speech signals leaving the exchange on HI'' are extracted from speech memory 2. At time t_1, the information corresponding to channel 1 is read in row 1; at t_2, of channel 2; et cetera.

The reading of speech memory 1 and the writing in speech memory 2 are controlled by a time-switching memory similar in operation to the space-switching memory. If, for example, at time t_x, channel p of the incoming highway is handled through the switching stage, the coded identity of row p is written in row x of the incoming time-switching memory.

The readout at time t_x of this row of the incoming time-switching memory causes the readout of row p of speech memory 1, the speech codes there registered are transmitted through the switch toward the outgoing highway, and also causes the writing of the information coming from the switch into row p of speech memory 2.

The outgoing highway equipment functions in the same way as that of the incoming highway.

This type of switching stage thus makes it possible to handle a conversation inside the exchange at a time that is completely independent of its timing on the incoming and outgoing highways. In principle, this can be applied to any type of pulse-code multiplex exchange of the local network. But, as will be seen below, appreciable simplification of the equipment allotted to a highway is possible in many cases.

The establishment of a conversation through the switching stage involves the alignment of the time positions on the highways concerned. Consider, for example, a connection that is to be established between channel p of incoming highway HI and channel q of outgoing highway HO. The marker circuit of the switching stage begins the search for a time position that is simultaneously free on the input of the switching stage HI and on the output HO.

Let t_x be a free time. The connection is then established by writing, on row x of the 3 switching memories, the following information:

Code for channel p in the incoming time-switching memory.

Code for channel q in the outgoing time-switching memory.

Code for the outgoing highway in the space-switching memory.

The release of the connection is made by erasing this information.

In certain types of exchanges, it is necessary to make connections between incoming highways or between outgoing highways. This is particularly the case for a local exchange, the incoming highways of which serve concentrators. The local traffic between subscribers necessitates connections between incoming highways. These connections are made by local highways HL, each 1 comprised of 2 coupled outgoing highways HOA and HOB as shown in Figure 4.

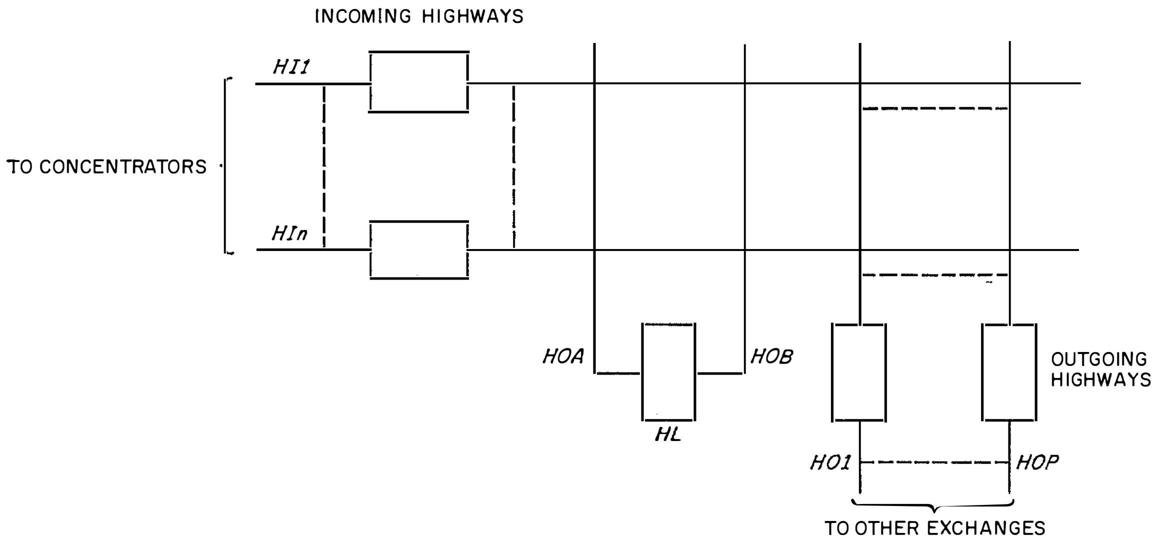


Figure 4—Highway diagram.

2.2.3 Simplified Highway Equipment

The memory equipment associated with each pulse-code multiplex highway at the input in the exchange can be simplified in many cases.

If subscribers' lines are connected to the exchange directly, the general organization of the switching stages interconnecting these subscribers to pulse-code multiplex highways remains the same as that of the concentrator shown in Figure 1.

For a pulse-code highway serving a group of subscribers, it is then possible to directly connect the outputs of the pulse coder and decoder to the input of the switch. No speech memory is necessary; the identity memory of the subscribers in conversation serves the same purpose with respect to the control circuit of the exchange as the incoming time-switching memory of Figure 3.

The equipment of a multiplex highway serving a remote concentrator can also be reduced. In comparison with the preceding case, where the stage of connection to the subscribers was in the exchange, it is sufficient to compensate for the propagation times between the exchange and the concentrator. For this, a supplementary delay is added at the input of the exchange to the information coming from the concentrator so that the total delay will be a multiple of 100 microseconds. This additional delay can be provided by a matrix memory of the same type as the speech memories or by a circulation memory such as a delay line.

Simplification is also possible for local highways such as shown in Figure 4. The 2 pulse-code multiplex highways constituting this local highway require only 1 time-switching equipment (2 speech memories and 1 time-switching memory).

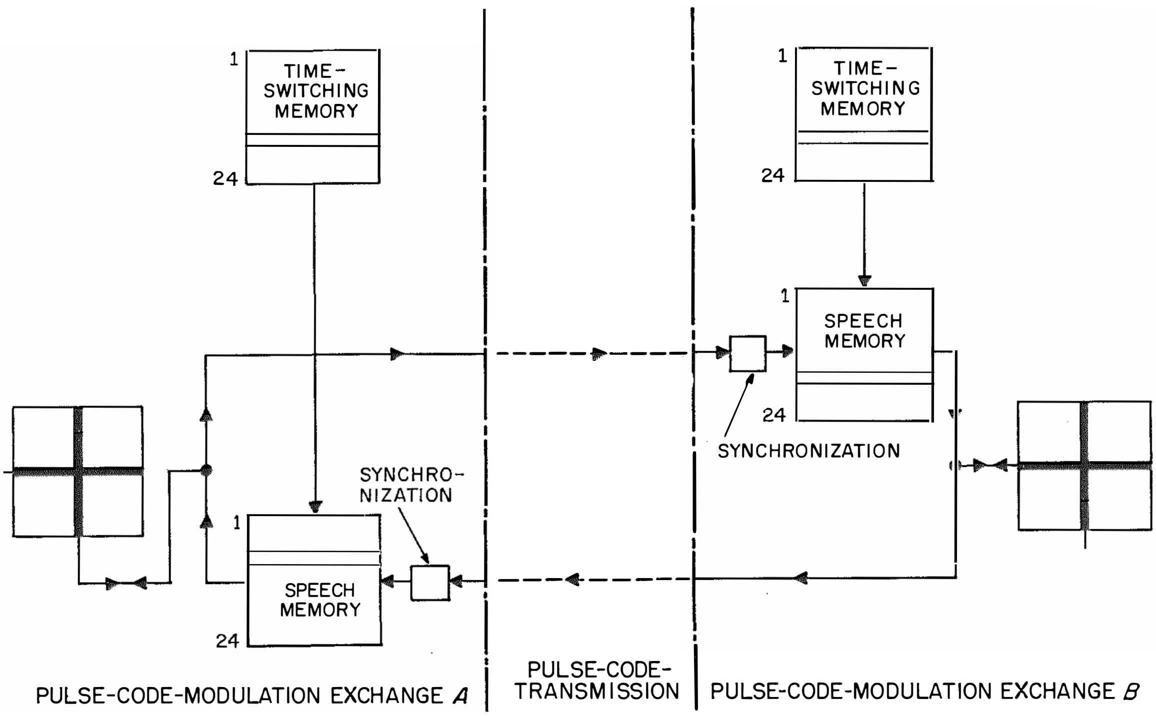


Figure 5—Highway connecting 2 pulse-code multiplex exchanges.

It may be noted that a pulse-code multiplex highway connecting 2 pulse-code exchanges of the type shown in Figure 3 has in each direction of transmission 2 speech memories, 1 in the sender and the other in the receiver exchange. As has been shown, this design permits complete independence of the 2 exchanges and of the transmission circuits.

The deletion of 1 memory is possible provided certain conditions be fulfilled between the control circuits of the exchanges.

Figure 5 shows one possible design. A single speech memory is placed in each exchange at the input of the information, which provides for ready synchronization.

Let us consider a call passing from exchange A to exchange B. The time position chosen by exchange A to handle the call will be used for the transmission direction from A toward B. If exchange B must use a different time for this same call, it must inform exchange A of the change so that A can proceed to the desired time shift for the information arriving from exchange B.

If, in the rest of the routing of the call, connection modifications are necessary in exchange B as at the end of dialling, for routing problems, et cetera, exchange A must be informed of these modifications.

Thus the saving on speech memories requires supplementary exchanges of information between central offices.

2.3 Synchronization Equipment

The principle of the synchronization circuits was described in the first article.

The problem consists in writing the coded speech signals of the 24 channels of a pulse-code multiplex highway in their correct place in the speech memory associated with the highway at the input of the exchange.

These signals may undergo time fluctuations because of imperfections of the transmission circuits and differences in the frequencies of the exchange clocks.

Figure 6 shows the circuits providing for synchronization of the information.

The writing and reading of information in the speech memory are done successively. The time allotted to a digit of 0.5 microsecond is divided into 4 elementary times, the first being

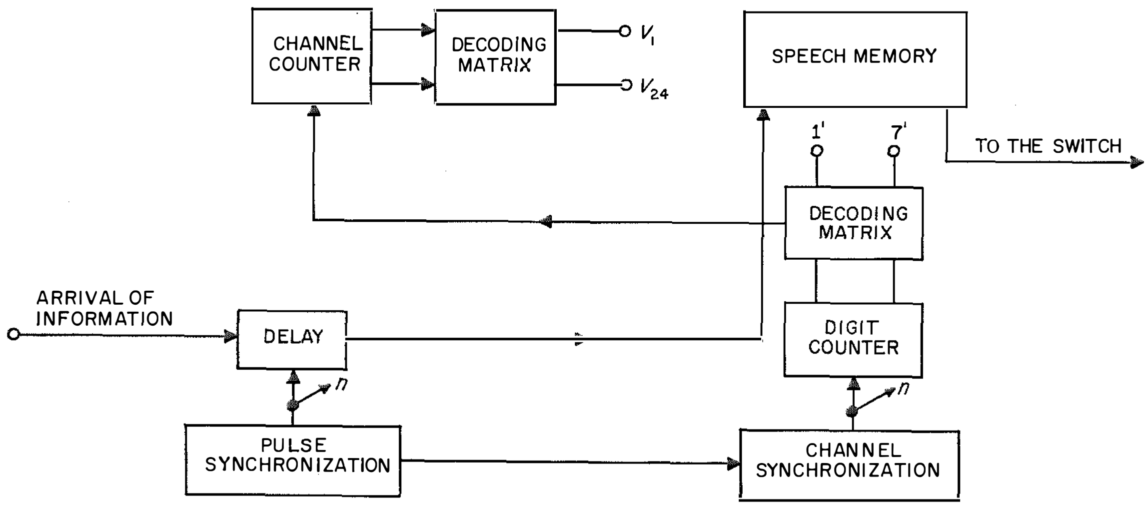


Figure 6—Sychronization circuits.

allotted to the writing in the memory, the third to the readout, the two others being used to select the points to be written or read.

The bits of information arrive in the exchange at any time; it is first necessary to individually fix each one of them in time before writing them in their place in the memory. This leads to dividing the synchronization circuits into two parts.

2.3.1 *Pulse Synchronization Circuit*

The incoming information is delayed by a variable time of less than $\tau = 0.5$ microsecond so as to arrive at the input of the memory at the intended elementary time.

The delay, which is produced by flip-flops, varies in a discontinuous manner and can take the 4 following values: 0, $\tau/4$, $\tau/2$, and $3\tau/4$.

The adjustment of delay is made by a pulse synchronization circuit common to several multiplex highways. This circuit continually checks the time delay of each highway. To do this, it compares the arrival time of the information on a highway to that of the time pulses delivered by the exchange clock and then estimates from this the necessary delay on this highway.

When the delay on a given highway increases from $3\tau/4$ to τ, the addressing circuits for writing the information in the memory are modified and the delay introduced in the highway is returned to 0.

In the same way, when the delay is 0 and the information arrives too early, the addressing circuits of the memory are modified and a delay of $3\tau/4$ is introduced into the highway.

2.3.2 *Channel Synchronization Circuits*

The address of the memory point where the information must be written, at a given time, is derived from 2 counters; 1 counts by 8 and gives the identity of the column of the memory and the other counts by 25 and gives the identity of the row of the memory.

These counters operate with a certain time shift with respect to the exchange clock. This corresponds to the shift of the synchronization channel with respect to the local time t_{25}.

A channel synchronization circuit common to several multiplex highways also works in time division. It is successively connected to all the multiplex highways and serves two purposes.

It checks for correct synchronization of the 2 counters in the following manner. When the channel counter reaches the count of 25, it checks that the incoming information conforms with the synchronization code. A decision is taken only after 3 successive checks to avoid the possibility of an error due to the transmission circuits. The counters are resynchronized when their phases are wrong.

It modifies the addressing of the information in the memory when the pulse synchronization circuit has to change the delay from $3\tau/4$ to 0 or vice versa. This modification consists of putting the digit counter of the highway concerned forward or backward by 1 time unit.

The synchronization problems are simpler when the clocks of the 2 switching equipments connected by a multiplex highway are synchronized. The connections between concentrators and exchanges are of this type. The synchronization of the concentrator consists of correctly setting the counters with respect to the synchronization channel, and then no delay is necessary.

3. Evaluation of Possible Traffic Improvement In Highway Efficiency

3.1 Traffic Possible by Time Position Alignment

The establishment of a connection through an exchange necessitates, as shown in 2.2.2, the existence of a time position that is simultaneously free on the input and on the output to be connected. Therefore, a certain internal blocking results in all time-division switching systems.

Computations have been made of the traffic possible on time-division multiplex highways [3–5].

In the case of 24-channel multiplex highways, the possible traffic on a highway is about 12 erlangs, admitting 1 lost call out of 100. The corresponding efficiency of the highways is 50 percent.

As an example, let us consider an average traffic per subscriber at the busy hour of 0.1 erlang. Each concentrator can then serve 120 subscribers. A local exchange of 9000 subscribers will interconnect 75 incoming highways arriving from the concentrators and 75 outgoing highways going to other exchanges or serving local traffic.

Elimination of internal blocking due to time-position misalignment is possible by applying a rearrangement operation [6, 7].

3.2 Principle of Rearrangement

If we consider, in a switching stage between pulse-code multiplex highways, a connection to be established between an incoming highway HI having only the time position t_x free and an outgoing highway HO having only the time position t_y free, the problem cannot be solved without rearrangement.

A rearrangement is then made by releasing the connection that occupies the time position t_x of HO, which makes it possible to establish the desired connection, then by re-establishing the connection that was just released at time position t_y. This re-establishment may perhaps require releasing another connection, which will be itself re-established at another time position, and so forth until a free path is found for each connection.

Take, as an example, a 5-by-5 switch interconnecting inputs 1, 2, 3, 4, 5 to outputs *a, b, c, d, e.* Each input and each output serve 1 pulse-code multiplex highway (see Figure 3).

Column *A* of Table 1 shows the connections between the inputs and outputs during t_x and t_y, 2 of the 24 time positions. At time t_x, input 1 is connected to output *b,* input 2 is free, input 3 is connected to output *d,* et cetera; at time t_y, input 1 is connected to output *d,* et cetera.

TABLE 1
SUCCESSIVE ARRANGEMENTS

A			B			C			D			E		
	t_x	t_y		t_x	t_y		t_x	t_y		t_x	t_y		t_x	t_y
1	b	d	1	b	d	1	b	d	1	b	*d*	1	**d**	b
2		a	2		a	2		a	2		a	2		a
3	d	e	3	d	e	3	d	*c*	3	**e**	**d**	3	e	d
4	a		4	a	**c**	4	a	c	4	a	c	4	a	c
5	e	c	5	e	*c*	5	**c**	**e**	5	c	e	5	c	e

Let us assume that a new connection is to be established between input 4 free at time t_y and output *c* free at time t_x, this output not being connected to any input at this time.

The connection 4-*c* cannot be made without modification of the connections already established.

The connection 5-*c* existing at time t_y is released. This change makes connection 4-*c* possible at time t_y. Column B shows the connections that are then made.

The connection 5-*c* is then re-established at time t_x by releasing the connection 5-*c,* which is itself re-established at time t_y by releasing the connection 3-*e* (Table 1, Column *C*).

The same process, consisting in permuting the time positions t_x and t_y during which 2 connections are established, is continued.

The connections 3-*e* and 3-*d* are permuted in time, the connection 1-*d* being released (Column *D*).

The connections 1-*b* and 1-*d* are permuted (Column *E*).

The re-establishment of the connection 1-*b* at time t_y does not necessitate any release, the output *b* being free at time t_y. The rearrangement of the connections is finished.

The process just described can be generalized for any size exchange. It is easily shown that the number of connection permutations always remains lower than the number of inputs or of outputs of the exchange.

Calculations have been made of the average number of connections that must be modified in order to establish a new one for various sizes of multiplex switching stages. This number is relatively low; in the most pessimistic cases, it is necessary to modify on the average one conversation to establish a new one.

Application of the rearrangement method does not introduce any additional equipment into the speech circuits of a pulse-code-modulation exchange; only a slight increase is required in marker equipments.

When the connections between pulse-code-modulation exchanges are designed as in Figure 5 (1 speech memory in each exchange; see paragraph 2.2.3), any permutation of time on a highway required by 1 exchange for a rearrangement must be made known to the corresponding exchange.

On the other hand, if the highways are equipped with 2 speech memories, as in Figure 3, the rearrangements are made independently in each exchange.

3.3 Traffic Possible by Rearrangement

Since the rearrangement method eliminates internal blocking in switching stages between pulse-code multiplex highways, the efficiency of these highways depends only on the amount of traffic handled in 1 direction.

A highway connecting a concentrator to an exchange constitutes a trunk of 24 channels, which can handle a traffic of about 15 erlangs, permitting 1 lost call out of 100.

Assuming traffic of 0.1 erlang per subscriber at the busy hour, the concentrator could serve 150 subscribers. A local exchange of 9000 lines would have under these conditions 60 incoming highways.

The efficiency of the outgoing highways handling the total traffic, that is, 900 erlangs, depends on the number of directions in which this traffic is handled. Efficiencies of the order of 80 percent can be expected, which leads to 47 outgoing pulse-code multiplex highways.

It will be recalled that in the case of an exchange operating by time-position alignment (paragraph 3.1) having the same number of lines and handling the same traffic, 75 incoming highways and 75 outgoing highways were necessary.

The rearrangement method therefore allows an important saving in equipment, as much from the point of view of transmission as from the point of view of switching.

Improvement in the efficiency of the highways is possible by other methods than that described. Notably, the increase in speed of electronic elements makes it possible to envisage multiplexing a greater number of channels on the inputs and outputs of the space-switching stage; for example, 48 instead of 24. The blocking due to the time-position alignment is then decreased.

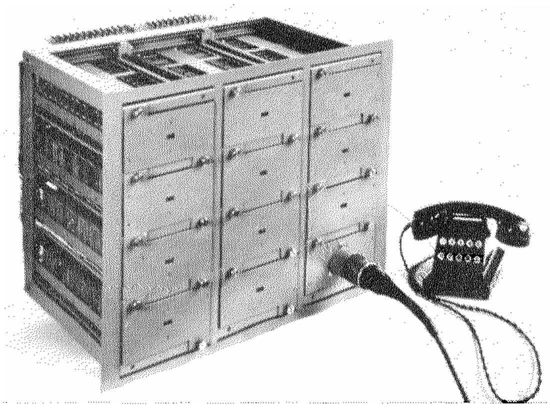
4. Switching Models

Theoretical and experimental work has led to the construction of test models of a concentrator and of a pulse-code-modulation exchange.

The model of the concentrator is illustrated in Figure 7. The equipment common to the subscribers; coder, decoder, memory, and control circuits, was constructed as for a real concentrator, and 28 subscribers' lines were equipped. A concentrator serving 150 subscribers' lines would have only the slightly greater volume of 0.15 cubic meter (5.6 cubic feet).

The model of the pulse-code multiplex exchange has 2 incoming pulse-code multiplex highways, one able to be connected to the concentrator, the second serving 3 registers and an operator's position, and 3 outgoing pulse-code multiplex highways, 2 of which are associated to form a local highway.

Figure 7—Experimental concentrator equipped for 28 subscribers' lines.



These highways are fully equipped with speech and switching memories. The design of the switching stage conforms to Figure 3.

The control circuit would be suitable for an exchange of several thousand subscribers' lines so far as its essential elements are concerned.

The control circuit for a pulse-code-modulation exchange is simpler than that for an exchange having a space-division speech circuit. In fact, all the information written in the switching memories of the speech circuit of a pulse-code exchange is accessible to the control circuit. The latter then requires only very few memories. Moreover, the transfer of information between speech and control circuits is accomplished with only a few interconnecting elements due to the multiplex operation of the speech circuit and to the digital nature of the signals that it routes.

The model required elementary logic circuits having relatively high-speed performance: response time < 0.1 microsecond. Fast transistors and diodes were used.

The storage points in the speech memories use a capacitor associated with 2 diodes, this capacitor being charged or not according to whether the information to be written is 1 or 0.

5. Prospects

The switching models that were built have made it possible to verify the validity of the solutions of the problems met in applying pulse-code modulation to telephone switching.

From the point of view of switching, the advantages are very important decreases in volume, weight, and power consumption of the equipment; and a very great flexibility in operation.

The use of the same type of modulation for both transmission and switching presents, among numerous advantages, a very easy exchange of supervisory information between exchanges that makes possible the locating of subscribers' line concentrators conveniently near the subscribers and remote from the exchange.

The very great progress made in recent years in both performance and cost of fast digital techniques, now makes it possible to consider pulse-code-modulation solutions that will be competitive with conventional designs. It is very likely that even the near future will show still further improvement in the situation.

6. References

1. P. Mornet, "Application of Pulse-Code Modulation to an Integrated Telephone Network: Part 1, Advantages of Pulse-Code Modulation," *Electrical Communication,* volume 38, number 1, pages 23–31; 1963.

2. A. Chatelon, "Application of Pulse-Code Modulation to an Integrated Telephone Network: Part 2, Transmission and Encoding," *Electrical Communication,* volume 38, number 1, pages 32–43; 1963.

3. S. Van Mierlo and H. H. Adelaar, "Système de télécommunications," French Patent 1 108 100; November 1953.

4. H. H. Adelaar and D. G. N. Hunter, "Use of 'Stantec Zebra' to Calculate a Traffic Table for

a Three-Link Time-Division-Multiplex Telephone Exchange," *Electrical Communication,* volume 36, number 3, pages 189–196; 1960.

5. L. R. F. Harris, "Time Sharing as Basis for Electronic Telephone Switching," *Proceedings of the Institution of Electrical Engineers,* volume 103, Part B, pages 722–742; March 1956.

6. E. Touraton and J. P. Le Corre, "Commutation entre Voies de Communication Multiplex," French Patent 1 212 984; October 1958.

7. J. G. Dupieux, J. P. Le Corre, and P. Seneque, "Etage de Commutation en Multiplex dans le Temps et ses Circuits de Commande Associés," French Patent Application 1 313 830; November 1961.

Jean Le Corre was born in Guiler-sur-Goyen, France, on 24 August 1927. He graduated as a radio-electrical engineer from the Ecole Supérieure d'Electricité of Paris in 1950.

In 1954, he joined Laboratoire Central de Télécommunications and was engaged in work on electronic telephone switching. He is now head of the electronic switching department.

Experimental Pulse-Code-Modulation Transmission for Local-Area Telephony

K. W. CATTERMOLE
D. R. BARBER
J. C. PRICE
Standard Telecommunication Laboratories; London, England

E. J. E. SMITH
Standard Telephones and Cables Limited; London, England

1. General Description

1.1 Purpose of Development

Pulse-code modulation, being extremely resistant to perturbations in the transmission path, is commonly postulated for use on obviously poor transmission media. Another application, potentially no less important, is to extend the transmission capacity of conventional paths such as multi-conductor cable. Paired cable intended for voice-frequency traffic is a poor medium for multiplex or other wide-band signals, on which it superimposes large and variable amounts of crosstalk and noise. These degradations are readily withstood by a binary signal, in which only the presence or absence of a pulse has to be detected, and which can in consequence be regenerated without accumulation of impairment.

The increase in capacity offered by this technique is most likely to be needed on junction routes between exchanges in large cities or conurbations,* which combine a substantial investment in multi-conductor cables with a large and growing traffic. Looking ahead a little further, it is also in these areas that the combined problems of transmission and switching may best be solved by digital techniques. Consequently, this equipment was designed for junction working; specifically, for interconnecting rotary exchanges in a designated experimental location, although there is nothing to preclude its use elsewhere.

1.2 Facilities

The system basically provides 24 channels, but, since one channel is allotted to synchronizing, 23 speech paths are provided, each with associated signalling facilities. Each speech waveform is sampled at 8 kilohertz, a high enough rate to permit transmission of the normal 0·3 to 3·4 kilohertz pass band. Each sample amplitude is quantized into one of 70 levels. Since this number is inadequate for transmission of speech signals having a wide range of mean volume, amplitude compression is used to extend the volume range by about 25 decibels, the ratio of clipping level to the smallest quantum interval being then about 56 decibels. Two similar paths, one in each direction, are combined by voice-frequency hybrid coils.

The coded speech requires 7 binary digits for each sample. An 8th digit is appended to carry signalling information. This could convey 2-condition signalling directly, but since the rotary exchanges require 3 or 4 conditions a further stage of time-division is used to break down this digit into 2 binary digits at 4 kilohertz each. The resulting signalling capacity, namely 2 high-speed channels in each direction associated with each speech channel, is adequate for any known type of switching plant.

1.3 Line Transmission

The total number of binary digits generated per second is the product of the sampling rate (8 kilohertz), number of digits per sample (8), and number of channels (24), namely, $1 \cdot 536 \times 10^6$. These are assembled sequentially for transmission on a single pair in each direction.

The attenuation of 0·64-millimetre diameter paired cable at 1·536 megahertz is about 25 decibels per kilometre; crosstalk margins vary considerably but in the worst cases either near end or far end can be less than 40 decibels on a kilometre length. Repeaters are therefore

* A great aggregate of urban communities, con + urban.

necessary. The mean spacing used is 1 kilometre with some scatter to permit the use of existing manholes. This interval, decided on early in the project, is unnecessarily short. The sending level is 2 volts peak to peak into 100 ohms, and a typical receiving level (after equalization) is about 100 millivolts. The repeaters will tolerate fluctuations of input level within the range ±6 decibels at the expense of some deterioration of noise rejection.

The repeaters are supplied with power over the signal pair. The whole chain is supplied in series from the transmitting terminal, each repeater dropping 9 volts with a current supply in the range of 25 to 40 milliamperes.

Power separation, as well as matching and balancing, requires the repeaters to have line transformers on input and output. These transformers do not pass direct-current or voice-frequency components of the code signal; the resulting distortion of long sequences of marks or spaces would make their regeneration uncertain unless some special form of signal were used. The code used in this equipment avoids long sequences, so making possible the regeneration of signals in which each mark is a unidirectional pulse.

1.4 Terminal Equipment

A block diagram of the transmitter is shown in Figure 1. Each speech input, after filtering and

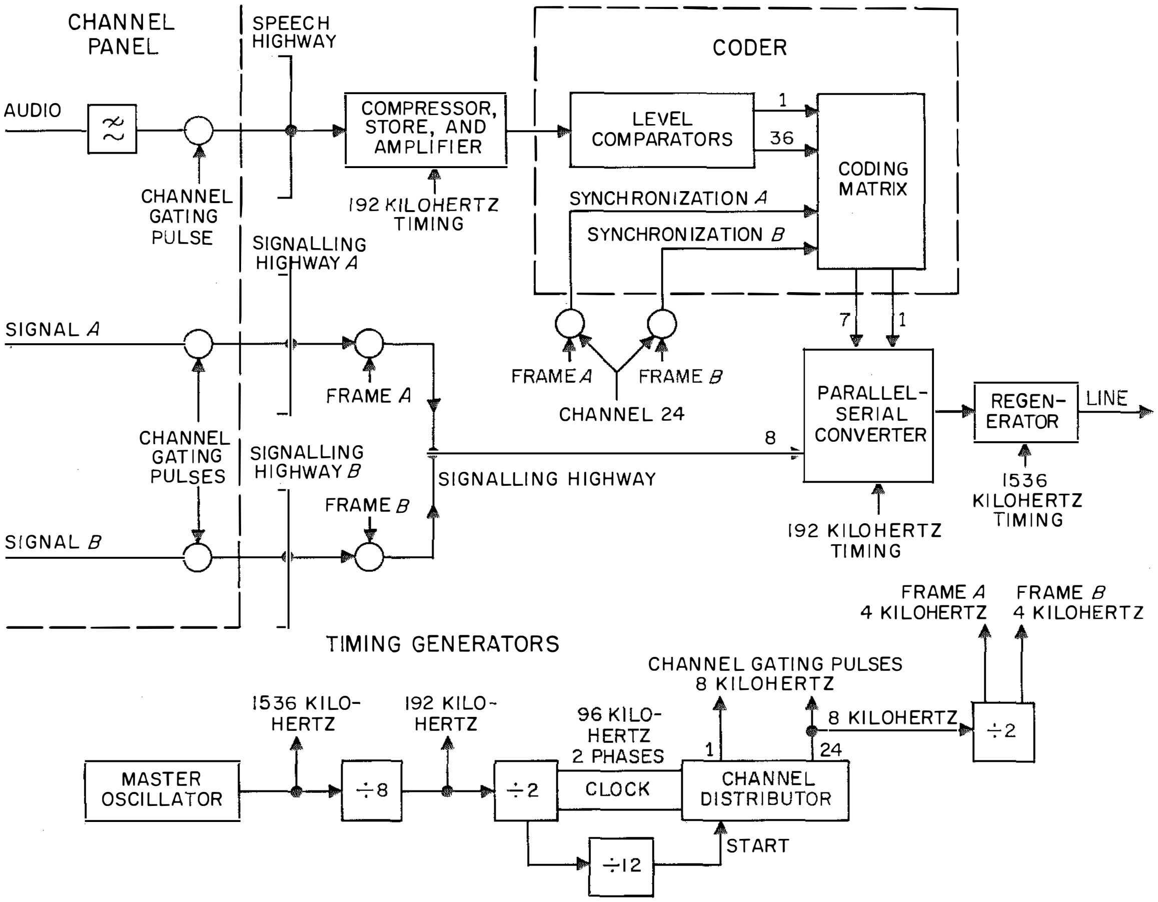


Figure 1—Transmitter.

amplitude limiting, is sampled for 2·6 microseconds in every 125 microseconds. The conversion efficiency of the storage-type modulator [11] is high and no channel amplifier is needed. The channels are combined on the speech highway, where the signal takes the form of multiplex amplitude-modulated pulses with a guard space between channels approximately equal to the sample width. The amplitude range of the pulses is reduced by the compressor. The compressor output is sampled again, and the sample amplitudes are stored as a charge on a capacitor to provide a constant-amplitude input to the coder throughout the coding operation.

The coder contains an array of amplitude comparators, biased to operate at different levels, to define the quantizing intervals. By an artifice explained in Section 3.2 each comparator serves to define 2 levels. The outputs of the comparator array define the sample level; they are connected to a matrix of diodes that energizes the appropriate digit outputs to generate a 7-digit code character. The digits are applied simultaneously to 7 of the 8 distinct inputs of a parallel-to-serial converter. They appear in sequence at the single output, and after regeneration are launched into the line.

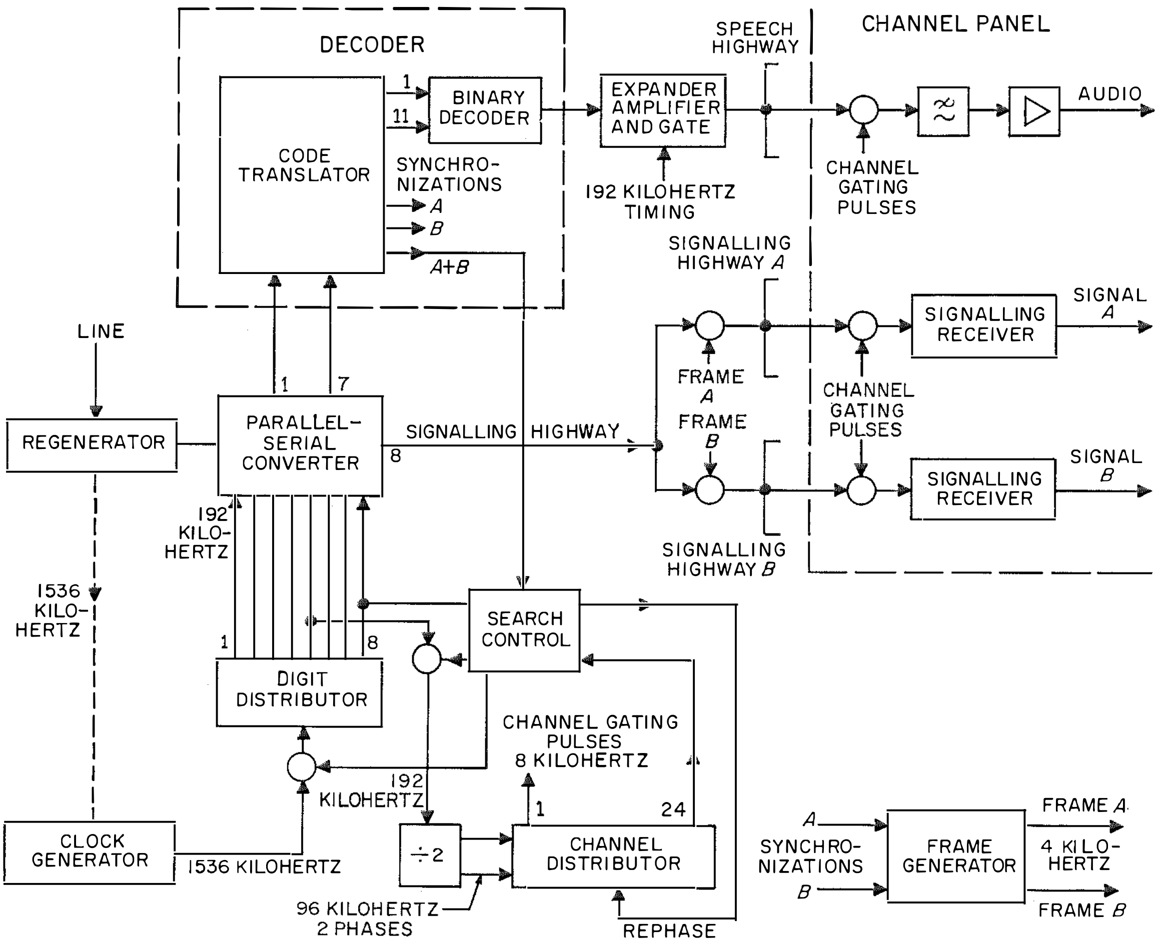


Figure 2—Receiver.

The number of signalling conditions necessitates the use of two binary signals, A and B, for each speech channel. Each signal input is sampled, and the signals from the various channels are assembled on signalling highways A and B. The latter are connected through alternately energized gates to the common signalling highway, which carries a complete frame (125 microseconds) of A signals followed by a complete frame of B signals, and so on. This combined signal is applied to the remaining input of the parallel-to-serial converter and appears as the 8th digit in the sequential output.

To identify the channels, so that they may be correctly separated at the receiver, distinctive code characters are transmitted in channel 24. To identify the frames so that signalling digits A and B may be separated, these synchronizing characters differ in alternate frames producing synchronizations A and B. Each character is generated by energizing an appropriate line in the diode matrix during the synchronizing channel period; the coder is inhibited at this time.

A block diagram of the receiver is shown in Figure 2. Much of this is obviously complementary to corresponding parts of the transmitter. Each incoming code character is arranged in parallel form, translated to a binary number by an array of saturable magnetic cores, and then converted by the decoder into an amplitude-modulated pulse. This, after amplitude expansion complementary to the initial compression, represents the quantized approximation to the input sample. The amplitude-modulated pulses are distributed to the individual channels; each channel audio signal is recovered by filtering, again with high conversion efficiency, and amplified to the required output level.

The signalling digit is separated and distributed via frame and channel gates to signalling receivers, each of which provides as output a make contact for actuation of the relay sets.

The process of searching for and maintaining synchronism with the transmitter requires some apparatus with no counterpart at the other terminal. The code translator array includes elements responsive to each synchronizing character. During normal running, a synchronizing character is received, and recognized, during the period allotted by the receiver channel distributor to channel 24. If the receiver is out of step with the transmitter, this recognition does not occur, and its absence initiates a process of searching. The counting rate of the digit distributor is changed, by inhibiting some of its clock pulses, with the effect that every possible group of 8 adjacent digits is offered in turn to the synchronizing detector. Eventually the synchronizing character, if present, is recognized; this stops the search, and starts the receive distributor running normally in synchronism with the incoming characters and frames.

2. Regeneration

2.1 Regenerable Signals

The process of regeneration has two aspects. One is the standardization of the individual pulses, which must be indistinguishable by amplitude, shape, or duration. The second aspect is the allocation of pulses to discrete positions on a regular time scale. This offers no especial difficulty at the transmitter, where a timing wave may be taken from the master oscillator. At the repeaters and the receiver, the only timing information available is that which can be recovered from the pulse-code-modulated signal. That is to say, we must extract a timing wave at the mean digit frequency and of constant phase in relation to the mean digit position despite the facts that some digits are mark and others space, more or less at random, and that any digit may be perturbed during its passage through the line.

If a signal is to be regenerable, its information content per unit time must not exceed the capacity of the medium. This is necessary but not, however, sufficient; since there are various practical hazards that may be overcome only by some form of restriction on the signals to be transmitted. We may, for instance, refrain from fully utilising the available bandwidth, signal-to-noise ratio, or repertoire of code sequences. Any of these procedures amounts to the provision of excess capacity as the price of some other benefit. The point may be clarified by discussion of two specific benefits desired.

As noted in Section 1.3, the exigencies of line transmission include the generation of some form of signal that will not suffer unduly from the inability of line transformers to pass direct current and very-low-frequency alternating-current components. Possible solutions may be classified according to the period of time over which the average (loosely the direct-current level) is reduced to negligibility. In order of increasing time period the possibilities are:

A. Period of 1 digit: use of dipole impulses, either amplitude-modulated (Figure 3B) or phase-modulated (Figure 3C).

B. Period of a few digits (from 2 up to character length): alternate mark inversion (Figure 3D).

C. Period of 1 character: use of a low-disparity code, in which limited numbers of marks and spaces occur (Figure 3E).

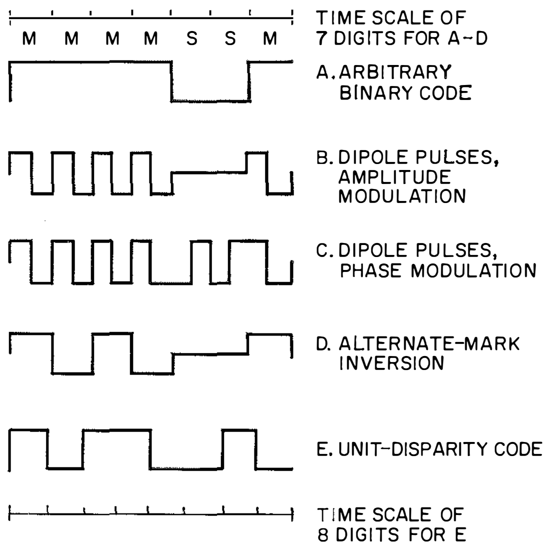


Figure 3—Alternative forms of coded signal.

D. Period of 1 cycle of modulation: ensure that any constant conditions that occur use characters of low disparity, reserving the high-disparity characters of the code for levels excited intermittently.

Method *D* alone is inadequate, since the time period is inconveniently great; although it is a useful adjunct to other methods. Method *A* tolerates the greatest low-frequency cut-off, but occupies by far the greatest bandwidth. Method *B*, requiring 3 distinguishable levels, is less resistant to crosstalk or noise. Of the practicable methods, *C* appears to be the most efficient in its use of informational capacity. It required more development, since no suitable codes or coding methods were known as previous pulse-code-modulation work invariably used binary-number or reflected binary codes. A variety of codes and coding methods has been devised [1-4] and the one used in this equipment will be described in Section 3.

If we define disparity as the excess of marks over spaces in any one character, then the disparity of a binary number of n digits (or any other code having the full repertoire of characters, 2^n in number) ranges from $+n$ (all marks) to $-n$ (all spaces); and the number of consecutive marks or spaces that can occur is indefinitely large. At the other extreme, it is possible for all the characters of the code to have the same disparity. A compromise seems admissible, and the one chosen is that the disparity of a speech character shall be ± 1. That is to say each character in a code of $2n + 1$ digits has either n marks and $n + 1$ spaces, or vice versa. The number of levels pertaining to

TABLE 1
NUMBER OF LEVELS FOR VARIOUS NUMBERS OF DIGITS

Number of Digits	Number of Levels	$\log_2$ (Number of Levels)
5	20	4·33
7	70	6·12
9	252	7·97
$2n + 1$	$2(2n + 1)!/n!(n + 1)!$	

various numbers of digits is shown in Table 1. The redundancy is equivalent to about 1 binary digit.

With speech conveyed by a 7-digit unit-disparity code, and signalling by an independent digit, each 8-digit character has 3, 4, or 5 marks. The longest run of marks or spaces that can occur in an arbitrary sequence of characters is 9.

Timing information may be derived only from transitions of the signal waveform. According to the form of the signal, these may depend on the presence of marks or of transitions between mark and space. (Phase-modulated dipole pulses are an exception.) In a code using the full range of disparity, the density of marks or transitions can fluctuate over a wide range: as an extreme, there may be none at all. Even if we dismiss complete absence as unlikely to be sustained, fluctuations of say 8 to 1 are undesirable for two reasons. Firstly, temporal fluctuations in the density of any single pulse train increase the difficulty of extracting a sufficiently regular timing wave. Secondly, differences of mean density among the various signals in a cable reduce the crosstalk margin proportionately: a high-density signal perturbs the timing of a low-density signal more than 2 equal signals perturb each other.

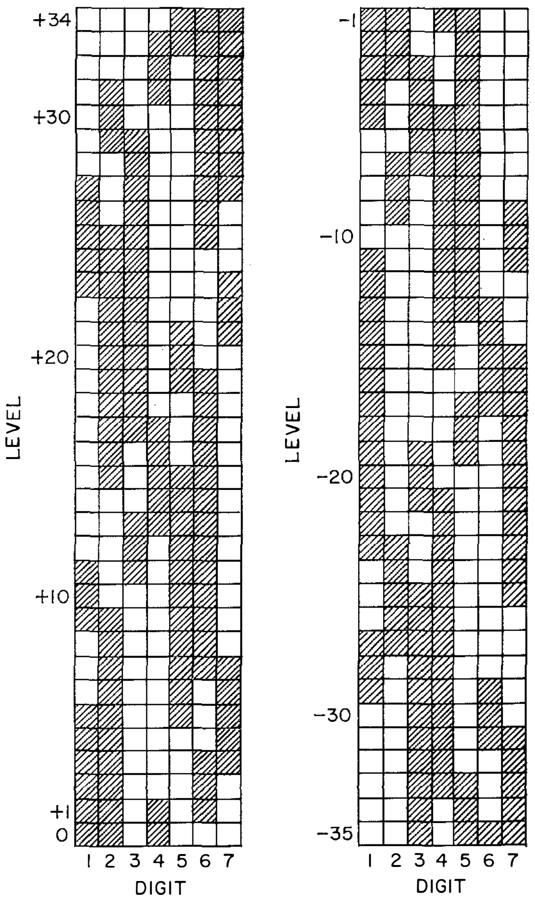


Figure 4—The code.

A low-disparity code, by definition, constrains the density of marks. The density of transitions is also constrained, though not so closely; moreover, it is easy to ensure that characters belonging to levels near 0 (which occur most frequently) avoid extreme transition densities. The code actually used (Figure 4) is a reasonable solution to each of the problems discussed.

To make the repeaters as simple as possible, pulses occupying only half the digit period were used in the first trial. This served its immediate purpose, but transmission of such pulses is wasteful of bandwidth. Later work, not described here, postulates a bandwidth approximately half that required in the original system.

2.2 Repeaters

Use has been made of 2 types of self-timing repeaters, whose block diagrams are shown in Figure 5. The essential difference is that the timing waveform is derived from the output or the input signal, respectively. Each has its own advantages, but on the whole input timing is preferred; it allows a wider tolerance on the mean frequency, and it is applicable to more-ideal regenerators with narrow sampling.

Several functions are, of course, common to all types of repeater. The line attenuation is equalized to within 1 decibel for a line section of average length, from low frequencies up to almost the digit rate, by an equalizer having 2 resonant sections. The signal level is raised by a 2-transistor preamplifier to about 2 volts peak, enough to operate a discriminator or gate.

The design of the timing filter is a compromise. To obtain a reasonably pure unmodulated sine wave from an input signal with virtually random modulation, a very narrow pass band is desired. However, the phase shift of the extracted sine wave must be confined within perhaps 10 degrees for any likely frequency drift of oscillator or filters; so frequency tolerances set a lower limit to the bandwidth. The practical solution was a quartz-crystal filter with a bandwidth of about 3·5 kilohertz. A 10-degree phase shift occurs at ±300 hertz, which is substantially wider than a practicable crystal tolerance.

It has been shown possible to use linear addition of the signal and timing wave [5, 6] but the use of a combined discriminator and timing gate has been preferred, even in an avowedly simple repeater, since the gate determines pulse width as well as time of occurrence, so permitting the use of a simple output stage; the gate isolates the controlled signal from the timing, on which it has little or no influence; and variations in timing-wave amplitude have less effect. A quite-satisfactory version used a saturated-transistor switch as a timing gate, but a second model using a current switch exhibited less intersymbol interference and will be described here.

In Figure 6A, transistors *T1, T2,* and *T3* are emitter-coupled, although with the low supply voltage used in the repeater it is not possible to define the emitter current precisely. The output is taken from the collector of *T3,* which can draw the full current only when *T1* and *T2* are cut off by positive excursions of their bases. The timing wave applied to *T1* ensures that it diverts current from the other transistors for half of each digit period. During the other half, current flows in *T2* or *T3* depending on whether the base of *T2,* driven by the amplified

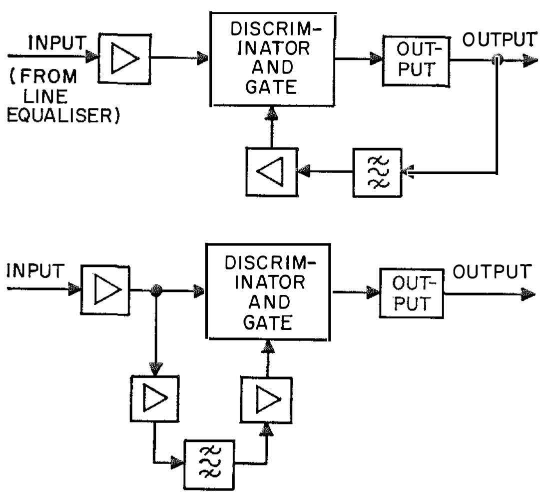


Figure 5—Repeater diagrams.

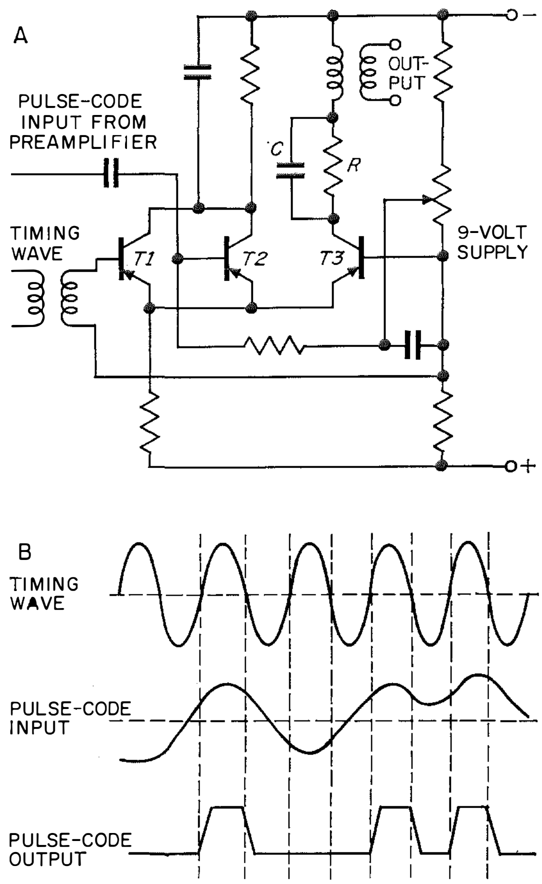


Figure 6—Combined discriminator and timing gate.

pulse-code-modulated signal, is negative or positive. The effect is that the circuit acts as a combined timing gate and amplitude discriminator, as illustrated by the waveforms of Figure 6B. Since a substantial output power is drawn from the collector of *T3,* this transistor is on the verge of saturation at pulse peaks; so better waveforms are obtained by building out the load to a constant resistance, using the capacitance-resistance network shown in series with the effective inductance and resistance of the transformer-coupled line.

It will be noted that, while the gate can restrict the duration of output pulses below that of the input, it cannot augment the width of narrow pulses. Consequently, it is dependent on a bandwidth restriction in the combination of line, equalizers, and preamplifier, which produces roughly the pulse shape shown in the figure.

All repeaters tested were capable of rejecting symmetrical interfering signals whose peak-to-peak amplitude was 6 decibels below signal under favourable conditions and 12 decibels below over a range of input levels and patterns.

The repeaters are supplied with power over the the signal pair. The consumption of each one, namely 20 milliamperes at 9 volts, is sufficiently low for several to be supplied in series over the lightest gauges of cable in service.

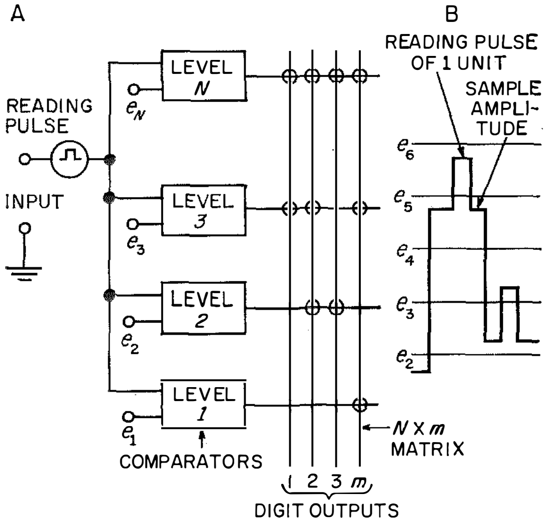


Figure 7—Coder for arbitrary code.

3. Coding and Decoding

3.1 Code

To provide an easily regenerable binary signal, as discussed in Section 2.1, a unit-disparity code was desired.

The various digits of a parallel coder are independently generated or regenerated, and large errors can occur if several digits are simultaneously in a state of indecision. It is, therefore, an advantage to employ a unit-distance code, in which characters belonging to adjacent levels differ in only 1 digit. Moreover, as will appear in Section 3.2, the structure of a parallel coder is simplified by use of a unit-distance unit-disparity code. The 7-digit code shown in Figure 4 has both these properties. It is also cyclic: the 2 extreme characters are suitable for juxta-position, so that the assignment of an origin is quite arbitrary. Also, any digit column in the raster is equivalent to any other when translated by some multiple of 10 places, so the assignment of digit positions is arbitrary. Ignoring obvious permutations there are 12 essentially different codes of this type. The choice of code and origin was based mainly on 2 criteria: (A) the characters belonging to levels near 0 which occur most frequently, should have an average number of transitions between mark and space and (B) these same characters should have the minimum probability of simulating suitable synchronizing characters.

3.2 Principle of Coder

We first refer to a perfectly general method for generating an arbitrary code, and then show how it may be simplified in the case of a unit-disparity unit-distance code. In Figure 7, consider an array of level comparators with a series of equally spaced bias levels e_1, e_2 . . . e_N. Each comparator gives an output pulse when the common input exceeds its threshold. For example, with the input waveform shown in Figure 7B, the first sample surpasses all the thresholds up to e_4, and so actuates several comparators when it is applied. To actuate only 1 comparator, a small pulse known as a reading pulse is superimposed on the input signal to be coded, giving an increment equal to the spacing between levels and therefore surpassing 1 more threshold (e_5 in the example). Thus an output from level r initiated by the reading pulse indicates that the input amplitude lies between e_{r-1} and e_r.

Any desired m-digit code may be obtained from a coding matrix with N inputs and m outputs and connections at the crosspoints so that each level energizes the digit outputs that are to be marks in the corresponding character of the code. The crosspoint connections are made by means of any elements capable of constituting an OR gate, for example, diodes, transistors, or windings on saturable magnetic cores. The comparators may employ semiconductor elements as in this equipment or saturable magnetic cores [1, 7].

This arrangement as it stands is rather lavish with comparators and crosspoints. However, the structure of a unit-distance unit-disparity code permits it to be simplified. Consider any 4-mark character of such a 7-digit code. Each adjacent character may differ from it in only 1 digit, and must contain 3 or 4 marks; it must therefore be a 3-mark character containing 3 out of the 4 marks specified. Also, the two 3-mark characters bracketing any 4-mark character must have different combinations of 3 out of the 4 marks. We may therefore deduce that 3- and 4-mark characters alternate in the code raster, and that each pair of 3-mark characters bracketing any 4-mark character contains between them all of its 4 marks.

These properties may be verified for the code used by reference to Figure 4. They lead to the coder [1] shown in Figure 8. This is similar to the previous arrangement, except that:

A. Comparators are provided only for odd levels, from 1 to $N+1$ (N is assumed even).

B. The reading pulse amplitude is 3 units (that is 1·5 times the spacing of the comparator threshold).

C. For a $(2n+1)$-digit code, there are n crosspoint connections in the matrix for each comparator. For example, 3 for a 7-digit code, corresponding to all the 3-mark characters.

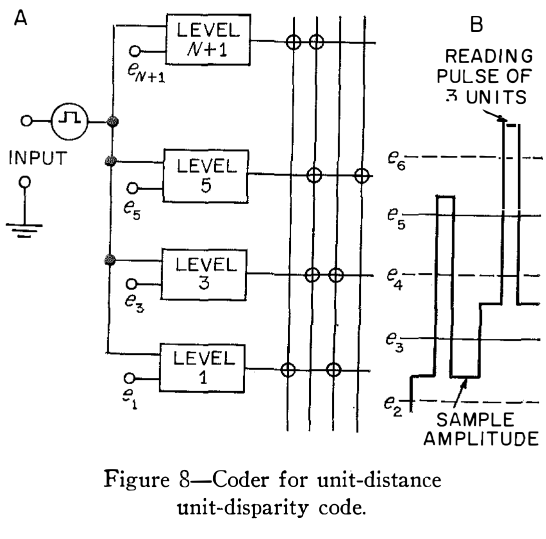


Figure 8—Coder for unit-distance unit-disparity code.

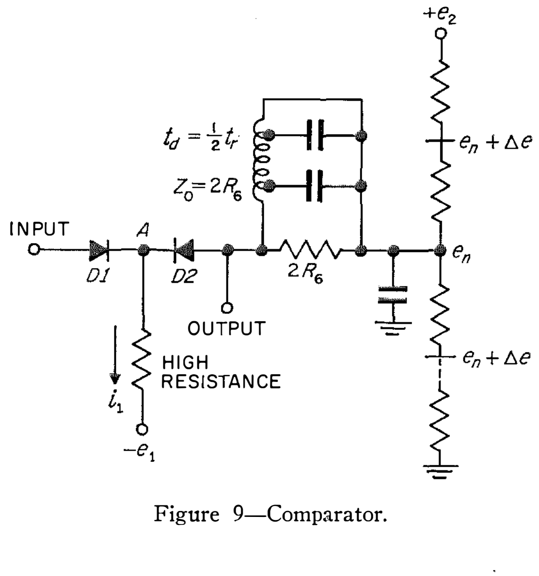


Figure 9—Comparator.

The result of these changes may be seen in Figure 8B. Here the full horizontal lines indicate the actual thresholds e_1, e_3, e_5, et cetera, and broken lines are equally spaced virtual thresholds e_2, e_4, et cetera. Any input lying above a real but below a virtual threshold (for example, the right-hand sample in the figure) causes the reading pulse to actuate 1 comparator and so generate an n-mark character. Any input lying above a virtual but below a real threshold as the left-hand sample in the figure, causes the reading pulse to actuate 2 comparators and so generate the $(n+1)$-mark character that is bracketed by the 2 n-mark characters pertaining to the comparators.

This specialization has not only reduced the number of comparators to just over half and the number of crosspoints to fewer than half; it also widens the tolerances on threshold levels and reading-pulse amplitude. Any likely variation in these magnitudes leads only to irregular spacing of quantum levels, whereas in the basic arrangement small errors can cause complete failure. In consequence, although the general parallel coder is rather unattractive the specific version for a unit-disparity unit-distance code is entirely practical.

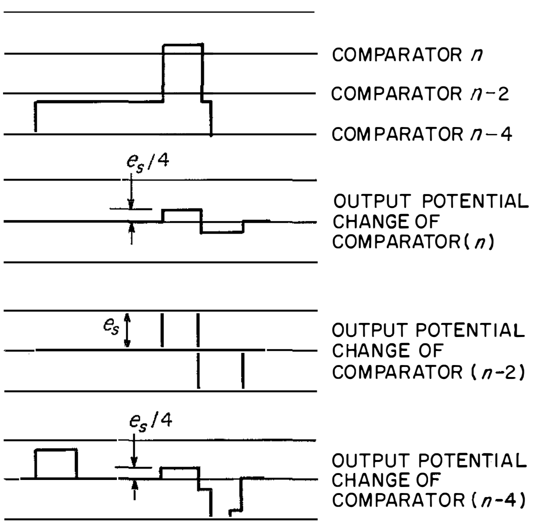


Figure 10—Waveforms of the comparator.

3.3 Practical Coder

A practical comparator is shown in Figure 9. The coder contains 36 such circuits, with their inputs connected in parallel and their reference potentials forming an arithmetic progression with a common difference Δe volts. If the input be sufficiently negative with respect to the reference potential e_n, diode D1 is cut off and a current i_1 flows through D2 and the short-circuited delay line. A sufficiently positive excursion cuts off D2 and diverts the current through D1 and the signal source, so generating a change of potential $e_s = i_1 R_6$ at the output terminal. The full output occurs only if the input excursion includes the whole interval from $(e_n - V_f)$ to $(e_n + V_f + e_s)$, where V_f is the potential drop across a diode passing a forward current i_1. Partial coverage of this comparator window diverts only part of the current and produces a proportional output. Reflection from the short-circuited line cancels any output after a time t_r.

A typical set of waveforms is shown in Figure 10. Only those outputs coincident with the reading pulse are significant; comparators below n-4, which would be energized only by the edges of the pulse-amplitude-modulated signal, are not shown.

Each comparator output drives 1 row of the matrix via an amplifier, which is represented by a current generator in Figure 11. The terminals a are connected to digit gates, which conduct only during the period of the reading pulse. Figure 12 shows the set of gated signals corresponding to the waveforms of Figure 10. A full output e_s from any comparator yields an amplitude e_d at this point; the total amplitude for any digit is proportional to the sum of the outputs of the comparators having a matrix element for that digit.

The fact that the comparators have a variable output as the input traverses the window, rather than a binary output dependent on a sharp threshold, would appear to invalidate the explanation given in Section 3.2. However, by making a binary decision after passage through the matrix it is possible to arrive at the same result. The decision threshold is set to $e_d/2$ volts (see Figure 12).

The effect of the subsequent binary decision is that only if it is driven through more than half its range can a comparator contribute to the output a mark peculiar to itself. Any input level causes either 1 or 2 comparators to be adequately driven by the reading pulse, so generating the 3- and 4-mark characters respectively. Critical levels occur at the centre of every interval defined by the comparator bias voltages. Thus the example shown yields the 3-mark character 1101000 corresponding to level n-2.

3.4 Principle of Decoder

Just as the coder generates digits in parallel, so the decoder requires parallel inputs. The first process towards decoding is therefore to separate the incoming serial digits. The digit distributor generates 192-kilohertz pulse trains in 8 phases, each corresponding to a digit. For each of the 7 speech digits a gate, operated by a pulse train of appropriate phase, separates that digit from the incoming pulse-code-modulated signal and admits it to a magnetic storage element. The 8th digit period, which in the transmission path is used for signalling, is occupied in the speech path by the decoding process. Early in this period, therefore, the stored digits are read out simultaneously and pass after amplification to the parallel decoder.

The decoder, like the coder, is most easily explained (as, indeed, it was devised) by starting with a general method of decoding or translating and then seeing how it can be simplified when we take into account the structural properties of the unit-disparity unit-distance code. Consider an array of detectors, all supplied with the pulse-code-modulated signal, but each responsive to only 1 character from the code (Figure 13). Binary output from these detectors may be used to drive the rows of a ma-

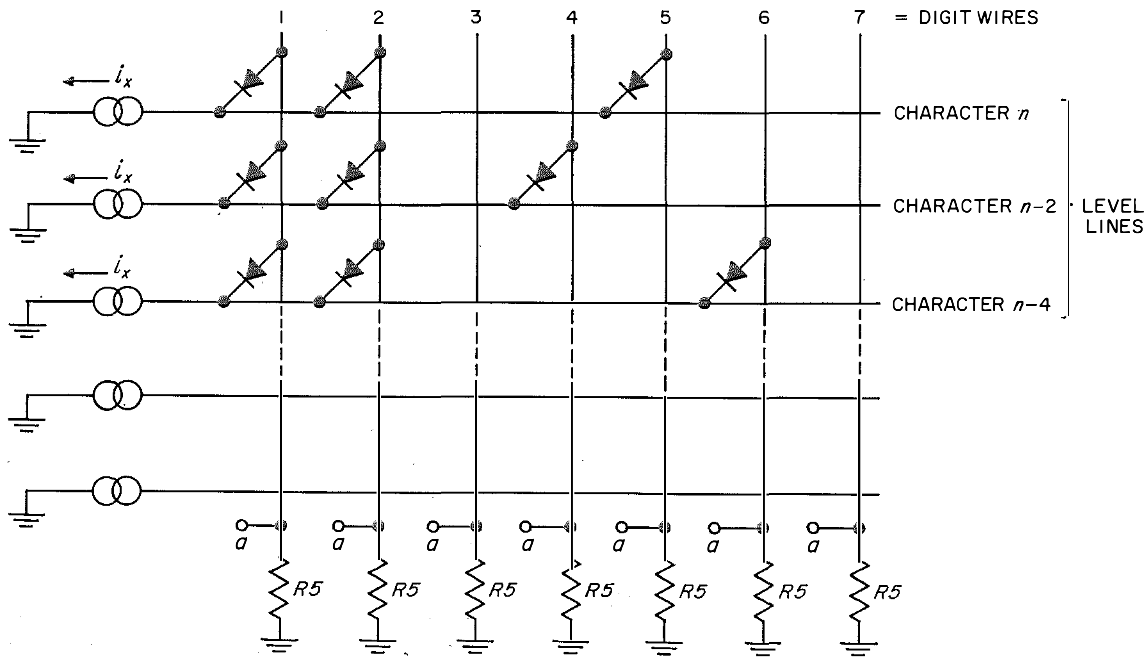


Figure 11—Coding matrix.

trix, similar in principle to a coding matrix (Figure 7), whose crosspoints may be connected to digit columns so as to yield any desired output code, in particular a binary number that is easily decoded.

The simplification possible in a decoder or translator for unit-disparity unit-distance code depends on 2 properties: adjacent characters differ in only 1 digit, and characters with positive and negative disparity alternate so that in a 7-digit code there are both 4- and 3-mark characters respectively.

These properties lead to the decoder or translator [8] shown in Figure 14. There are half as many detectors as in Figure 13, each responsive to either of 2 adjacent characters. If we can make a detector responsive to one character, we can equally well make a detector responsive to a pair of characters by omitting any discrimination of the digit in which they differ and operating solely on the remaining digits, in this case 6, which they have in common. Now the binary-number translation of such adjacent pairs also differs in only 1 digit: the least significant. Consequently, all but 1 of the digits of the translation can be obtained from a reduced matrix of $N/2$ rows and m-1 columns, for a code with N levels and m digits. The least significant digit, the only one differing in the translation, may be found by checking the property in which the characters of a pair differ, their disparity. Positive or negative disparity in the input corresponds to a final 1 or 0 in the output. This requires an auxiliary detector, of about the same degree of complexity as the others.

The construction of the detectors has not so far been specified. Given parallel inputs, the array of detectors can be built as a matrix rather like the output matrix. In this case, specialization to the unit-distance unit-disparity code halves the number of rows in the matrixes and reduces the number of crosspoints in each row by $1 \cdot 5$ on the average.

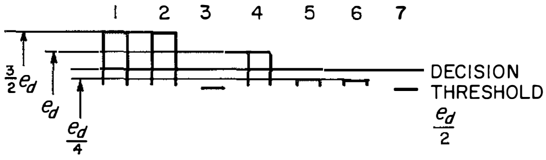


Figure 12—Coding matrix output.

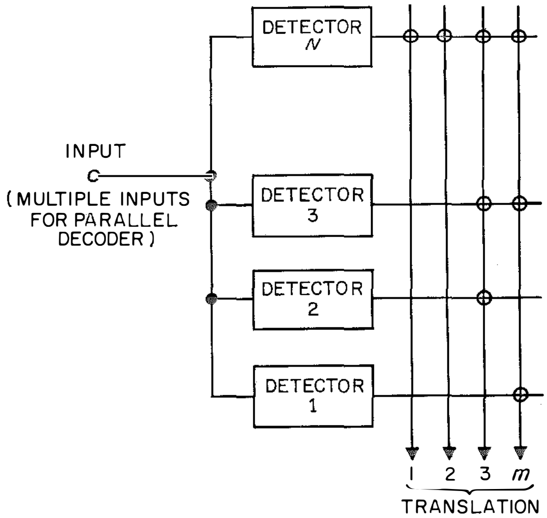


Figure 13—Code translator for arbitrary code.

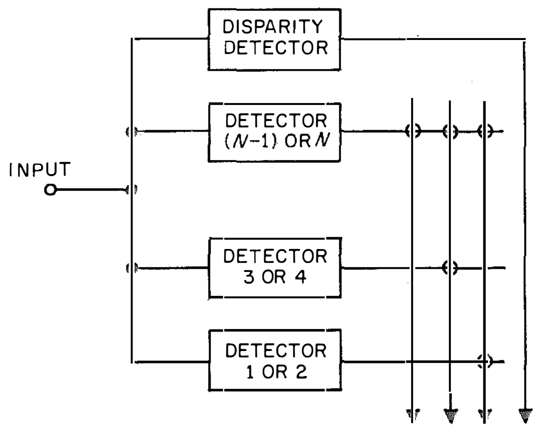


Figure 14—Code translator for unit-distance unit-disparity code.

3.5 Practical Decoder

The code translator employs an array of saturable magnetic cores, each of which is equivalent to a detector plus a line of the matrix on the scheme of Figure 14. The ideal core for this function would have no coercivity or hysteresis: its B–H characteristic would have a sharp threshold, so that an arbitrarily small magnetomotive force in either direction produced saturation flux in that direction. For simplicity we start by explaining the action in terms of ideal elements.

The array of cores comprising the code translator, shown in Figure 15, is traversed by windings carrying digit currents derived from the serial-parallel converter. Standard unit currents flow in the digit windings corresponding to marks of the input code character: no current flows in those corresponding to space. Each core is wound so as to be responsive to a different pair of characters. The method is best explained by taking a specific example. Each core is polarised by current through a bias winding in a sense that we will call reverse.

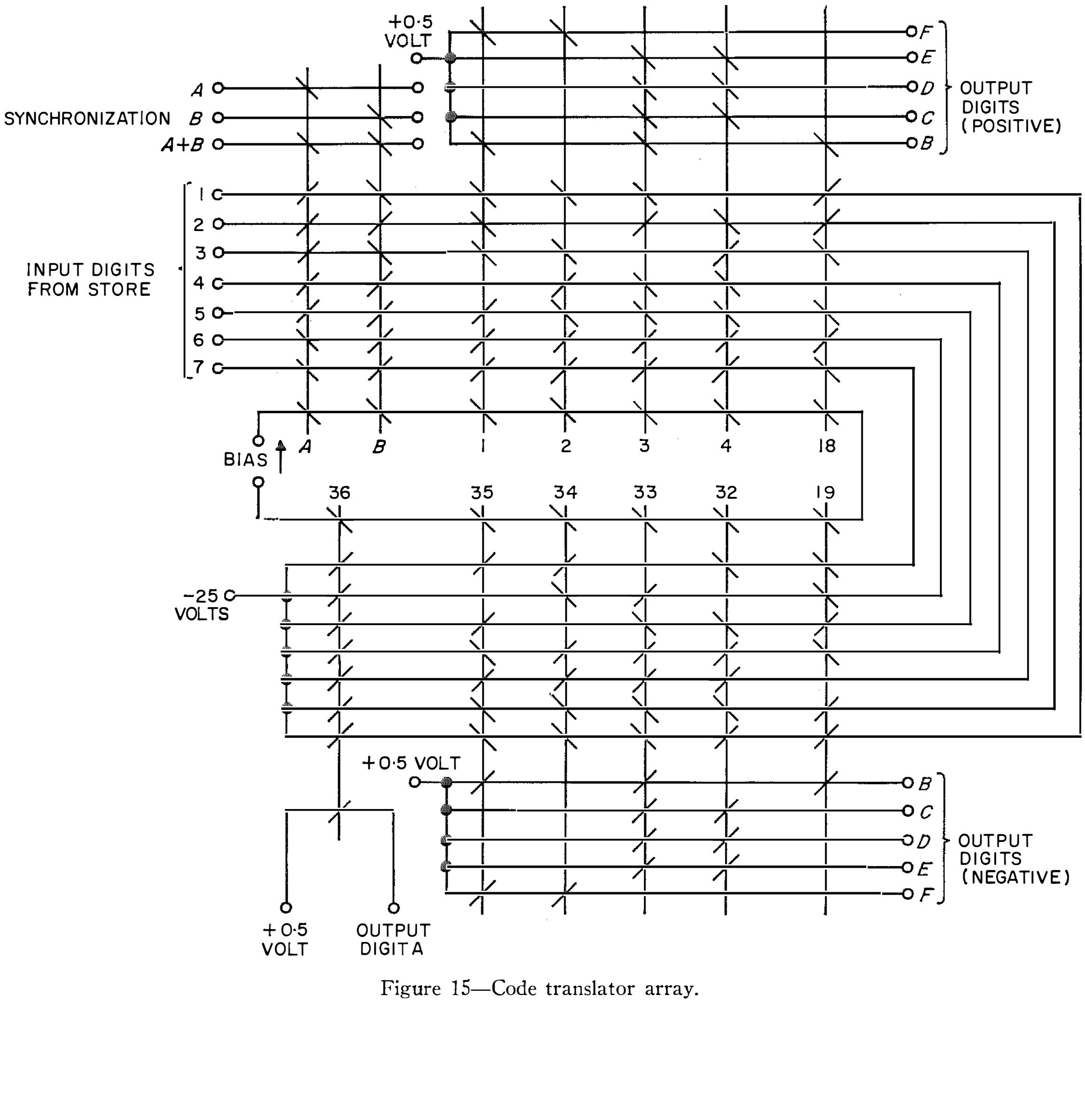


Figure 15—Code translator array.

This bias can be overcome only by unit currents in at least 3 digit windings. Core 1 has windings in the forward sense for digits 5, 6, and 7, and so is energized only if all these digits are marks. It also has windings in the reverse sense for digits 1, 2, and 3: so if a mark occurs in any of these positions it reduces the total magnetomotive force below the threshold and there is no significant change of flux. Digit 4 is not applied to this core and so may be mark or space indifferently. This core therefore responds to either of 2 characters: 0001111 or 0000111, corresponding to levels +33 and +34 respectively. Similarly, any other core responds to a pair of adjacent characters, the higher level being even numbered.

The next step is to generate the digits of a binary number corresponding to the even-numbered level. The cores are divided into 2 groups, dealing with positive and negative levels respectively. Each group has 5 output lines *B, C, D, E, F* allotted to digits with weights 2, 4, 8, 16, 32 respectively; since we are dealing with even numbers, weight 1 is not needed at this stage. The level chosen as example, namely 34, is the sum of 2 and 32; so core 1 requires to be connected to outputs *B* and *F*.

These outputs are pulsed in the negative sense whenever the core, having been switched from reverse to forward magnetization by the input digits, is reset by the bias. The negative pulses trigger digit regenerators that apply standard output pulses through appropriate weighting resistors to a common output terminal.

It remains to distinguish the pairs of adjacent levels. This is done by the supplementary core 36, which is biased and wound so that it responds to any group of 4 marks but ignores a smaller number. The output of this core, digit *A*, is given the weight −1 unit, so that, for example, level 33 is generated as the sum of +34 from the action of core 1 and −1 from the supplementary core.

Two further cores *A* and *B*, driven in the same way as the level cores, respond to the synchronizing characters 11111000 and 01011110 respectively. Individual outputs trigger the frame generator (see Figure 2) while the common output supplies the main synchronizing pulse to the search control unit.

3.6 Compression and Expansion

The number of quanta provided for each polarity, namely 35, is only of the same order as the volume range occurring in the telephone network. That is to say, if quantizing were linear and a loud talker's speech were moderately clipped, a soft talker would just excite 1 quantum of each polarity. A coder with a greater, but still practicable, number of levels would allow him to excite 3 or 4. This is so obviously inadequate that non-linear quantizing is accepted as essential. It is conveniently provided by inserting a common amplitude compressor before the coder and a complementary expander after the decoder.

The ideal compression characteristic would provide constant speech quality over a large part of the required volume range, which implies that it should be roughly logarithmic. Departure from this law, and degradation of quality, are inevitable at the lowest levels where any practical characteristic approaches linearity. The characteristic used is mainly determined by the non-linear devices available, but is near enough to the ideal to be useful; it is shown in Figure 16, with a logarithmic curve (broken line) for comparison. The ratio of the highest to the lowest threshold is raised by the compandor from 31 to about 56 decibels.

The amount of compression that may usefully be employed is limited by the departure of the quiescent level from 0. The gating pedestals and other inaccuracies in the equipment are such that a wider range of levels would not be worthwhile. Two effects of such displacements should be distinguished. Firstly, suppose a displacement occurring at the transmitter to be conveyed through the system and reproduced at the receiver. The effect is that small signals are quantized more coarsely than they should be; the effect on quantizing noise can be computed [9]. Secondly, suppose that a displacement present at the compressor is not conveyed to the coder and so is not reproduced at the expander. The effect is that compression and expansion are no longer complementary, being centred on different points; so the speech is distorted asymmetrically. Direct coupling of the compressor, expandor, and common amplifiers was not feasible. However, the direct-current component of a complete frame may be practically annulled by reversing the sense of the gating pedestal in alternate channels.

One of the most stringent requirements imposed by the passage of multiplexed pulse-amplitude-modulated signals through common equipment is the provision of adequate low-frequency transmission to ensure satisfactory crosstalk margins. Circuits preceding the compressor need a cut-off frequency of 2 to 3 hertz, whereas in subsequent circuits compandor advantage permits the use of cut-off frequencies ranging up to 8 hertz.

The compandor is of the instantaneous type using germanium diodes as the non-linear element. The basic circuit of the compressor is shown in Figure 17. For a high degree of compression, $R1 >> R2$. The peak signal current in the diodes is chosen to be 5 milliamperes, and they then have a forward resistance that is nominally 75 ohms. To ensure an adequate match of compressor-expander characteristic over the operating temperature range, the diodes are placed in an oven. Since the equipment is designed to work at a maximum temperature of 45 degrees centigrade (40 degrees ambient + 5 degrees bay rise) the oven is set to regulate at 50 ± 1 degrees centigrade.

The output of the decoder is connected directly to the expander-amplifier. The expander is given a characteristic that is complementary to that of the compressor by using a compressor network in a feedback path. A simplified circuit is given in Figure 18, where $R4$ includes the output impedance of the amplifier viewed at the emitter, together with series resistance if necessary.

The problem of low frequency cut-off is simplified by the fact that the decoder output is a dipole pulse. Since the 2 halves, although of equal area, are not exactly the same in shape, they are not equally treated by the expander. The crosstalk advantage is, therefore, not more than about 10 decibels; nevertheless the total crosstalk of the expander-amplifier is very low.

The dipole must, of course, be converted to a single pulse for demodulation. This is done by gating out 1 half with a diode bridge.

3.7 Signalling

There is a great variety of switching plant, and a corresponding variety of signals. Although 2 signalling conditions are often enough, 3 or 4 are sometimes needed. Although a slow signalling channel is often adequate, a relatively fast one is sometimes needed.

Thus a transmission system for universal application must have a signalling channel of very much greater capacity than is needed in the less exigent situations; and in a carrier system, this is not easy to provide. In a pulse-code-modulation system, this difficulty does not arise. The addition of 1 binary digit at the usual speech sampling rate provides adequate capacity whilst occupying only a small fraction of the total bandwidth. The reason for the easy compatibility of signalling and pulse-code-modulation is that the signalling information is essentially digital, and so is well matched to the signalling channel provided by an auxiliary binary digit.

Consideration of signalling problems was forced on the designers of this system by the fact that the first installation was to be on a junction route between rotary exchanges. This type of exchange requires up to 4 signalling conditions, and the total delay of a loop path must be small enough to permit revertive impulsing. The signalling conditions as present in the rotary exchange are encoded into 2 binary digits designated A and B, by means of a relay set at each terminal. Since the relay set takes up part of the limited time available, the permissible delay in the pulse-code-modulation equipment is only 2 milliseconds. The 4-kilohertz A and B digits, obtained by time division of digit 8 as shown in the block diagrams, can achieve this with a reasonable margin and are adequate for any known signalling system.

4. Synchronizing

It is necessary to keep the sampling, decoding, and distributing functions of the receiver in step with the incoming signal with reference to 4 distinct time scales, thus:

A. The digit period. Incoming digits must be sampled by the receive regenerator at, or near, the centre of each digit period.

B. The character period. The 8 digits of each character must be identified so that the first 7 may be decoded and the 8th routed to the signalling highway.

C. The frame period. The 24 channels of each frame must be identified and distributed.

D. The signalling-frame period. Alternate frames, although indifferent to speech samples, must be identified so that A and B signals may be distributed.

The 1536-kilohertz timing wave that defines the centre of the digit period is extracted from the incoming signal by filtering, as described in section 2. The phase relation of the timing and pulse-code-modulation waveforms issuing from the receive regenerator is determinate, so time scale A may be taken for granted in this context.

For the purpose of synchronizing, time scales B and C are amalgamated. Distinctive characters transmitted in channel 24 are recognized by 2 cores in the code translator (Figure 15). Their occurrence during the period allotted by the receive distributor to channel 24 confirms the receiver in its normal or LOCK condition. Thus both character and channel synchronism are necessary to maintain LOCK; failure of either initiates a SEARCH process.

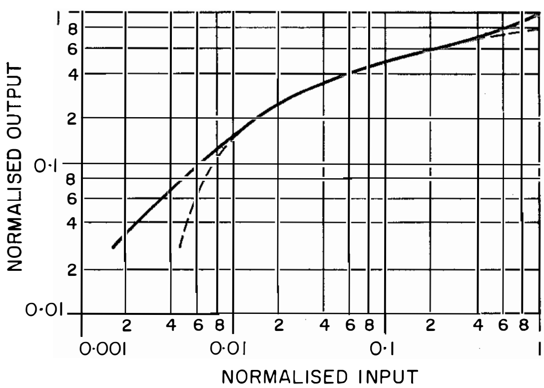


Figure 16—Compressor characteristic.

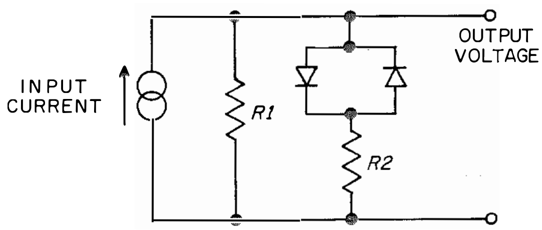


Figure 17—Compressor.

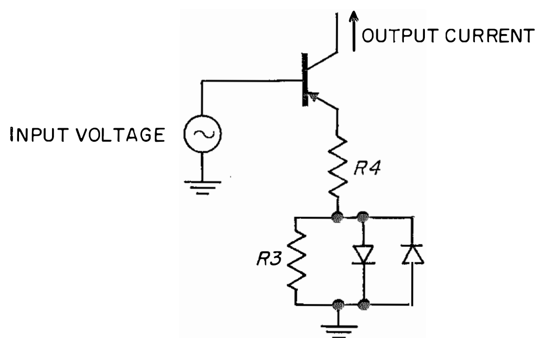


Figure 18—Expander-amplifier.

By means of a gate shown in Figure 2, the search control unit admits a group of 8 clock pulses to the digit distributor and then blanks a group of 3. This change of count enables the incoming pulse train to be scanned for synchronizing characters; every possible time position is examined during a period of 1·375 milliseconds equal to 11 frames. Meanwhile, the channel distributor is stopped. Recognition of a synchronizing character resets the control to LOCK; the distributor clock supplies are restored and the channel distributor is started at the correct time by a rephase signal.

Noise in the transmission path is just as likely to impair the synchronizing channel as any other. Searching, if it were to be initiated in response to a single imperfect synchronizing character, would be more commonly spurious than due to genuine loss of synchronism. Moreover, interruption of synchronism would disturb every channel more radically than would any error in a speech sample; so this spurious searching would make a major contribution to the system noise. For this reason, the receiver is dislodged from a sustained LOCK condition only after a succession of 3 absent synchronizing characters, although during the course of a search it responds to their individual presence or absence as rapidly as possible.

Should a search be halted fortuitously by a single simulation of a synchronizing character, it is taken up again at a corresponding point in the next frame. These properties are obtained by the use of a differential counter that registers successes and failures in identifying synchronizing characters.

These provisions [10] yield an adequately high speed of search while reducing to insignificance the hazards of falsely simulated synchronizing characters or obliterated true ones. Synchronism is restored after any momentary loss in less than 1·5 milliseconds. This enables calls in progress to be retained despite the interruption.

The 4-kilohertz framing generator that determines time scale D is directly triggered by the A and B synchronizing pulses educed respectively by the 2 synchronizing characters transmitted in alternate frames. This means that it has no protection against obliteration of synchronizing characters by noise, though there is some guard against simulation. Single errors are, however, of little importance to the signalling, because the operate or release time of even the high-speed relays is much greater than the sampling interval.

The choice of synchronizing characters is a compromise between conflicting factors. It is not possible to choose truly distinctive characters without wasting much of the informational capacity of the normal coded signals, so some risk of simulation, for example, by the end of 1 channel character plus the beginning of another, must be accepted. The probability of simulation may be reduced by some departure from the low-disparity property of the normal characters, but this obviously cannot be taken to an extreme without risk of faulty regeneration. Also, the 2 characters must not be easily confused. The final choice was 11111000 and 01011110; it will be noted that these fall within the normal disparity range but adopt an unusually close grouping of marks.

5. Conclusions

The field experiment was conducted over a junction route in Madrid on 18 lines provided by the Compañia Telefonica Nacional de España. The route joined the Norte and Delicias exchanges, on the north and south sides of the city respectively; it passed through Gran Via, the largest and busiest exchange in the city, but was not terminated there. Over the whole length of 6·2 kilometres, pairs of 0·64-millimetre diameter were available in each of 2 cables, laid in the same duct for about 4·4 kilometres but diverging for the remainder.

The pairs were part of either 900- or 600-pair cable, the remainder of which was carrying normal traffic. Repeater sites with a nominal spacing of 1 kilometre were chosen from the existing manholes; the actual section lengths ranged from 968 to 1134 metres, with an average of 1040 metres.

Repeaters developed and manufactured by Standard Telecommunication Laboratories and Standard Electrica (Madrid) were installed by the latter on 6 pairs, 3 in each direction. The 2 directions of transmission were in separate cables. The 6 pairs chosen included those that on preliminary measurements had shown the worst crosstalk.

Terminals developed and manufactured by Standard Telecommunication Laboratories and Standard Telephones and Cables were installed at the Norte and Delicias exchanges. Each installation comprised 2 terminals, 1 fully equipped and the other having only 3 sets of channel panels. Relay sets designed and manufactured by Le Matériel Téléphonique were interposed between the rotary exchanges and the signalling circuits of the pulse-code-modulation equipment. Rotary switching plant, comprising a line finder and first selector, was installed at Norte so that calls could be originated in isolation from the main exchange. There was no special switching plant at Delicias; the incoming pulse-code-modulation junctions were connected to the main exchange in the usual way. The studies made included:

A. Measurements of transmission loss, crosstalk, and noise on the cables.

B. Checks of various parts of the equipment in isolation.

C. Objective measurements on the pulse-code-modulation system in isolation.

D. Recording of conversational speech, random sentences, and articulation material passed through the pulse-code-modulation system in isolation, for subjective evaluation.

E. Operational check with pulse-code-modulation junctions linking the exchanges.

This trial has shown the feasibility of pulse-code-modulation operation on local area cables. The more important properties observed are listed below.

A. Speech quality is good at optimum volume, tolerable over a range of perhaps 20 to 25 decibels, and intelligible over a much-wider range, perhaps 40 to 45 decibels. At optimum volume, it remains tolerable after transmission through 3 or 4 systems in tandem (simulated by recording and retransmission) and intelligible up to 9 or 10.

B. In a transmission system for universal application, moderate extension of volume range would be needed.

C. Signalling capabilities are adequate for the needs of rotary exchanges, which are the most exigent type.

D. Digit rates of 1·5 megahertz are feasible on installed paper-insulated multi-pair cables; the 1-kilometre repeater spacing could be increased.

E. Some form of pulse-code-modulated signal resistant to low-frequency cut-off is essential, and the unit-disparity code is a practicable means of achieving this.

F. Rapid recovery of synchronism enables calls to be held through momentary interruptions.

G. This equipment was used as a vehicle to try out a number of techniques, of which the more useful could form a basis for further development.

6. References

1. H. Grayson, K. W. Cattermole, and W. Neu, British Patent 849 891.

2. W. Neu, "Some Techniques of Pulse-Code Modulation," *Bulletin des Schweizerischen Elektrotechnischen Vereins,* volume 51, number 20, pages 8–17; October 1960.

3. K. W. Cattermole, "Binary Codes of Low Disparity for Pulse-Code Modulation." To be published.

4. K. W. Cattermole and D. R. Barber, British Patent 849 892.

5. E. D. Sunde, "Self-Timing Regenerative Repeaters," *Bell System Technical Journal,* volume 36, pages 891–937; July 1957.

6. L. R. Wrathall, "Transistorized Binary Pulse Regenerator," *Bell System Technical Journal,* volume 35, pages 1059–1114; September 1956.

7. A. T. Starr, K. W. Cattermole, and J. C. Price, British Patent 806 397.

8. E. Lambourn, British Patent 873 944.

9. B. Smith, "Instantaneous Companding of Quantized Signals," *Bell System Technical Journal,* volume 36, pages 653–709; May 1957.

10. K. W. Cattermole and J. C. Price, British Patent 863 669.

11. K. W. Cattermole and J. C. Price, "Efficiency and Reciprocity in Pulse Amplitude Modulation," Parts 1 and 2, *Proceedings of the Institution of Electrical Engineers,* Part B, volume 105, pages 449–470; September 1958.

K. W. Cattermole was born in Forest Gate, Essex, England, in 1923. He received the B.Sc. degree with first-class honours from London University in 1952.

From 1942 to 1947, he served with the Royal Electrical and Mechanical Engineers. After a year with the British Broadcasting Corporation as a transmitter engineer he joined Standard Telecommunication Laboratories in 1948. He has worked successively on the development of microwave radio links, time-division-switching, and pulse-code-modulation systems. He is currently in charge of the pulse-code-modulation laboratory in the line transmission division.

Mr. Cattermole is the author of a book on "Transistor Circuits," published in 1959.

D. R. Barber was born in Drumheller, Alberta, Canada, in 1927. He received the B.Sc. (Eng.) degree from the University of London in 1947. During the university vacations, he worked at Standard Telephones and Cables from 1944 to 1947.

From 1947 to 1950 he served with the Royal Electrical and Mechanical Engineers and subsequently returned to Canada. He joined Canadian General Electric where he was engaged in the development of microwave links.

In 1955, he joined Standard Telecommunication Laboratories, England, where he has been engaged in the development of repeater amplifiers and pulse-code-modulation inter-exchange telephony systems.

Mr. Barber is an Associate Member of the Institution of Electrical Engineers.

J. C. Price was born in Highbury, London, England, on 3 June 1921. He studied at the Sir John Cass College and in 1950 received the B.Sc. degree from the University of London.

He served in the Royal Air Force during the second world war, working on radar. From 1950 to 1960, he was a part-time science lecturer at Southgate Evening Institute. He did research work on fuses for Belling Lee from 1950 to 1953.

He joined Standard Telecommunication Laboratories in 1953 and has since worked on transistor pulse and logic circuits and on pulse-code-modulation systems.

E. J. E. Smith was born in Poona, India, on 10 February 1930. He became a graduate member of the Institution of Electrical Engineers in 1956.

From 1948 to 1953, he served with the Royal Electrical and Mechanical Engineers. He then joined Standard Telephones and Cables and has been engaged in the development of pilot regulating equipment for coaxial and open-wire systems, rural area systems, and more recently in the design of local area systems. Mr. Smith is now in charge of the local area and open-wire systems development section.

SEL-Taschenbuch (Reference Data Book for Communication Engineers)

A new reference data book, edited by H. Sarkowski, has been published in German by Standard Elektrik Lorenz. It presents brief descriptions, tables, and diagrams under the following chapter headings.

1. General Data for Communication Engineers
2. Components and Parts Used in Telecommunication
3. Electroacoustics
4. Teleprinting
5. Information Processing
6. Telephone Switching
7. Cables and Open-Wire Lines
8. Transmission
9. Radio
10. Power-Supply Equipment
11. Mathematical Section

The book is 11 by 15 centimeters (4.4 by 6 inches) and contains 312 pages. It is available from Verlag Berliner Union, Stuttgart, Urbanstrasse 12–14, at DM 11.40 per copy.

United Kingdom–Faroes–Iceland (Scotice) Submarine-Cable Telephone System

M. V. YOUNG
W. J. ARCHIBALD
Standard Telephones and Cables Limited; London, England

1. Introduction

The requirements of the International Civil Aviation Organisation for improved communication facilities among its stations in the United Kingdom, Iceland, Greenland, and Newfoundland added to the growing commercial demand for improved telephone and telegraph facilities between the United Kingdom, Faroe Islands, and Iceland resulted in the Scotice [1] project. Figure 1 illustrates how the loop formed by Cantat, Scotice, and Icecan [2] with London and Cornerbrook as the eastern and western terminals respectively may be switched rapidly to provide high-quality communication between all stations of the International Civil Aviation Organization even in the presence of a fault in one of the submarine-cable links.

Scotice provides telephone and telegraph service over a band of 72 kilocycles per second in each direction. The telegraph load may occupy up to ⅓ of this band. The system uses a single 0·46-inch (11·7-millimetre) diameter polythene coaxial cable. The South section of 290 nautical miles (537 kilometres) contains 10 submerged repeaters and links Gairloch in Scotland with Torshavn in Faroe Islands. The North section of 408 miles (756 kilometres) contains 15 submerged repeaters and 1 submerged equaliser between Torshavn and Vestmannaeyjar in Iceland. The 24 traffic channels, each 3 kilocycles per second wide, are transmitted over the bands from 24 to 96 and from 120 to 192 kilocycles per second in the A–B and B–A directions respectively. In addition, 3 engineering circuits of 2-kilocycle-per-second width each are similarly provided over the bands from 18 to 24 and from 192 to 198 kilocycles per second. Supervisory facilities of the frequency-doubler type monitor the loop transmission over the submerged repeaters and occupy the bands from 96 to 100 and from 192 to 200 kilocycles per second.

The North section. with terminal A at Torshavn and terminal B at Vestmannaeyjar, was laid in November 1961 by *H.M.T.S. Alert* of the British Post Office and the South section, with terminal A at Torshavn and B at Gairloch, was laid during the following month by *C.S. John W. Mackay* of the Commercial Cable Company.

2. System Design

2.1 Line Equipment

The North and South sections must meet the requirements of the Comité Consultatif International Télégraphique et Téléphonique under such adverse temperature conditions as are encountered when the cable is installed with allowance for the addition of 2 nautical miles (3·7 kilometres) of repair cable and with ⅓ of the telephone channels loaded with 24-channel voice-frequency telegraph signals.

The mean temperature of the sea bottom is 4·2 degrees centigrade with extremes of 2·2 to 6·2 degrees for the North section. For the South section, these values are 7 and from 4·5 to 9·5 degrees centigrade respectively.

2.1.1 *Submarine Cable*

The number of telephone channels to be accommodated being relatively small, a correspondingly thin coaxial cable is used. The centre conductor consists of a single copper wire of 0·128-inch (3·3-millimetre) diameter. The diameter over the polythene insulation is 0·46 inch (11·7 millimetres) and the return conductor consists of 6 copper tapes with 1 binder tape. This construction gives a characteristic impedance of 52 ohms and an attenuation at 200 kilocycles per second of 2·36 decibels per nautical mile (0·147 neper per kilometre) at 15·5 degrees centigrade. At the mean temperature of 4·2 degrees centigrade, the attenuation is 2·3 decibels per nautical mile (0·143 neper per kilometre) at 196 kilocycles per second. The type of armour used depends on the depth. In the deep water of both sections, type *D* cable is used, while in the South section each armour wire is separately taped. Heavier armour is used in shallower water and screened cable is installed for shore ends and for land cable. The temperature coefficient of attenuation is +0·17 per cent per degree centigrade, the pressure coefficient is +0·05 per cent per 100 fathoms, and the laying factor is −0·4 per cent.

2.1.2 *Submerged Repeaters*

The requirements for the repeaters are derived from the line characteristics and for the following conditions.

A. Changes in cable attenuation due to seasonal temperature variations at the sea bottom.

B. Assumed increased length of cable of 2 nautical miles (3·7 kilometres) due to future repairs.

C. Equalisation errors.

D. Overload point and basic noise provided by the valves employed.

An amplifier gain of 65 decibels at 196 kilocycles per second can be justified. Since the band-width of the system is relatively narrow, 40 decibels of over-all feedback can be applied to the amplifier and the intermodulation requirement as calculated by the method of Brockbank and Wass [6] is achieved. The nominal repeater levels are given in Table 1.

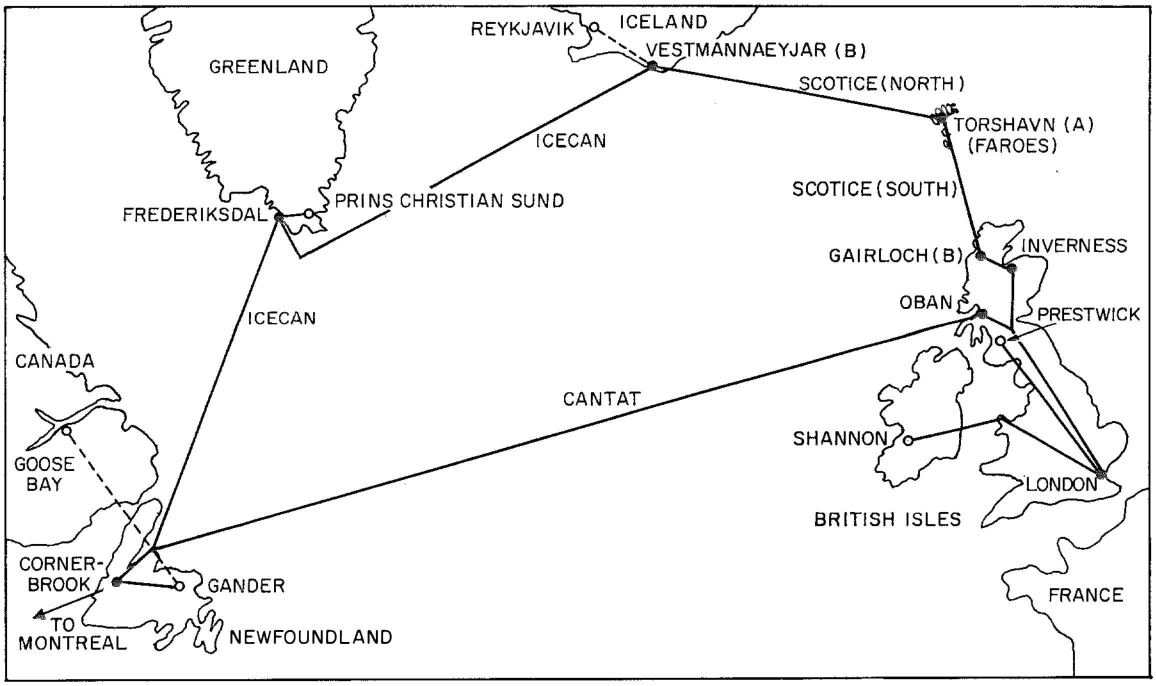


Figure 1—The Cantat, Scotice, and Icecan loop linking the air-traffic-control stations of the International Civil Aviation Organization, which are shown by circles. The broken line is a radio link from Gander to Goose Bay.

The schematic diagram of the submerged repeaters shown in Figure 2 illustrates that a common amplifier is used for both directions of transmission. The stability margin requirements of 40 decibels under normal conditions, 20 decibels under cable fault conditions in the transmission bands, and 20 decibels outside the transmission bands are achieved by the use of directional filters, frequency-selective bridge networks, and power separating filters.

The repeater amplifier providing a rising gain-frequency characteristic consists of a pair of 3-stage amplifiers connected in parallel between common input and output networks. Approximately 40 decibels of over-all voltage and current feedback is applied to both amplifiers

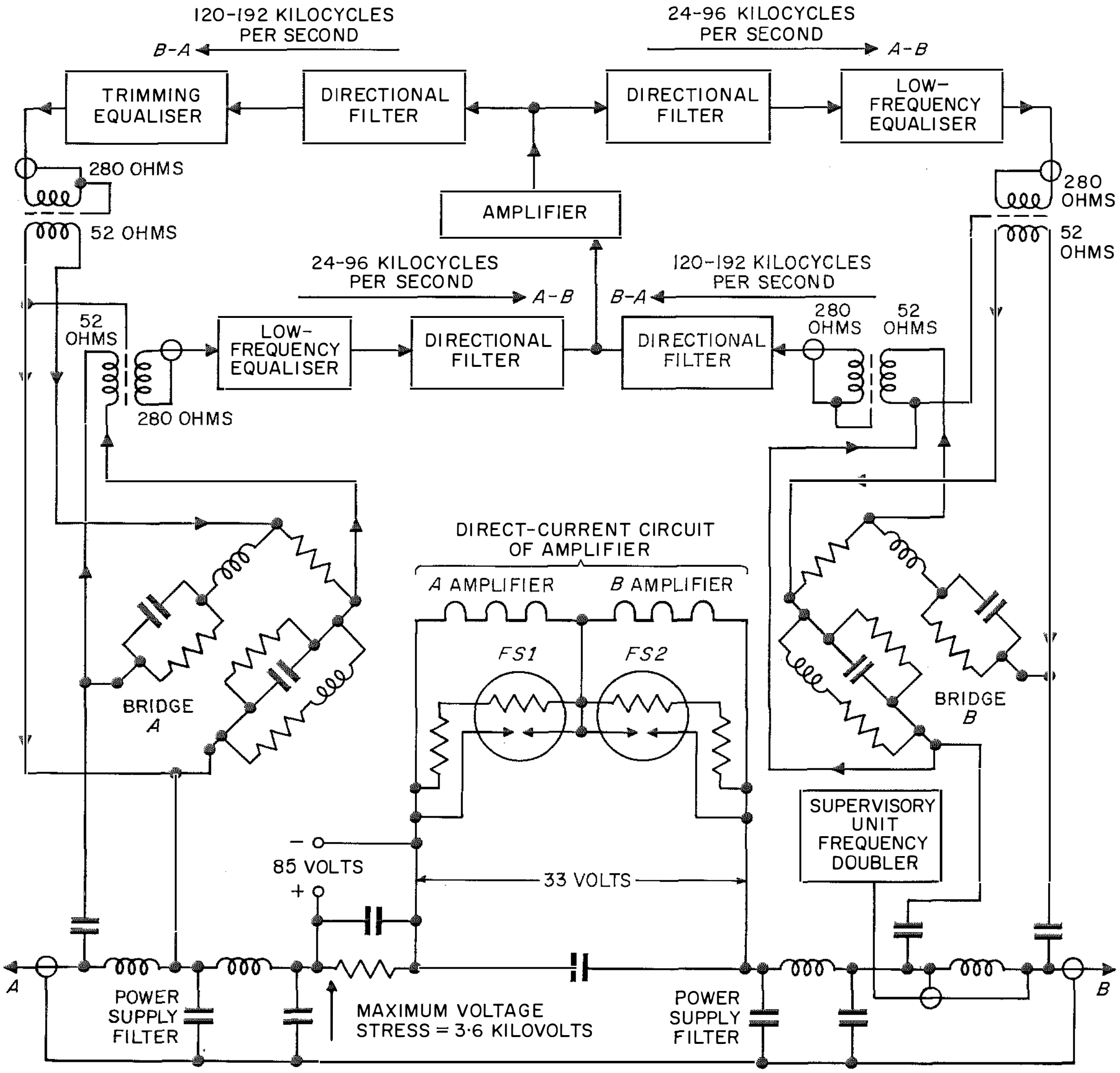


Figure 2—Diagram of submerged repeater. A direct current of 316 milliamperes flowing in the center conductor of the cable provides power to operate the amplifiers.

via a common bridged-T beta network. By this means, a fault within either amplifier changes the over-all gain by less than 0·1 decibel. The valves are special submerged-repeater types similar to the British Post Office [4] type *6P12*. The *5A/162* is used for the first 2 stages and *5A/166* for the final stage. The gain–frequency characteristic is controlled principally by the input transformer, output transformer, and beta network. Accordingly, these components are manufactured to extra-close tolerances and are hermetically sealed in dry nitrogen to insure stability.

In the event of a valve-heater failure, thermal short-circuiting fuse *FS1* or *FS2* operates to make a soldered joint across the fuse contacts, thus maintaining the power supply to the remaining amplifier.

The supervisory unit [7] contains an input crystal band-pass filter, a frequency doubler, and an output band-pass filter. Connection is made to the repeater via high-value protective resistors such that even in the event of a complete failure within the unit, the bridging loss is less than 0·15 decibel at all frequencies. In the North and South sections, each repeater has a different supervisory crystal-filtered mid-band frequency between 96 and 100 kilocycles per second. Thus, by selecting the appropriate frequency, loop transmission measurements can be made from the *A* terminal (Torshavn) to the *B* end of each repeater where the doubled frequency in the band from 192 to 200 kilocycles per second returns to the *A* terminal.

TABLE 1
NOMINAL REPEATER RELATIVE LEVELS

Frequency in Kilocycles per second	Input in Decibels	Output in Decibels
24	−39·2	−16·9
50	−43·7	−12·6
100	−51·4	− 7·6
120	−58·0	− 9·9
150	−59·6	− 5·8
196	−62·9	− 1·4

The electrical units are mounted in copper cans that, like the paper capacitor cans and connecting wire, are gold-plated to give an inert finish. The cans and the amplifier chassis are mounted on Perspex bars to withstand the maximum voltage stress of 3·6 kilovolts, then, after a period of dry-nitrogen flushing, the electric unit is sealed in a brass cylinder 7·75 inches (19·7 centimetres) in diameter. The internal unit is contained in a demountable pressure housing. The seal between the high-tensile-steel housing and the bulkheads is provided by a special arrangement of pre-pressurised *O* rings. The actual cable connections that pass through the bulkheads are tested at a pressure of 5 tons per square inch (787 kilogrammes per square centimetre) for a month before they are actually used. External to the bulkheads are the jointing chambers and cable-armour-clamping arrangements. The demountable housing is also used for the submerged equaliser and is described in more detail elsewhere [5].

2.1.3 *Submerged Equalisers*

The submerged equaliser used in the North section is described in detail elsewhere [5]. Briefly, the internal unit consists of a chassis containing power-separating and directional filters in the sealed unit. Space is provided on the chassis for mounting equaliser sections and cable simulators. The design of the equaliser may take place while the cable is being laid so as to permit compensation for changes in cable attenuation due to unpredictable laying effects. Kits of hermetically sealed capacitor units, resistor units, inductor units, and cable-simulator units are used to build up the circuit configurations required. After assembly of the necessary components on the chassis, this is sealed into its demountable housing and can be laid in the same way as a repeater. The designing, building, and testing of an equaliser, including adjusting the length of the cable and making a sea-cable joint, may take up to 16 hours. Figure 3 shows the equalisation error over the North section of Scotice.

2.2 Terminal Equipment

2.2.1 *Channelling Equipment*

High-efficiency 3-kilocycle-per-second channelling equipment is used to assemble 16 channels into a full group (60–108 kilocycles per second) and a further 8 channels into a half group (60–84 kilocycles per second).

2.2.2 *Main Transmission Path*

The full group is treated as group 5 and the half group as part of group 4 of a standard coaxial-line system. The subsequent modulation and demodulation processes are shown in the simplified block schematic diagrams of Figure 4, which shows one of the two *A* terminals at Torshavn, Faroe Islands, and in Figure 5, which shows the *B* terminal at Gairloch, Scotland, or Vestmannaeyjar, Iceland. The frequency allocation is shown in Figure 6.

2.2.3 *Speaker Equipment*

Both local and omnibus telephone and telegraph engineering channels are required. The band available in each direction is 6 kilocycles per second, some of which is, however, already permanently occupied by an edge pilot and some of which must also be available for repeater supervisory purposes. Three channels of 2-kilocycles per second each are provided, 2 of which are used for speech circuits and a portion of the third channel is available for at least 2 voice-frequency telegraph circuits. These 3 channels are translated in 2 stages to the 54-60-kilocycle-per-second band and combined at the group distribution frame with group 5 in the *A–B* direction. They are translated to the 84-90-kilocycle-per-second band and combined with group 4 in the *B–A* direction.

2.2.4 *Edge Pilots*

An edge pilot is transmitted at 23 kilocycles per second in the *A–B* direction and at 193 kilocycles per second in the *B–A* direction. Each pilot is monitored and recorded at the receiving end. A further edge pilot of 120 kilocycles per second can, however, be obtained in the *B–A* direction by injecting 60 kilocycles per second into group 5 at the *B* terminal.

2.2.5 *Carrier Supply*

This is largely conventional and based on a crystal-controlled 124-kilocycle-per-second master oscillator divided down to 4 kilocycles per second. The special frequency of 576 kilocycles per second is picked off as a harmonic of 4 kilocycles per second, and 672 is derived by modulation of 576 and 96 kilocycles per second. Since failure of either of these special-frequency supplies or of either of the two group carriers of 564 and 612 kilocycles per second would result in loss of all traffic or a considerable part of it, all carrier supply amplifiers are duplicated and change-over from standby to working is automatic in the event of failure.

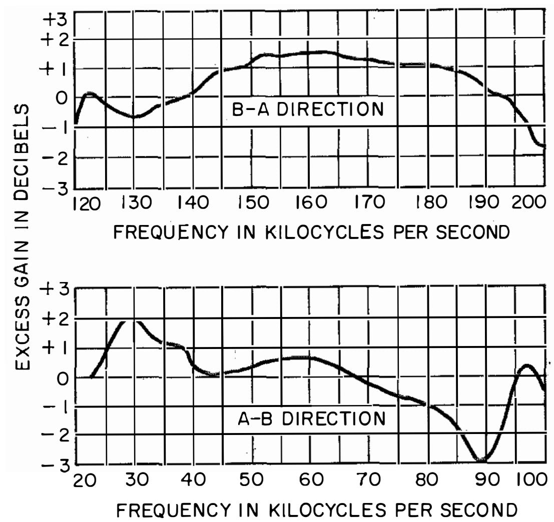


Figure 3—Equalisation error for the North section of Scotice from cable head to cable head. There are 15 repeaters and 1 equaliser in this 408-nautical mile (576-kilometre) run of cable.

2.2.6 *Frequency-Comparison Pilots*

It is desirable that the British standard of 60 kilocycles per second be accurately transmitted from Gairloch to Torshavn and Vestmannaeyjar and then be available (still as a frequency standard) for onward transmission to Reykjavik and over Icecan towards Greenland. To avoid the errors that modulation by incorrectly synchronised carrier supplies could introduce, the 60-kilocycle-per-second standard is doubled at Gairloch and transmitted to Torshavn as 120 kilocycles per second. At Torshavn, it is divided down to 60 kilocycles per second for local frequency-comparison purposes, then further divided down and transmitted as 20 kilocycles per second along to

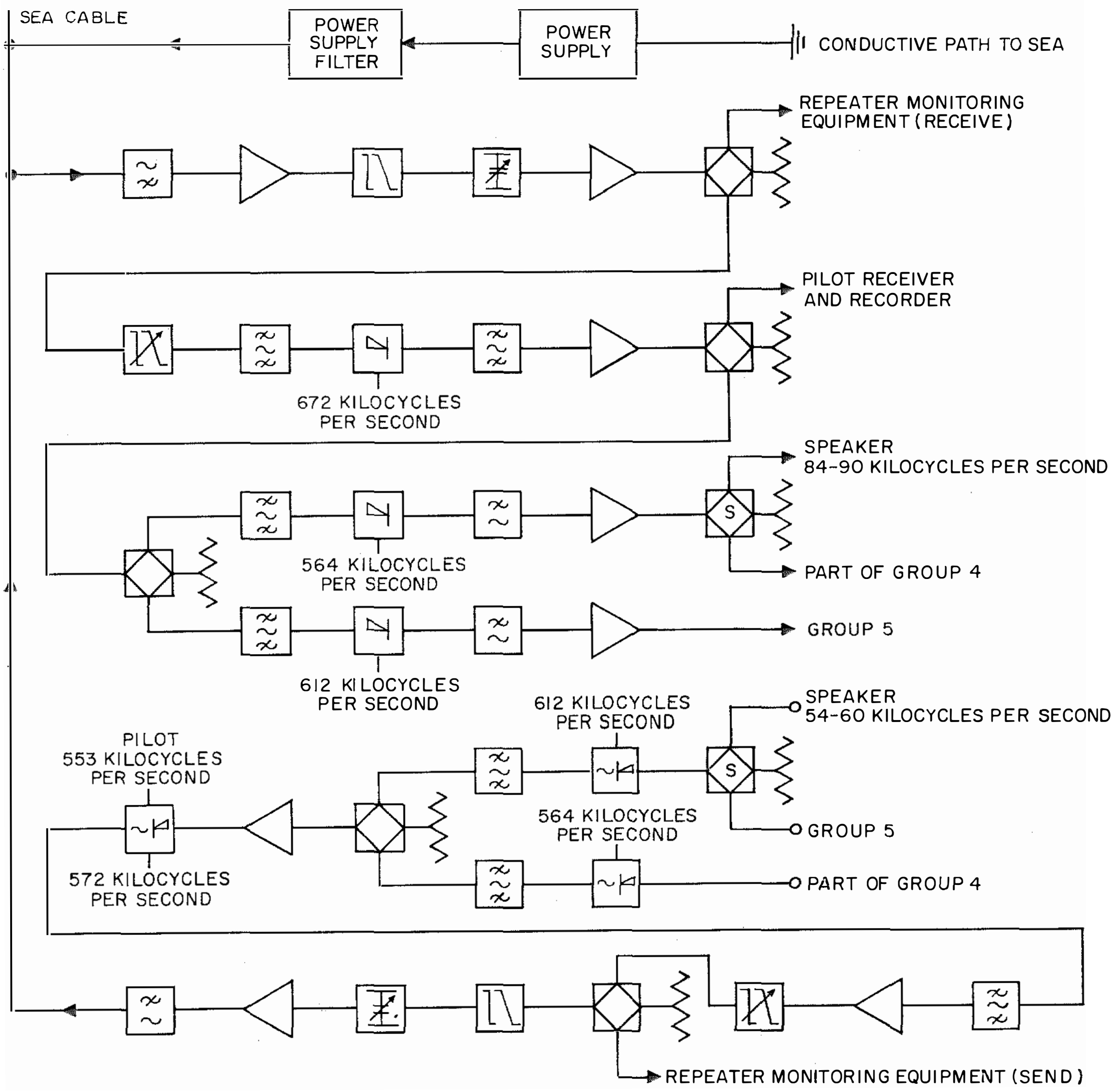


Figure 4—Simplified diagram of main transmission path at *A* terminal.

Vestmannaeyjar, where it is multiplied back to 60 kilocycles per second.

2.2.7 *Through Circuits*

Channelling equipment for all 24 channels is provided at Gairloch and Vestmannaeyjar, but at Torshavn only the half group is translated to audio frequencies, the full group being a "through" group to Gairloch and Vestmannaeyjar.

2.2.8 *Power Supply*

The power equipment is required to supply a constant direct current of 316 milliamperes that varies by not more than ±1 per cent. It is designed for double-end feeding and each terminal normally supplies half of the total system voltage. Either terminal can automatically supply the whole voltage in the event of failure at the other terminal. The equipment is connected in series with the centre conductor of the cable and return is via the sea. The total system voltages of the North and South sections are 2440 and 1670 volts respectively. Working and standby power-feeding units are provided at both Gairloch and Vestmannaeyjar. At Torshavn, in addition to the working units for each of the two terminals, there is a standby unit that can be switched into service in either the North or South section.

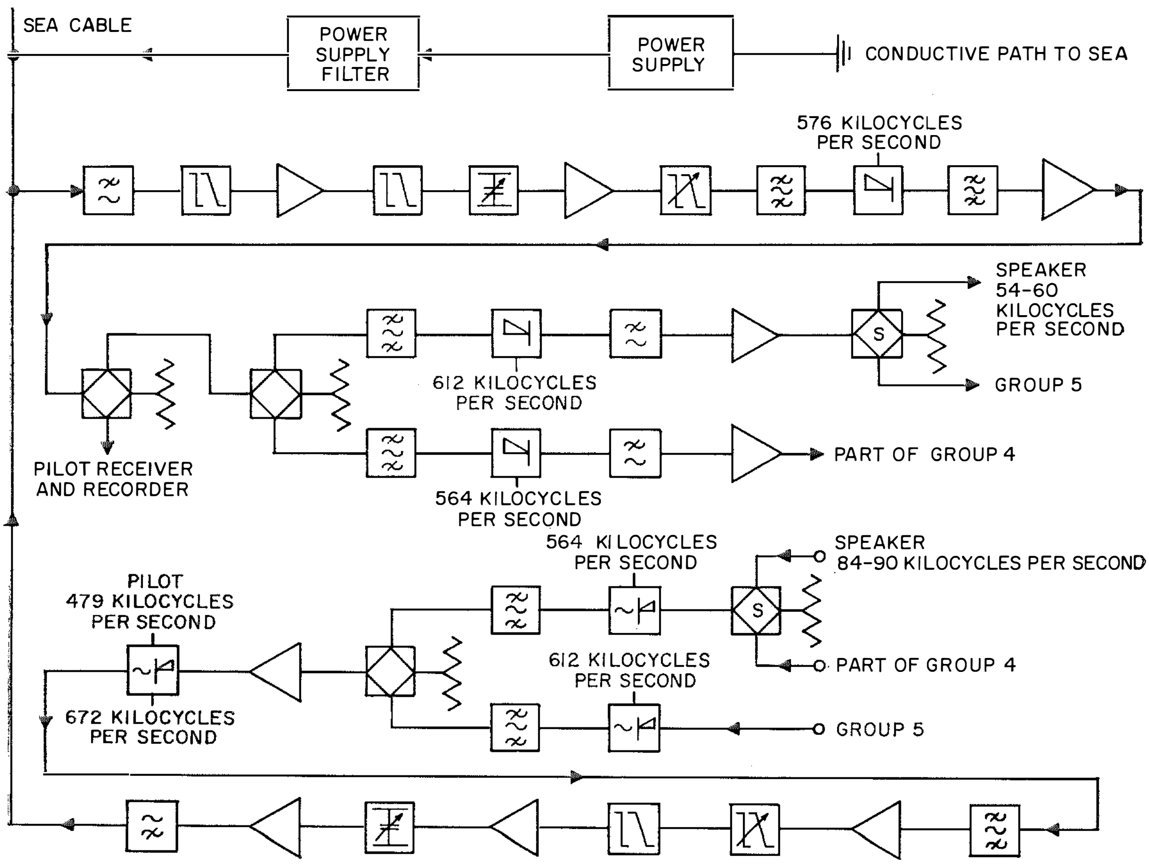


Figure 5—Simplified diagram of main transmission path at *B* terminal.

3. Manufacture of Submerged Repeaters and Equalisers

The submerged repeaters and equalisers were manufactured at North Woolwich in special fully air-conditioned shops often called "dairies." This manufacturing organisation was set up originally to produce the submerged repeaters for the Newfoundland–Nova Scotia link of the first transatlantic telephone cable [3]. Since that time, considerable experience has been gained and the facilities have been expanded and improved. To provide a thoroughly reliable repeater, considerable care must be taken with the mechanical and electrical design of its components. This care must start with the basic raw materials and continue throughout all the manufacturing processes. Most of the components are therefore manufactured within the company and those that are obtained outside are rigidly held to special specifications. The strict inspection pattern followed results in a large rejection rate for both raw materials and finished components, but this high standard of inspection in the early stages pays dividends in providing extremely uniform and stable units. The rejection of a complete repeater unit would be extremely serious in that it would indicate a weakness in the earlier inspection procedure.

A complete record of all components making up a repeater, of all processes, and of all inspection test results are filed individually for each repeater.

The submerged equalisers and their hermetically sealed components are manufactured in the same shops and to the same standard as the submerged repeaters.

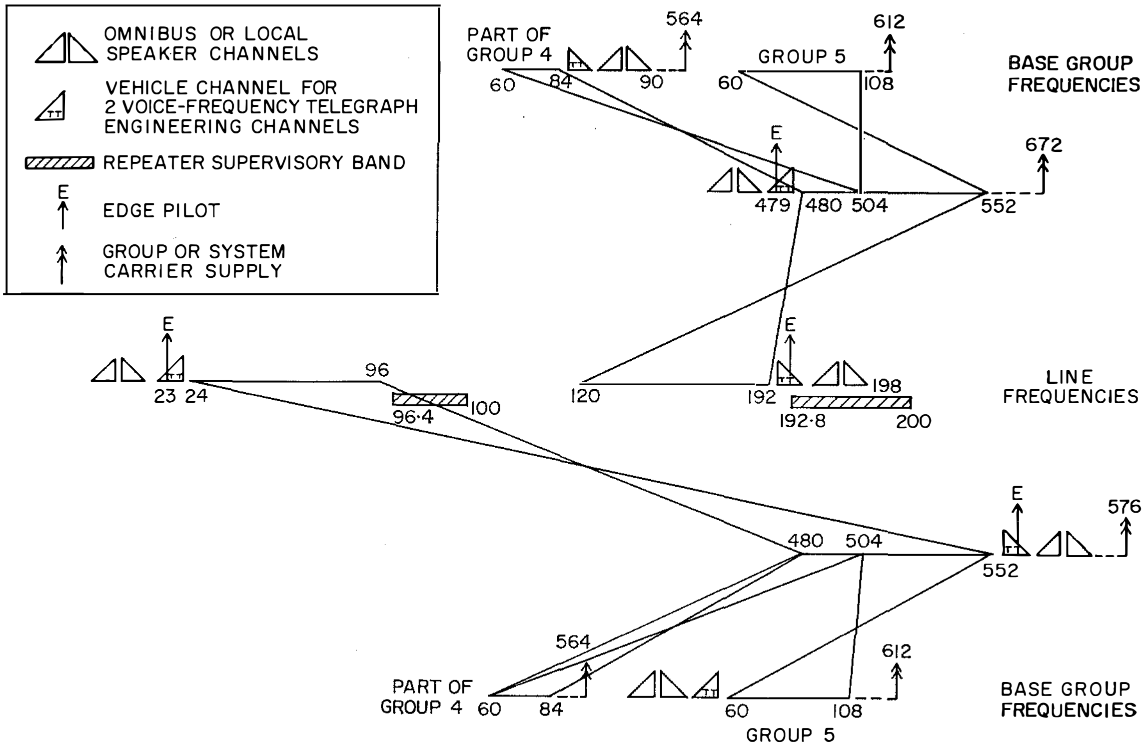


Figure 6—Frequency allocation.

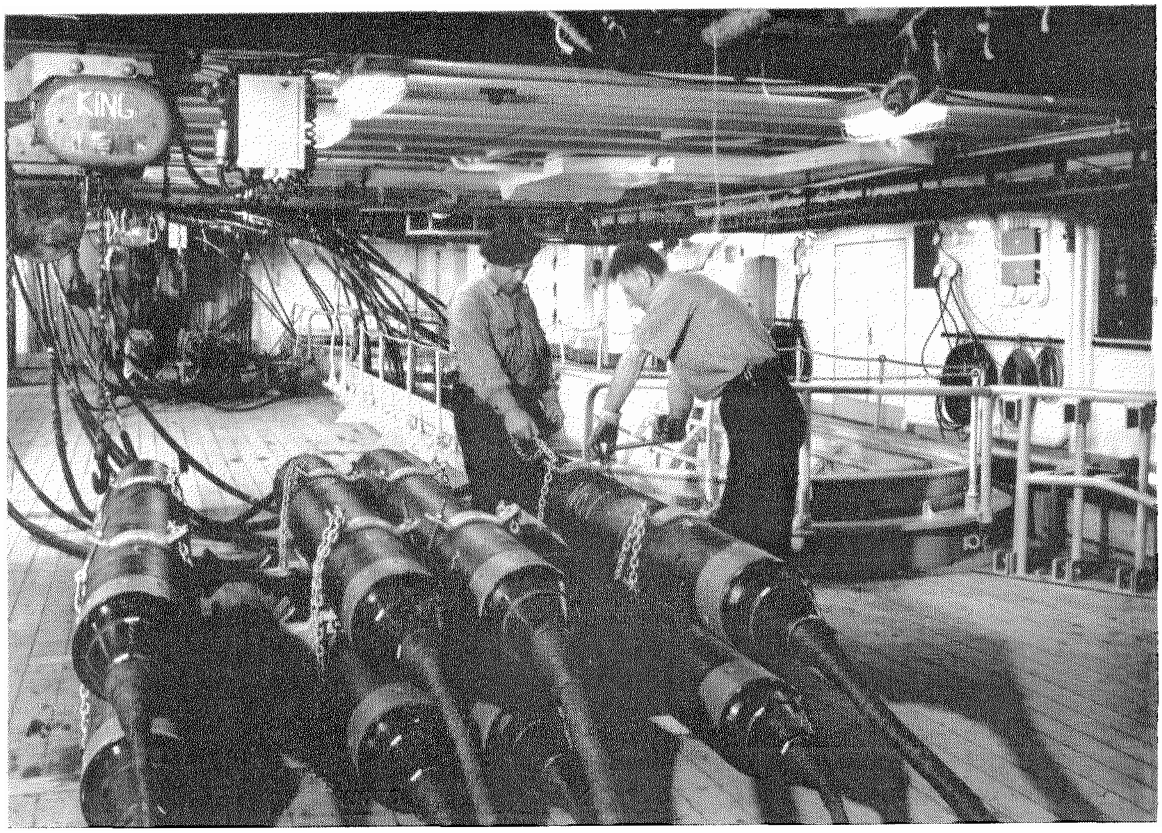


Figure 7—Centrecastle of *H.M.T.S. Alert* showing 8 repeaters in the foreground and 7 in the background for the North section of Scotice. They are connected to the cable that is stored in the cable tanks of the ship.

4. Cable Loading and Laying

The cableships were loaded direct from the submarine-cable factory at Southampton. The cable sections are approximately 27 nautical miles (50 kilometres) long and were linked by haulage hoses. While the cable loading continued at about 3 nautical miles (5·6 kilometres) per hour at Southampton, repeaters arrived at the rate of approximately three every two days from North Woolwich. Testing and jointing continued according to a coordinated schedule commencing with the last section to be laid. Both North and South sections were jointed to form 2 blocks, the first section of the second to be laid being longer than nominal to allow for length adjustment during the laying. Since the North section was longer, an equaliser was included between the 2 blocks. The following description applies to the laying of the North section.

H.M.T.S. Alert picked up the Faroes shore-end cable, which had previously been laid by a smaller cableship, made the initial splice, slipped the cable from bow to stern, and began to lay towards Iceland. Laying from the stern has the advantage of speed and the ability to work in heavy seas. The disadvantage is that there is a minimum speed, depending on weather, at which the ship can be controlled. During the weather experienced throughout

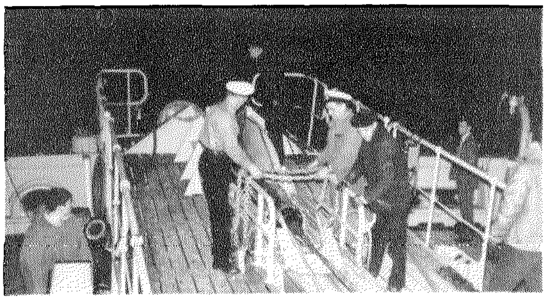


Figure 8—A repeater about to be laid.

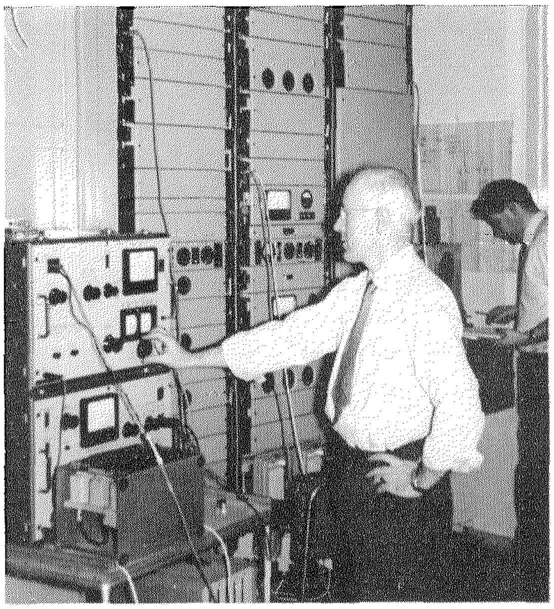


Figure 9—Testing the North section at Vestmannaeyjar.

this lay, in which the wind was usually of force 7 to 8, the minimum speed was about 2·5 to 3 knots. Thus, as the equaliser position was approached, a count-down of 16 hours was initiated, on the assumption that the ship's speed would be approaching 6 knots. The power unit aboard ship energised the repeaters continuously except during jointing operations. Thus, throughout the lay, communication with Faroes was maintained over the cable using carrier speaker equipment. As each repeater was laid, the shore station transmitted the supervisory signal to the last repeater in the block and observed the returned signal. Throughout the laying of the first block, measurements of insertion loss between Faroes and the ship as a function of frequency were made twice while laying the cable from one repeater to the next. This provided a careful check on the change in attenuation as the cable reached sea-bottom conditions. The insertion loss at each frequency was plotted against nautical miles of cable laid and by extrapolating the curves to the equaliser position a comparison with the target values for that point provided a forecast of equalisation error. The forecast equalisation requirement indicated that two bridged-T sections should be used to increase the loss at low frequencies. Since these sections had little loss at high frequencies, 1 nautical mile (1·9 kilometres) of cable simulation was added to the equalisers in each direction of transmission so that the lightly armoured deep-sea cable would not come too near the Iceland shore. The equaliser work including jointing both blocks together took less than 12 hours. The change in cable attenuation predicted for this period was very close to the actual change.

The equalisation procedure is described in detail elsewhere [5]. Throughout the remainder of the lay, insertion loss and supervisory measurements were made once per repeater section.

The over-all insertion-loss measurements made between Iceland and Faroes agreed very closely with the final measurements aboard ship, and at high frequencies with factory measurements of cable and repeaters. At low frequencies, the laying factor was slightly greater than predicted. Further over-all measurements made with equipment of the type shown in Figure 9 indicated that the design objectives were met and the circuits between London and Reykjavik were opened to traffic on 22 January 1962.

5. References

1. H. E. Madsen, "New Submarine Cable Scotland–Faroe Islands–Iceland," *Teleteknik,* volume 4, number 2, pages 40–43; 1960.

2. "The Scotice–Icecan Submarine Cable Projects," *Post Office Electrical Engineers Journal,* volume 53, page 56; April 1960.

3. R. A. Meers, "Manufacture of Submerged Repeaters for the Newfoundland–Nova Scotia Link," *Post Office Electrical Engineers Journal,* volume 49, pages 400–403; January 1957.

4. J. O. McNally, G. H. Metson, E. A. Veazie, and M. F. Holmes, "Electron Tubes for the Transatlantic Cable System," *Post Office Electrical Engineers Journal,* volume 49, pages 411–419; January 1957.

5. B. M. Dawidziuk and F. L. Jarvis, "Adjustable Submersible Equaliser," *Electrical Communication,* volume 37, number 4, pages 88–105; 1962.

6. R. A. Brockbank and C. A. A. Wass, "Non-Linear Distortion in Transmission Systems," *Journal of the Institution of Electrical Engineers,* volume 92, Part III, pages 45–56; March 1945.

7. R. O. Roe, "Improvements in Arrangements for Locating Defective Submerged Repeaters," British Patent 656 188; 17 February 1948.

8. Recommendations of Comité Consultatif International Télégraphique et Téléphonique Study Group 1, Geneva; April 1960.

6. Acknowledgments

The successful conclusion of this project is the result of the work of a team too numerous to name individually. The authors express their thanks to colleagues in the Lines and Submarine Branches of the British Post Office for assistance with the cable laying and over-all testing under difficult conditions. Thanks also to their colleagues in, and to the management of, Standard Telephones and Cables Limited.

7. Appendix—Performance of Typical Submerged Repeater

Insulation resistance $= 2 \times 10^4$ megohms.

Direct-current resistance at 22 degrees centigrade.

Current in Milliamperes	Direct-Current Resistance in Ohms
5	334·0
20	334·0
50	337·3
100	351·6

Voltage drop at 316 milliamperes = 120.

Insertion gain in a 52-ohm circuit.

Frequency in Kilocycles per Second	Gain in Decibels	Frequency in Kilocycles per Second	Gain in Decibels
24	22·33	120	48·18
30	24·59	130	50·17
40	27·93	140	52·07
50	31·02	150	53·95
60	33·99	160	55·71
70	36·71	170	57·41
80	39·26	180	59·03
90	41·57	190	60·60
98	43·53	196	61·47
100	43·98	200	62·02

Harmonic distortion in decibels referred to 1 milliwatt.

60-kilocycle-per-second fundamental level at *B* terminal +11

120-kilocycle-per-second-harmonic level at *A* terminal −63

180-kilocycle-per-second third-harmonic level at A terminal -70

Impedance, return loss in decibels against 52 ohms.

Frequency in Kilocycles per Second	A Terminal	B Terminal
20	35·5	29·0
40	24·0	27·0
70	19·0	12·0
100	12·5	8·0
120	11·5	11·0
140	14·0	15·5
170	19·5	26·0
200	21·5	22·5

Noise in decibels referred to 1 milliwatt.

A terminal between 120 and 200 kilocycles per second $= -61\cdot0$.

B terminal between 20 and 100 kilocycles per second $= -75\cdot5$.

Supervisory.

Fundamental Level at B Terminal in Decibels Referred to 1 Milliwatt	Second-Harmonic Level at A Terminal in Decibels Referred to 1 Milliwatt
−12	−21·6
− 2	− 8·4
+ 8	+ 2·7

M. V. Young was born in Ambleside, England, in 1918. He was educated at Denstone College and at the Manchester College of Technology, where he received the degree of B.Sc. Tech. in 1940.

He was commissioned in the Royal Corps of Signals and saw service in South East Asia Command and in Germany.

He joined Standard Telephones and Cables in 1946. After a period on coaxial line systems, voice-frequency telegraph systems, and telegraph repeaters, he has, since 1954, been associated with the over-all planning of submarine-cable systems and engineering of submarine-cable terminal equipment.

Mr. Young is an Associate Member of the Institution of Electrical Engineers.

W. J. Archibald was born in Scotland, in 1920. He was educated at Dollar Academy, Scotland. In 1941 he received the Higher National Diploma in electrical engineering at the Heriot-Watt College, Edinburgh, and became a graduate of the Institution of Electrical Engineers.

During the war he was commissioned in the Royal Electrical and Mechanical Engineers and served in the radar maintenance organisation of the Air Defence of Great Britain. He was later appointed Deputy Assistant Director of Mechanical Engineering for wireless, line communication, and radar equipment at General Headquarters, India.

In 1946 he joined the transmission division of Standard Telephones and Cables at North Woolwich. Until 1950, he was engaged in the design of audio systems. He then transferred to submerged repeater systems, working mainly on electrical circuit design.

Mr. Archibald is an Associate Member of the Institution of Electrical Engineers.

Shipboard-Adjustable Submerged Equaliser

B. M. DAWIDZIUK
F. L. JARVIS
Standard Telephones and Cables Limited; London, England

1. Introduction

The continual advances in the design of submarine cables and submerged repeaters have been responsible for the rapid increase in the number of submarine telephone cables that are now in operation or are planned for the near future [1, 2].

The laying of the first transatlantic telephone cable known as *TAT-1*, in 1956 [3], was so successful in providing high-grade telephone channels between the continents that it was decided within 3 years of its opening to traffic to provide another similar system, *TAT-2*, for an additional 36 circuits between the United States and Europe.

For good-quality telephone circuits, consideration must be given to the variation of signal level along the line, including equalisation errors caused by failure of repeater gain and cable loss to compensate for each other within the design margins. The equalisation errors have to be limited to, say, 3 decibels, if the use of a "mop-up" equaliser is to be avoided. Thus, on a transoceanic submarine cable for telephony, system equalisation is of paramount importance. Unfortunately, the exact properties of the equaliser depend on the characteristics of the cable and repeaters as laid and can therefore be accomplished only during the laying of the cable, possibly in difficult weather conditions.

Early oceanic submarine cable systems using the Bell Telephone System flexible unidirectional repeaters had a number of submerged equalisers having different fixed equalisation characteristics [4]. From a stock carried on board the cableship, equalisers were selected and spliced into the cable during the laying according to the system requirements. With the relatively low line frequencies involved, this method proved to be quite successful.

However, using a single cable with bidirectional rigid repeaters and working at higher line frequencies, successful operation of a long submarine cable system will depend largely on the ability to equalise the cumulative errors along the system at those intervals where the magnitude of the error approaches the system design limit.

Because of the rather involved nature of the system errors, the equaliser should be capable of being adjusted aboard the cableship and then laid in deep water in the same manner as the repeaters.

The pressure-resisting housing developed for our submerged repeaters and first used for the Denmark–Norway cable [5], being of the so-called demountable type, offered the possibility of adjusting an equaliser and enclosing it in a pressure housing on shipboard.

The first submerged equaliser using such a housing was adjusted during the laying of the Anglo–Swedish cable [6, 7]. Several equalisers of a similar type have also been laid during the installation of the Cantat system [8].

A detailed description of the type of adjustable submerged equaliser initially developed for the recently completed Scotice and *Med-1* projects is given in this paper. The pressure housing, being the key to the problem of providing a submerged equaliser that is capable of being adjusted on board ship, is dealt with in some detail.

2. Design Objectives and Sources of Equalisation Errors

The main design objective was an equaliser capable of being adjusted aboard the cableship with relative ease and speed and that provides a loss characteristic to compensate for any possible system errors. It is therefore worthwhile to examine the sources and nature of the system equalisation errors to appreciate the design of these networks. The errors may be either predictable or unpredictable.

2.1 Predictable Errors

Predictable errors are those that are known before assembly of the system aboard the cableship; they are as follows:—

A. Design errors E_d are the discrepancies between the gain of the laboratory model repeater and the expected cable loss. They are normally small and known at an early stage of project engineering.

B. Cable errors E_c cover the difference between the predicted cable attenuation characteristic and the actual cable loss figures as measured on the manufactured cable lengths in the factory. Their magnitude depends on whether the cable is of a new design or of a design that is well established and within the percentage loss limits given in the cable specification.

If the cable employed for the project is of a type already used on other systems, then E_c is small. It may be significant for a cable of new design for which only estimated loss figures were available when the final repeater gain-frequency characteristic had to be accepted.

C. Repeater errors E_r involve the difference between the repeater gain-frequency characteristics of the laboratory model and of the production units. Normally E_r is small. The over-all summation of these 3 errors gives the expected equalisation error E_e.

The frequency characteristic of E_e can be predicted to a fair degree of accuracy when the values for each of the contributing factors are known. The expected error can be compensated by a fixed equaliser unit, which as a practical matter must be designed at a late stage in the production of the cable and repeaters.

2.2 Unpredictable Errors

All changes in the cable characteristic during the laying are unpredictable errors. Such changes are truly unknown as to their extent and character, but they may be of two categories:—

A. Cable-length error E_l is a change in the cable attenuation that is approximately proportional to the cable attenuation constant; this type of error can be normally corrected by provision of facilities for cable-length adjustment at the end of each ocean block.

B. Cable-shape error E_s covers those cases of change in the frequency-loss characteristic of the cable during laying. Normally, significant changes in the shape of the attenuation curve are not expected, except perhaps, at the lower end of the system frequency spectrum.

3. Submerged Equaliser Requirements

3.1 Electrical Requirements

The examination of the possible error sources indicates that the equaliser should have a fixed equaliser unit to compensate for E_e and an adjustable equaliser to compensate for the unpredicted errors and for any residue of errors from inadequate equalisation of E_e by the fixed unit.

In some cases it may be difficult to design and manufacture a fixed equaliser in time for the laying operation; the adjustable unit will then have to provide for all the resultant errors, $E = E_d + E_c + E_r + E_l + E_s$. The equaliser must also provide the facility to insert cable simulator sections as and when required. In addition, power separating filters are needed to separate the transmitted signals from the power sent through the cable to operate the repeater amplifiers.

Directional filters separate the two directions of transmission, which simplifies the design of the adjustable equaliser sections as each direction of transmission can be considered independently.

3.2 Mechanical Requirements

From the mechanical standpoint, the design should aim to meet a number of requirements.

A. The pressure housing should, of course, be capable of being opened and subsequently closed and sealed on shipboard.

B. There should be some easy means of checking the sealing of the housing.

C. The completed equaliser should have the same expected reliability as that of the submerged repeaters in the system.

D. The equaliser should be capable of being completed on any cableship and not only on one specially equipped for this purpose.

E. The number of components for the adjustable units should be as few as possible consistent with flexibility and adequate coverage of foreseeable equalisation requirements.

F. All parts and components should be robust.

G. The design should provide for simple and speedy assembly and wiring on shipboard to reduce delay during the laying of the system.

It was appreciated that some special equipment would be needed to meet requirement *A*, which would have to be portable to comply with *D*.

4. Description of Equaliser

4.1 Mechanical Construction

From the mechanical aspect, the submerged equaliser can be divided into 3 main parts; the pressure housing, the apparatus unit that carries the fixed elements and also accepts the adjustable elements, and the parts and components for making up the adjustable elements.

4.1.1 *Pressure Housing*

The pressure housing, originally designed for submerged repeaters, was developed basically over a number of years in conjunction with Messrs. Vickers Armstrong [9]. Refinements were added as a result of further development carried out in conjunction with the British Post Office. The housing is suitable for working pressures up to 4 tons per square inch (630 kilograms per square centimetre); this is equivalent to a maximum depth of water of 3400 fathoms (6218 metres).

In principle, the housing consists of a high-tensile-steel tube or casing with its ends closed by high-tensile-steel bulkheads sealed into the casing by means of *O* seals. No elaborate equipment is required on shipboard to carry out the sealing operation.

Figure 1 is a sectional view of part of one end of a pressure housing, showing the sealing of the bulkhead. The space to the left from the coil of slack cable needed during assembly to the outer face of the bulkhead is open to the sea. The bulkhead, which is a close fit in the end of the casing, has two main *O* seals carried in grooves round its periphery. The space between the seals is connected via small radial holes with 2 cylinders bored into the bulkhead from the outer face; both of these cylinders are sealed against the entry of water by close-fitting pistons equipped with *O* seals. In the lower of the two cylinders, a small hole extends through the bulkhead to permit filling the inside of the housing with dry nitrogen, and the inner end of the cylinder is also sealed by a piston and *O* seal. In addition, the small hole is also sealed off by a soft copper disk attached to the end of the piston, which is forced into intimate contact with the metal at the bottom of the cylinder by a load of 3·5 tons (3600 kilogrammes); this load is applied via the piston by tightening its clamping screw with a pre-determined torque. The purpose of the disk is to provide a diffusion seal against the possible entry of water vapour.

The enclosed space formed by the 2 main *O* seals and the sealed-off cylinders is filled with petroleum jelly, and the jelly compressed to a pressure of 4 tons per square inch (630 kilogrammes per square centimetre) at an ambient temperature of 22 degrees centigrade by means of the pressurising piston and its pressure screw in the lower of the two cylinders shown in Figure 1. Thus, in addition to the sealing afforded by the *O* seals, there is a barrier of petroleum jelly that is held at a pressure in excess of that of the surrounding water.

In very-deep water, where the pressure is high and the temperature low, possibly of the order of 5 degrees centigrade, then due to the pressure/temperature coefficient of the petroleum-jelly system, which is approximately $\frac{1}{16}$ ton per square inch per degree centigrade (9·8 kilograms per square centimetre per degree centigrade) the pressure in the system would tend to fall to a value that would possibly be less than that of the water. To maintain the

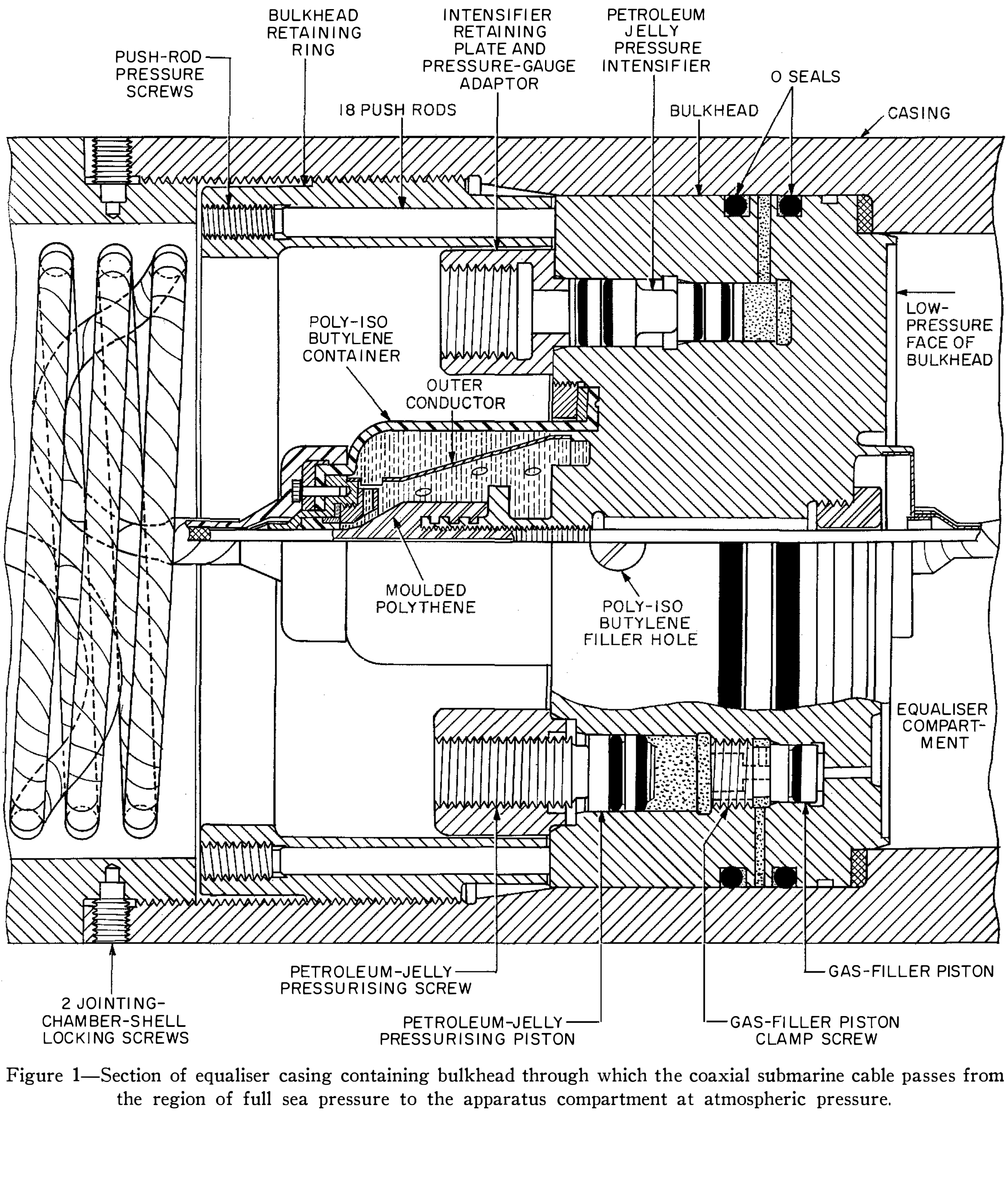


Figure 1—Section of equaliser casing containing bulkhead through which the coaxial submarine cable passes from the region of full sea pressure to the apparatus compartment at atmospheric pressure.

pressure in the petroleum-jelly system always greater than that of the water, the other cylinder in the bulkhead is provided with a pressure-intensifying device; this device will automatically raise the pressure in the system to a value that is proportionately greater than the pressure of the surrounding water. This is achieved by having the outer portion of the cylinder bored to a larger diameter than the inner portion and providing each portion with a separate piston, the inner piston being in contact with the contained petroleum jelly and the outer piston being exposed to the sea. As the ratio of the piston diameters is 1·36:1, the pressure in the system will therefore always be at least 1·36 times that of the external water pressure. The available travel of the pistons is sufficient to provide a maximum increase in pressure of the petroleum jelly of about 4·5 tons per square inch (709 kilogrammes per square centimetre).

All *O* seals are moulded of special hard-grade synthetic rubber that is particularly resistant to oil and sea water. The hardness of the seals is such that, when used with the small working clearances provided, there is no detrimental extrusion of the seals, even when they are subjected to pressures of the order of 7 tons per square inch (1102 kilogrammes per square centimetre).

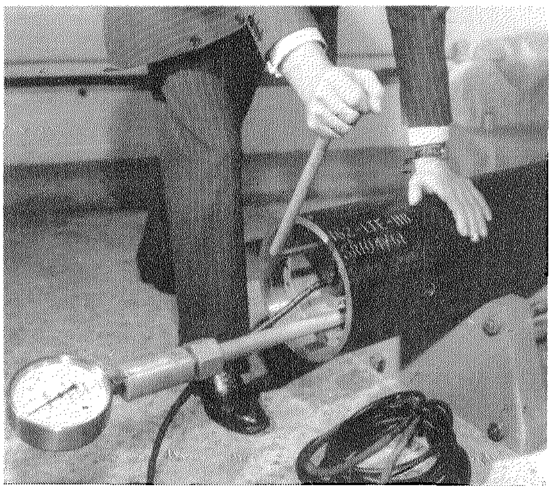


Figure 2—Measuring device showing pressure of petroleum-jelly seal. Neither pressure nor jelly is lost in connecting the device or in making a measurement so measurements may be repeated at any time without penalty.

The plate attached to the face of the bulkhead to retain the intensifier pistons is provided with a large threaded hole that is extended by a smaller plain hole through the plate to give access to the face of the outer piston. This hole, apart from giving sea water easy access to the piston, also provides for applying a pressure-measuring device to the bulkhead to determine the pressure of the petroleum jelly during the compressing operation. This measurement may be repeated at any time to check the sealing of the bulkhead. Figure 2 shows the device in operation. It should be appreciated that this pressure measuring device can be applied to, or removed from, a bulkhead without loss of pressure or petroleum jelly. This design, incorporating a pressurising system and a sensitive means of measuring the pressure, provides a ready method of checking the effectiveness of the seal. With the pressure-measuring device applied to the bulkhead during the pressurising operation, any defects in sealing will be detected by a fall in pressure. This test is quite sensitive because, as the volume of the bulkhead pressure system is very small, of the order of 2 or 3 cubic inches (33 to 49 cubic centimetres), any slight leak will show up as a drop in pressure in quite a short time. The procedure on shipboard, therefore, is to leave the measuring device in position on the bulkhead for a short time after pressurising to check sealing.

Between the seating on the inner face of the bulkhead and the bulkhead-retaining shoulder in the casing, a lead gasket is fitted as a diffusion seal against the possible entry of water vapour that may have passed the *O* seals. In fairly deep water where the pressure is of the order of, say, 1 ton per square inch (157 kilogrammes per square centimetre), it is evident that the bulkhead will be forced against the lead gasket with sufficient pressure to "flow" the lead so that it makes intimate contact with the adjacent steel faces and provides an adequate seal. As the areas of the bulkhead and gasket are approximately 63 and 13 square inches (406 and 84 square centimetres) respectively, a water pressure of 1 ton per square inch (157 kilogrammes per square centimetre) will produce a stress on the lead of nearly 5 tons per square inch (787 kilogrammes per square centimetre). However, to cater for comparatively shallow water where the pressure would be insufficient to stress the gasket beyond the flow limit of lead, arrangements are provided to pre-stress the gasket with a load of approximately 45 tons (45 720 kilogrammes). With this load on the gasket, the stress will be approximately 3·5 tons per square inch (551 kilogrammes per square centimetre). (The flow stress of lead is approximately 2·5 tons per square inch (394 kilogrammes per square centimetre).) To achieve the initial pre-stressing of the lead, 18 equispaced push-rods, sliding in close-fitting holes in the bulkhead retaining ring, bear at their inner ends against the face of the bulkhead and are provided at their outer ends with pressure (thrust) screws. These screws are tightened with a torque that gives a thrust of 2·5 tons (2540 kilogrammes) on each rod. The compressed push-rods constitute very strong springs that provide a small amount of "follow-up" without excessive loss of pressure in the event of a subsequent slight extrusion of lead between the bulkhead and the casing. The risk of extrusion, however, is very slight, as a test on a gasket fitted in a housing having normal clearances between the bulkhead and the casing has shown that, even with the bulkhead loaded with over 300 tons (305 megagrammes), apart from an initial slight extrusion that took place during the first few hours, there was no further extrusion over the period of test which lasted for 135 days.

From the foregoing it will be seen that the sealing of the bulkhead into the casing has 3 lines of defence; namely, 2 *O* seals and the lead gasket. In addition there is a barrier of petroleum jelly held at a pressure that is always greater than that of the surrounding sea water. Thus, even in the event of the failure of an *O* seal that would completely release the petroleum-jelly pressure, there will still be an *O* seal and the lead gasket. Furthermore, it will be appreciated that the sealing operation can be carried out and checked using only a few simple tools.

The cable entry through the bulkhead is formed by a castellated stem machined integral with the bulkhead, through which passes a small-diameter polythene-insulated cable, the cable forming short tails on each side of the bulkhead. A polythene moulding is formed on the cable at the outer end which unites homogeneously with the polythene of the cable and also extends over the castellated portion of the stem. The contraction of the moulding on to the stem will provide a seal against the entry of water, but as an additional safeguard the stem is processed before moulding to ensure an effective bond between the polythene and the metal of the stem. In deep water, the contraction of the moulding on to the stem will be enhanced by the external pressure of the water. The bore of the stem is threaded to form circumferential serrations into which the softened polythene of the cable is forced during the moulding operation; this provides an effective keying that prevents extrusion of the cable through the bulkhead, even with hydraulic pressure of 5 tons per square inch (787 kilogrammes per square centimetre). In fact every production seal is subjected to a water pressure test at 5 tons per square inch for a period of a month; during the test a daily check is made of humidity in an airtight container sealed over the low-pressure inner face of the bulkhead. Whilst under pressure all seals are also electrically checked for insulation resistance and ionisation-inception voltage, and are subjected to a proof test of 40 kilovolts of direct current for 1 minute. This type of seal was developed by the British Post Office and is used on all their deep-water submerged repeaters.

The cable tail on the high-pressure side of the bulkhead is provided with a tinned-copper braid outer conductor that is connected to the bulkhead via a perforated metal cup. Completely enclosing the moulded cable seal and the outer conductor is a moulded synthetic-rubber bell that is vacuum filled with a viscous insulating fluid (poly-iso-butylene) [10]. Thus, water is prevented from contacting the junction of dissimilar metals formed by the outer-conductor cup and the bulkhead; this latter would, in the presence of water, cause electrolytic corrosion of the metal. When subjected to a hydrostatic pressure the bell will, of course, collapse slightly until the pressure in the insulating fluid is comparable with that of the surrounding water.

The cable tail on the low-pressure side of the bulkhead is jointed to a corresponding tail from the apparatus unit, which is contained within the pressure housing. The outer-conductor braid of the apparatus unit is extended over the bulkhead tail and connected to the bulkhead via a thin metal cup; this operation is carried out during the assembly of the submerged equaliser in the factory.

When dispatched from the factory, the tail cable on the high-pressure side of the bulkhead is jointed to a co-axial test lead; during installation operations on board the cableship, this lead is removed, and the tail is jointed to a corresponding tail on the pre-terminated end of the sea cable; after jointing the outer conductors of the tails, the whole of the tail cables are protected by layers of self-adhesive polythene and polyvinyl-chloride tapes.

The method of anchoring the sea cable to the end of the housing, shown in Figure 3, follows the commonly accepted practice whereby the armour wires of the cable are formed back over a tubular coned member. This armour cone is threaded onto the cable and the wires are clamped between it and a surrounding anchor plate having a hole complementary in shape to the outside of the armour cone. The armour wires, where they are formed back round the nose of the armour cone, are individually located in radial slots in an anti-rotation plate to prevent any possible torsional movement of the sea cable relative to the pressure housing during the laying operation.

By adopting a removable collet made in two halves and fitted into the anchor plate to form the outer coned clamping member, the ends of the sea-cable sections may be terminated before loading onto the cableship. This terminating operation includes jointing-on of the small-diameter cable tail by means of a tapered moulded joint, jointing of the outer-conductor braid of the tail to the outer-conductor tapes of the sea cable, forming back of the armour wires over the armour cone, and laying-up and serving with spun yarn the turned-back armour wires. This greatly reduces the amount of work to be performed on the ship when installing the equaliser.

It is interesting to note that this pressure housing has been adopted by the British Post Office for shipboard-adjustable submerged equalisers on trans-oceanic cable systems such as Cantat and Compac.

4.1.2 *Apparatus Unit*

The apparatus unit within the pressure housing consists basically of 2 parts, a factory-sealed capsule that houses the fixed electric units of the equaliser, and an open chassis, flexibly attached to one end of the capsule, into which the adjustable units are mounted on shipboard. The complete apparatus unit, prior to assembly in the pressure housing, is shown in Figure 4. When assembled in the housing the free end of the chassis is rigidly secured to the removable bulkhead; thus on board ship, when the bulkhead is withdrawn from the casing the chassis is also withdrawn just clear of the casing to enable the adjustable units to be fitted. Polytetrafluoroethylene rubbing pads, which are a close fit in the casing, are secured to the end of the chassis and to the free end of the capsule, to allow the apparatus unit to slide freely in the casing during the withdrawal operation.

The sealed capsule houses the power-separating filters, directional filters, and a fixed equaliser, if one is needed, and is filled with dry nitrogen. Cable tails, which are subsequently jointed to the cable tails of the respective bulkheads, are led into the capsule via seals in the end plates. In this manner, all high-voltage connections are sealed into a dry atmosphere and are not exposed during shipboard operations. The connections from the directional filters to the adjustable units are made via large ceramic-insulated terminals sealed through the end of the capsule.

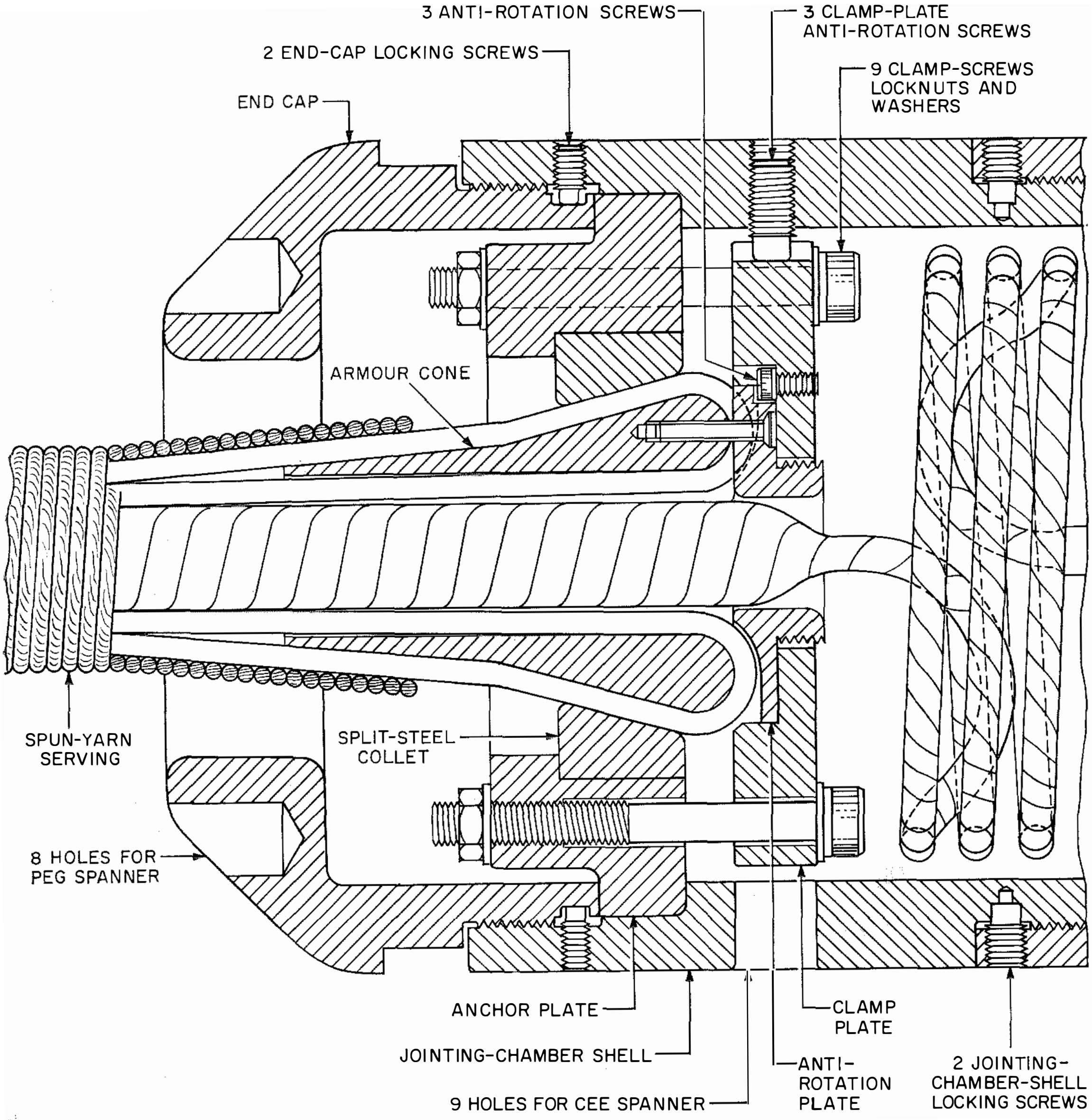


Figure 3—End of equaliser casing to which the submarine cable is anchored. All of this section is at sea pressure.

It should be noted that these terminals will be required to carry only external wiring that is of comparatively low impedance to ground.

The framework inside the sealed capsule consists of 4 longitudinal brass bars attached at their ends to the castings forming the end plates of the capsule; the framework is similar in principle to the open chassis shown in Figure 4. Within and secured to these bars, are mounting plates carrying the electric components. As in standard submerged-repeater practice, no metals are employed in the construction that are known to promote the growth of metallic whiskers. All insulation is of cast methacrylate (Perspex) and in those positions where large differences in potential may occur, long leakage paths and ample air clearances are provided that are adequate to prevent breakdown or ionisation when high line voltages have to be accommodated.

All components are made under dairy conditions of controlled humidity, temperature, and cleanliness, and are rigorously tested before acceptance. The assembly and wiring of the components are also carried out under similar dairy conditions.

After inspection of the wired assembly, the brass tube forming the shell of the capsule is fitted, and the ends are sealed to the 2 end castings. The interior of the capsule is then

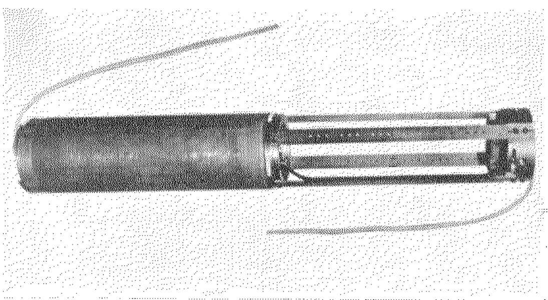


Figure 4—Apparatus unit. The closed section contains the fixed electric units assembled in the factory. The open structure will hold the adjustable electric units assembled on shipboard.

thoroughly dried by flushing through with dry nitrogen, introduced through small holes in the end castings, for a period of 24 hours; after which the holes in the castings are finally sealed.

The open chassis section of the apparatus unit for mounting the adjustable units of the equaliser is formed by 4 steel mounting bars secured at their ends to gunmetal rings by screws and locating dowels; these latter are fitted to facilitate accurate replacement of a bar after it has been removed to allow the units to be mounted. The end ring of the chassis adjacent to the sealed capsule is secured to the periphery of a flexible diaphragm, which in turn is secured at its centre to the end casting of the capsule; holes in the diaphragm clear the projecting terminals and cable tail on the end of the capsule. The larger ring at the other end of the chassis carries the fixing screws for attachment to the removable bulkhead of the pressure housing and also forms a coiling-down space for the cable tail after it has been jointed to the corresponding tail of the bulkhead. Attached to the inner face of the larger ring is a pocket to accept a desiccator of silica gel which is slipped into position immediately before re-insertion of the apparatus unit into the housing after fitting the adjustable units on shipboard.

There are 6 sets of fixing holes in the mounting bars of the open chassis that will each accept either 1 adjustable equaliser section or 2 cable-simulator units. Thus, a total of 6 equaliser sections or 12 cable simulators, or a combination of both, can be accommodated. Figure 5 shows an assembly of 1 equaliser section and 1 cable simulator in each of the 2 transmission paths, mounted in the chassis and wired to the appropriate terminals of the capsule ready for re-insertion into the housing.

4.1.3 *Adjustable Units*

To meet the standard of manufacture set by the fabrication of the submerged repeaters, no component should be exposed at any time other than in an air-conditioned dairy atmosphere. Whereas this requirement is easily met for the fixed elements by sealing them in a capsule before leaving the factory, the problem is not quite so simple for the adjustable units. Two possible lines of approach were considered initially:

A. To set up a working space on the cableship having full dairy conditions.

B. To fit all components into individual hermetically sealed cans before leaving the factory.

The objection to *A* is obvious; if the ship is not already equipped with a dairy, it would be very costly to provide the necessary air conditioning on a temporary basis, even assuming that space would be available. Although *B* also has objections, including those of cost and bulkiness, it offered the best alternative. It also offered the advantage that there would be no comparatively fragile components to handle under the far-from-ideal conditions that exist on a ship.

The final solution was to seal a number of components of the same type into 1 can to provide a range of values in the largest number of steps. It was decided to supply a number of cable simulators as factory-sealed units having different values of equivalent cable length (insertion loss).

To simplify assembly of the equaliser sections, all component units of a particular type were made identical in mechanical size, fixing, and number and position of terminals. Thus, physically, any equaliser section can be made up using only 4 basic parts; namely, resistance unit, capacitance unit, inductance unit, and mounting chassis. All component units are manufactured under full dairy conditions as outlined for the sealed capsule in Section 4.1.2 and are flushed and filled with dry nitrogen before sealing.

The resistance unit contains 10 wire-wound resistors connected to form a constant-impedance bridged-T pad having tapped series and shunt arms; all tapping points are brought out via seal terminals. Only one type of resistance unit, of the necessary impedance, is required for a particular cable system.

The capacitance unit contains 3 capacitors wired in series with their junction points and the free ends brought out via seal terminals. Thus, by suitably arranging the external wiring to the terminals, a wide range of capacitance values can be obtained from 1 unit. For a particular cable system, for example, a 120-circuit system, the required over-all range of capacitance can be covered by a total of 5 different units.

The inductance units each contain 3 inductors having carbonyl iron or ferrite cores with their leads brought out separately via seal terminals. Each unit will give a wide range of inductance values, and a total of only 4 different units are necessary to cover the required over-all inductance range for a particular system.

The mounting chassis for the component units comprises a mounting plate and a spun copper tray secured to the plate; identical drillings in both members allow the cans of components to be mounted on the underside of the plate with their terminals projecting up into the tray. The chassis will accommodate 5 component units;

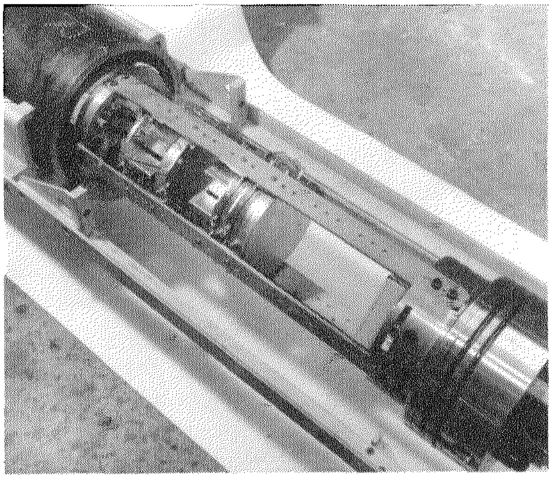


Figure 5—Adjustable units assembled and wired into chassis on shipboard before re-insertion into casing.

a resistance unit and series and shunt capacitance and inductance units. It is supplied fitted with individual blanking-off plates on the underside to cover the drillings for the series and shunt capacitance and inductance units; the plates are subsequently discarded in those positions where the electric design of the equaliser section necessitates the fitting of a component unit. Secured through the wall of the tray are 2 ceramic-insulated terminals having long leakage paths, and 2 metal terminal posts; these will form the input and output terminals of the completed equaliser section. Lugs are provided on the mounting plate for securing into the open chassis of the apparatus unit.

Figure 6 shows an assembled and wired equaliser section having resonant circuits in the series and shunt arms; thus, component units are mounted in all positions. The illustration shows clearly the simple nature of the assembly and wiring operations (to robust terminals) that are required to be performed on board ship.

It will be appreciated that the wiring to the component-unit terminals may be of high impedance to ground; to avoid deterioration of this condition, the tray containing the unit terminals and wiring is flooded with a sealing compound. The compound, which is poured at a temperature of 85 to 90 degrees centigrade, sets in approximately 20 minutes to a state that allows handling of the equaliser section. The heat-sink formed by the mounting chassis and unit cans prevents any undue rise in temperature of the components during the sealing operation.

The exposed input and output terminals of the completed equaliser section, being of low impedance, will not cause any significant deterioration in performance in the unlikely event of condensation forming on the insulators. However, as a further precaution, as has previously been stated, a silica gel desiccator is included in the open chassis of the apparatus unit to dry out the interior of the pressure housing after sealing.

The cable simulators use a mounting plate similar to that for the equaliser sections. Secured to the plate is a hermetically sealed can containing the electric components, which are mounted on a sub-plate of cast methacrylate sheet fitted into the can. For the external wiring, 2 ceramic seal terminals having long leakage paths and 2 metal earth posts project from the side of the can; as in the case of the equaliser sections, those exposed terminals are of low impedance to ground. The cable simulators are made in the factory under full submerged-repeater dairy conditions, and are flushed and filled with dry nitrogen before sealing.

The positions of the terminals on the cable simulators in relation to the lugs on the mounting plate are the same as those on the equaliser sections. This simplifies wiring between units mounted in the open chassis of the apparatus unit. Wiring bushes in the mounting plate, corresponding to similar bushes in the mounting plate of the equaliser sections, carry the interconnecting leads between units and provide intermediate supports for those leads that might otherwise be unstable due to their length. The wiring between units may be seen in Figure 5.

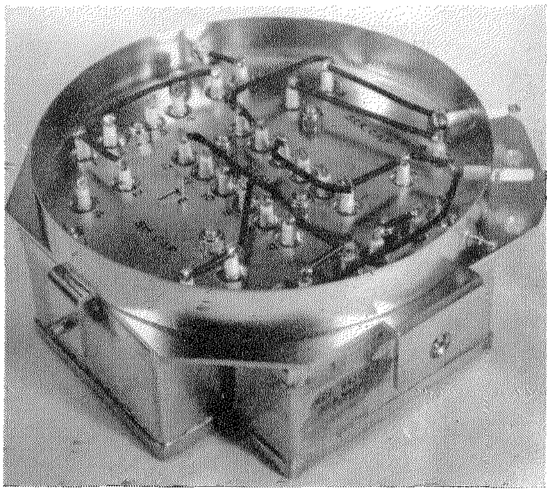


Figure 6—Assembled and wired adjustable equaliser section before the tray containing the terminals and wiring is filled with sealing compound.

4.2 Electric Circuits

At the very beginning, it was decided to provide, as far as possible, a common arrangement for the power separating filters. It was also decided that the adjustable equaliser circuits should be adequate for the 2 projects on hand, namely, the 18-circuit Scotice system with a frequency range from 20 to 200 kilocycles per second and the 120-circuit *Med-1* system with a frequency range from 60 to 1168 kilocycles per second.

A design covering both of these systems would then automatically cover a 60-circuit system with a frequency range from 20 to 552 or from 60 to 608 kilocycles per second.

4.2.1 *Power Separating Filters*

One of the basic considerations in the design of the power separating filters is the total value of capacitance required to be subjected to the line voltage stresses; it was decided to use the same volume of high-voltage capacitors in the equaliser as in the power separating filters in the submerged repeaters. On this basis, it was possible to develop the circuit shown in Figure 7, which provides for full separation of line voltage and line current from the transmission circuits. It also gives a loop-loss performance meeting the design requirement of 40 decibels minimum loss within the system frequency range.

Typical loss curves are shown in Figure 8. Curve *A* is for the 18-circuit system and curve *B* covers the case of 60- and 120-circuit systems.

To reduce the total volume of the capacitors required for the adjustable equaliser sections and directional filters, the nominal line impedance of 53 ohms is transformed to 75 ohms for 120- and 60-circuit equalisers and to 150 ohms for 18-circuit equalisers. The selected values of impedance match the impedance of standardized transmission-measuring equipment.

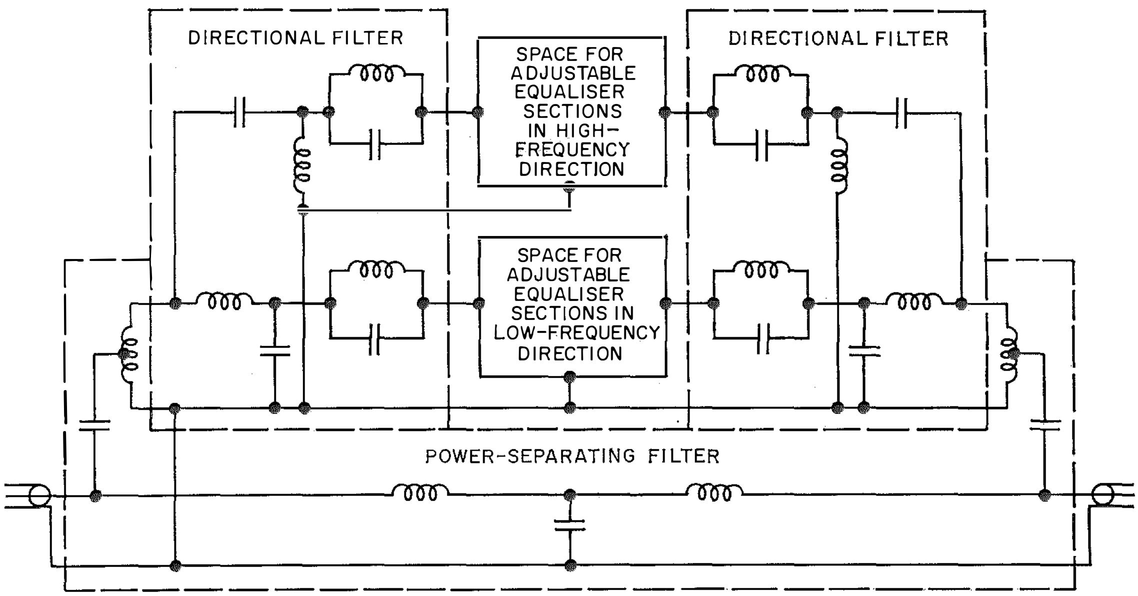


Figure 7—Submerged equaliser.

4.2.2 *Directional Filters*

Directional filters are fitted to ease the problem of designing the adjustable sections aboard the cableship. Since the equaliser is a passive device, it was decided that the minimum stop-band loss should be greater than 40 decibels. This requirement was met with only 2 half-sections in each path.

Directional filters are the only units in the equalisers that are dependent on the frequency range of the system.

4.2.3 *Adjustable Equaliser Sections*

As seen from the analysis of possible system errors outlined in Section 2, the adjustable equaliser sections must provide truly variable loss characteristics. Practical considerations, however, dictate definite limits to the possible range of available characteristics, and the following targets were set at the design stage:

A. Each adjustable equaliser section should provide any one of the 4 types of curves shown in Figure 9.

B. Each section to be of the constant-impedance type [11] with a maximum insertion loss adjustable in steps of 0·5 decibel within the range from 2 to 10 decibels.

C. The range of inductance and capacitance values must provide an insertion loss for the type-C equaliser of Figure 9 having a shape factor b from 1·2 to 4 where $b = f_r/f_b$, f_r = resonance frequency or maximum-loss point A_m for type C, and f_b = frequency where loss is half the maximum loss $A_m/2$.

In addition, the component values must provide for f_r to be any frequency between f_1 and f_2 where $f_1 = 0\cdot5 \times$ the lowest system frequency. For the 18-circuit system, $f_1 = 0\cdot5 \times 20 = 10$ kilocycles per second. $f_2 = 1\cdot5 \times$ the highest system frequency. For the 120-circuit system, $f_2 = 1\cdot5 \times 1\cdot2 = 1\cdot8$ megacycles per second.

To cover the above requirements, the range of adjustable capacitance units and adjustable inductance units were designed as follows.

4.2.3.1 *Capacitor Units*

The values follow the ratio 1:2:4, and terminals on the can allow for external series-parallel connections of the 3 capacitors.

The ratio 1:2:4 gives the widest range of values when employing all possible 16 combinations while providing values of capacitance at steps not exceeding 33 per cent. By suitably spacing the basic values of the capacitors finally selected for the 5 capacitance cans, a range from 160 picofarads to 0·407 microfarad has been obtained in steps that on the average are 10 per cent apart, with a small number of isolated cases where steps may reach 25 per cent.

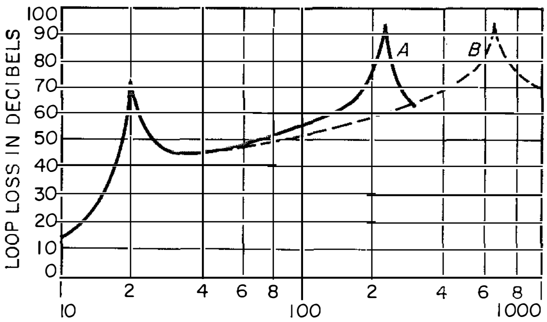


Figure 8—Loop loss of power separating filter.

4.2.3.2 *Inductance Units*

The method used to provide adjustable inductance is similar to that employed for capacitance units. Again the values are in the ratio 1:2:4, giving a total available range from 5 inductance cans of 0·77 microhenry to 6 millihenries in steps of about 10 per cent with isolated cases of a maximum 17-per-cent gap between the adjacent inductance values.

There are many intermediate values with spacings of less than 10 per cent because to cover the required frequency range from 10 kilocycles per second to 1·8 megacycles per second and to meet a minimum Q value of 60, some coils have been designed to cover only the high frequencies, normally low inductance values, and other units are for low-frequency work and have high inductance values.

4.2.3.3 *Resistance Unit*

By suitably strapping the external terminals of the resistance unit, 23 different loss values A_m are obtained in 0·5-decibel steps between 2 and 10 decibels with additional intermediate values. Of the 2 different designs available, one is a unit of 75-ohm impedance for the 120- and 60-circuit equaliser, and the other is a 150-ohm-impedance unit for the 18-circuit equaliser.

4.2.4 *Cable Simulators*

Simple equaliser sections having loss characteristics simulating the loss of a cable length are provided with loss values at the top system frequency from 1·5 to 12 decibels in 1·5-decibel steps. A selection of cable-simulator units is

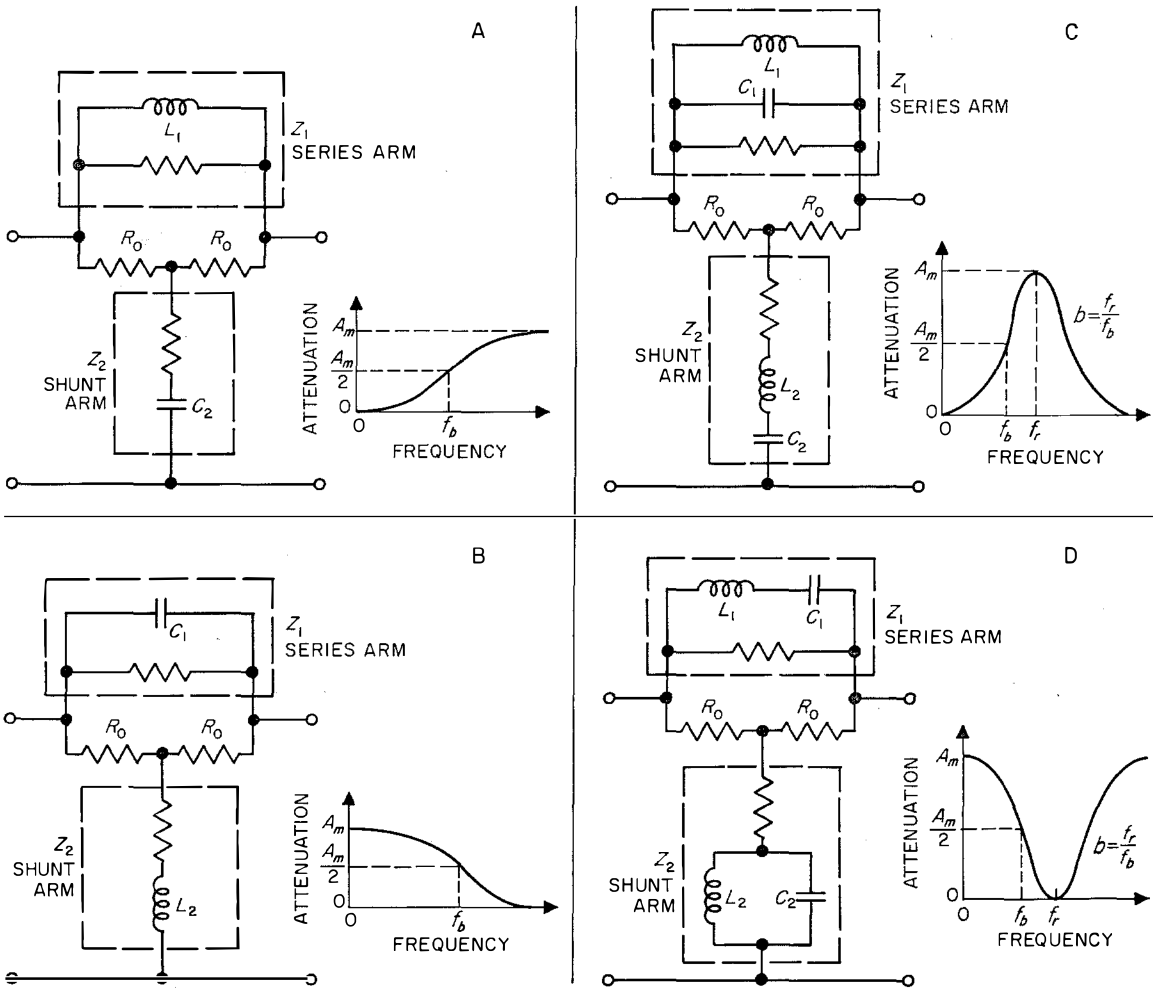


Figure 9—Circuit and attenuation-frequency characteristics of the 4 types of equaliser sections that must be provided for in the adjustable equalisers.

carried on the cableship during the laying; they are used as and when required.

5. Shipboard Assembly

One of the essential requirements during the laying of a deep-water submarine cable is to maintain a continuous paying out of the cable at normal laying speed with the correct laying tension. To avoid stopping the ship, the complete equaliser, including any cable length adjustment, must be completed before the ship reaches the position where the equaliser must be laid. Thus, from the beginning of the lay, results of the transmission measurements between the terminal station and the ship are plotted on a graph showing the system loss against the length of the laid cable. Curves are drawn for all test frequencies (say 12 frequencies in each direction of transmission). The points are normally on a straight line, so that after some time of cable laying the system loss at the end of the ocean block can be extrapolated for each of the test frequencies. The estimated system levels are then compared with the ideal system levels at this point, and the difference between the two gives the equaliser target characteristic.

5.1 Circuit Design

When the equaliser requirements are known, the designer decides on the types and number of sections to be used. The results are plotted on transparent logarithmic graph paper and are checked by superimposing the graph on curves showing the shape characteristics of equaliser sections with different values of shape factor b. A typical set of curves for a type-C section with $A_m = 3$ and 6 decibels is shown in Figure 10.

The equaliser designer then decides on the design parameters such as frequency of maximum loss, shape factor, loss value, half-loss frequency, et cetera, and estimates the basic equaliser loss. Then the cable length that is to contain the submerged equaliser is shortened to provide the effective system gain equivalent to the basic equaliser loss.

Calculated element values are checked against the available components by reference to tables. The value giving the minimum percentage difference from the design requirement is normally selected.

To provide an additional check on the calculations all available resonant frequencies have been tabulated by a computer as a function of the capacitance and inductance values, and such tables are available to the designer. The expected insertion loss of each section can be quickly calculated either by reference to a series of graphs similar to Figure 10 or by the use of tables relating design parameters to the loss characteristic of the particular type of equaliser section to be used.

After a check of expected loss characteristic the designer draws a wiring diagram from which the assembly and wiring are done. After a wiring inspection the loss characteristic is measured, and the equaliser section finally sealed as described in Section 4.1.3.

When the individual equaliser sections are complete, they are inserted into the open chassis of the apparatus unit and interconnected to give the required loss characteristic in each direction of transmission.

The total time required to design, check, wire, inspect and seal 6 equaliser sections and then fit these sections into the chassis, interconnect the units, close and seal the housing, and joint the completed equaliser into the cable, is estimated to be as long as 16 hours for a team of 5 people.

5.2 Equaliser Assembly

The description in Section 4 gives a fairly clear picture of the operations that are required on board the cableship to complete the assembly and wiring of the submerged equaliser and to seal the pressure housing; nothing has been said however about the actual operation of withdrawal of the apparatus unit from the housing and the subsequent re-insertion. It was appreciated that sufficient head-room was not readily available on shipboard to allow this operation to be carried out in the vertical plane, which is the obvious and simple method. Therefore equipment had to be designed to perform this operation with the equaliser in the horizontal position. This created quite a problem as the bulkhead is of necessity a very close fit in the casing. Any external means for guiding the apparatus unit out of the casing would have to be aligned coaxially to the bulkhead to a very high degree of accuracy. This problem was aggravated by requiring that the equipment be portable for use on any ship.

The final solution provided withdrawal gear which has the following characteristics.

A. Alignment adjustments need not be made when erecting the equipment for operation.

B. Completely self-contained and does not require any special fitments to be provided on the ship.

C. Small enough when dismantled to pack into a travelling case approximately 60 by 18 by 14 inches (152 by 46 by 36 centimetres); this case also houses the ancillary equipment required for pressurising the petroleum-jelly system in the bulkhead of the pressure housing.

The principle of the design is to withdraw the bulkhead almost completely from its close-fitting bore in the casing whilst allowing the guiding to be done solely by the bore and not by any external guide rails. This is achieved by means of a short lead-screw attached to the face of the bulkhead and a spherical nut running in hemispherical cup washers that are free to move radially with respect to the lead-screw; the nut and washers are restrained in the carriage, which is supported on the two main members of the equipment. Rotation of the nut will withdraw the bulkhead, but the angular and radial position of the nut will be governed entirely by the position of the lead-screw; thus no radial thrust will be applied to the screw during the operation. When the bulkhead is all but fully withdrawn, the final few degrees rotation of the nut firmly clamps the lead-screw, nut, and hemispherical washers to the carriage, and so prevents any further relative movement among these members. The carriage may now be run along the guide rails formed by the main members to complete the withdrawal, the clearance

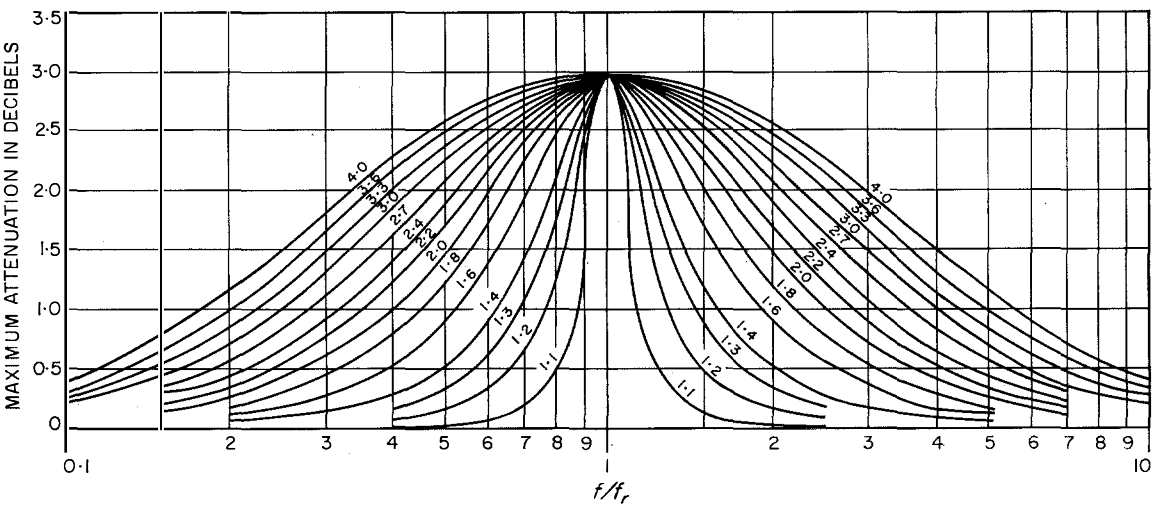


Figure 10—Normalised loss curves for a section of type-C over a range of $b = f_r/f_b$ from 1·1 to 4·0 for maximum attenuations at f_r of 3 and 6 decibels. The ordinate scale is doubled for the 6-decibel range.

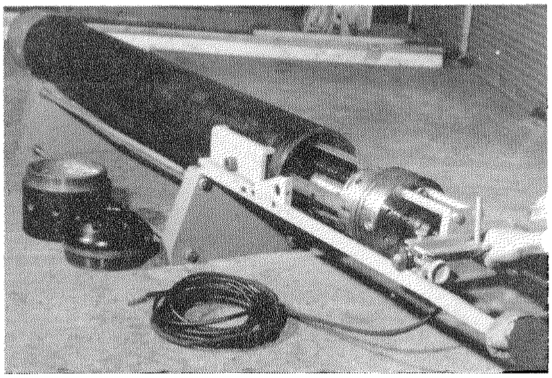


Figure 11—Withdrawing apparatus unit from the casing.

between the apparatus unit and the casing allowing for a fair degree of misalignment between the rails and the axis of the casing. The operation of completing the withdrawal by hand is illustrated in Figure 11.

For re-insertion of the apparatus unit and bulkhead into the casing, returning the carriage to its original position will just enter the bulkhead into its bore in the casing, and, after clamping the carriage to the brackets attached to the rails, rotation of the nut will drive the bulkhead fully home.

6. Conclusion

The adjustable submerged equaliser has been primarily developed for handling on board cableships, where it would be difficult and uneconomical to provide dairy conditions [12] for assembling and wiring submerged-repeater type of components. The adopted method is simple, fairly economical in the use of components, and provides truly adjustable equaliser characteristics within required limits.

Apart from the directional filters and 2 autotransformers in the power separating filter which are dependent on the system circuit capacity and, hence, on the system line frequencies, all other units are applicable to all projects within the frequency range of 20 kilocycles per second to 1200 kilocycles per second. This permits any unused inductance and capacitance units and cable simulator sections manufactured for a particular system to be available for another project.

The estimated time for building all of the possible 6 adjustable equaliser sections, together with extensive electrical and mechanical inspection and the subsequent housing operation, is 16 hours. With an average laying speed of 4 nautical miles per hour, this represents some 64 nautical miles of cable, or approximately $2\frac{1}{2}$ repeater sections for 18-circuit schemes, and 5 repeater sections in the case of 120-circuit schemes. Since equalisers are normally placed every 12th or 10th repeater, there is adequate time to determine the expected system levels at the end of the ocean block to be equalised and to design and complete the assembly and wiring of the submerged equaliser before it is due to be laid.

An equaliser described above has been successfully employed during the laying of the Scotice system [13] and demonstrated its full flexibility by providing some equalisation outside the transmission band to compensate for the excess gain at frequencies around 15 kilocycles per second. The equaliser was laid in 830 fathoms. The equaliser-handling gear operated perfectly, and the housing work, including pressurising the removable bulkhead, went smoothly.

Other units of the same design have been laid in depths of 1000 to 2000 fathoms on the *Med-1* and *Med-2* projects.

7. References

1. R. T. Halsey, "Global Communication," *Proceedings of the Institution of Electrical Engineers,* volume 109B, page 12; January 1962: Abstract 3816.E.

2. "Future Submarine Cable Plans Include Cable Ship," *Bell Laboratories Record,* volume 38, pages 390–392; October 1960.

3. "Transatlantic Telephone Cable," *Post Office Electrical Engineers Journal,* volume 49, part 4, pages 281–458, January 1957.

4. H. A. Lewis, J. M. Fraser, G. H. Lovell, and R. S. Tucker, "System Design for the North Atlantic Link," *Post Office Electrical Engineers Journal,* volume 49, part 4, pages 333–346; January 1957.

5. "Repeatered Submarine Cable Telephone Projects," C/2050, second edition, Standard Telephones and Cables, London.

6. "Anglo-Swedish Submarine Telephone Cable," *Post Office Electrical Engineers Journal,* volume 53, pages 165–166; October 1960.

7. F. Scowen, "Equalisation During Laying of the Anglo-Swedish Submarine Telephone Cable," *Post Office Electrical Engineers Journal,* volume 54, pages 48–50; April 1961.

8. "Cantat—A New Submarine Telephone Cable System to Canada," *Post Office Electrical Engineers Journal,* volume 54, pages 220–222; January 1962.

9. British Patent 742 837.

10. British Patent 847 858.

11. F. E. Terman, "Radio Engineers Handbook," pages 244–247; 1943: McGraw-Hill Book Company.

12. R. A. Brockbank, D. C. Walker, and V. G. Welsby, "Repeater Design for the Newfoundland–Nova Scotia Link," *Post Office Electrical Engineers Journal,* volume 49, part 4, pages 389–399; January 1957.

13. M. V. Young and W. J. Archibald, "United Kingdom–Faroes–Iceland (Scotice) Submarine-Cable Telephone System," *Electrical Communication,* volume 38, number 1, pages 76–87; 1963.

B. M. Dawidziuk was born in Bielsk-Podlaski, Poland, on 28 February 1926. In November 1941, he joined the Polish Forces in Russia as a volunteer and left the Soviet Union with the Polish Army in 1942 for two years in the Middle East and eighteen months in Italy. He arrived in the United Kingdom in 1946. He received an honours degree of B.Sc. (Eng.) in 1952 from the University of London.

On graduation he joined the transmission laboratories of Standard Telephones and Cables. After two years on development of the short-haul *N1* carrier system, he transferred to the submarine systems development group. Since then he has been closely associated with the design and development of submerged repeaters and with the engineering of submarine telephone cable projects, including participation in the laying of several cables in the North Sea and in the Mediterranean. He is now in charge of the submerged-repeater development section. Mr. Dawidziuk is an Associate Member of the Institution of Electrical Engineers.

F. L. Jarvis was born in Kent, England, in 1913. He was educated at the Gravesend Technical School and later took further courses at the Leicester College of Technology.

From 1929 to 1935, he was with W. T. Henley's Telegraph Works, where he was employed as a draughtsman in the research laboratories.

Joining Standard Telephones and Cables in the latter part of 1935, he was responsible for the mechanical design of transmission laboratory and factory test gear throughout the war years until 1945. In the early post-war period, he was engaged on the mechanical design of waveguide components for microwave systems. Since 1949, he has been actively engaged on the mechanical design of all aspects of submerged repeaters and submerged equalisers.

Variable-Reactance Frequency Multipliers

W. JANEFF

ITT Federal Laboratories, A Division of International Telephone and Telegraph Corporation; Nutley, New Jersey

1. Introduction

Variable-reactance frequency multipliers using high-power transistors provide a reliable and efficient solid-state frequency source for the microwave range. Their properties and applications are extensively discussed in the literature.

Interesting results have been obtained from development work on variable-reactance or varactor frequency multipliers for the microwave range as part of a tropospheric scatter program. Some comparisons are made with nonlinear resistances. The advantages of variable reactances as well as their theoretical limitations are pointed out. Also, certain points are considered that may be of interest to the designer who often must compromise among efficiency, output power, and circuit complexity, especially in cases where frequency-multiplier chains tunable over wide frequency ranges are required. An X-band frequency multiplier is discussed briefly.

The theoretical discussion presented in the appendix is restricted to the frequency doubler, the simplest and most efficient varactor frequency multiplier.

2. General Properties of Nonlinear Devices

One property of linear time-invariant elements is that no frequency conversion can take place in them. They are generally classified as "pure lossy" when they dissipate power, or as "lossless" when they do not consume power. They are purely inductive or purely capacitive when they draw reactive power from and return it to the energy source.

These definitions do not apply to nonlinear or time-varying elements. A nonlinear resistance or a nonlinear capacitance, although free from dissipative losses and characterized by a single-valued charge-voltage equation, accepts real power and represents to the prime source a power-consuming device; this same power, of course, can not be dissipated in the lossless element, and it is released at other frequencies. Nevertheless, the nonlinear element, although lossless, appears as a power-consuming device to the prime source. Since no power is retained, the over-all power input to the device equals zero.

2.1 Nonlinear Resistance

When a nonlinear resistance is energized by a direct-current source and an alternating-current source in series, the following observations can be made regarding direct current and currents at fundamental and harmonic frequencies.

(A) At short-circuit conditions $R_{load} = 0$, the highest-order harmonic generated by the nonlinear resistance depends only on its degree of nonlinearity. A square-law characteristic, for instance, will give rise to a second harmonic, all higher-order harmonics being equal to zero.

(B) The same nonlinear resistor can give rise to any desired harmonic if external circuits are used to reapply additional voltages or currents at frequency multiples to the nonlinear device. This can be seen physically by the fact that under these conditions a highly distorted voltage (or current) is applied to the nonlinear device. These observations apply to nonlinear resistances as well as to nonlinear reactances.

The general power relations for resistive harmonic generators are governed by the following equations set forth by C. H. Page [7].

$$P_n \leq \frac{1}{n^2} P_1 \tag{1}$$

$$P_0 \geq (n^2 - 1) P_n \tag{2}$$

where P_0, P_1, and P_n are direct-current power, power at the fundamental frequency, and power at the nth harmonic, respectively; n is the order of the generated harmonic.

Page's equations state that:

(A) In any conversion process with nonlinear resistances, direct-current power is generated;

(B) Even with lossless conversion, where all fundamental power is converted and all undesirable harmonics are absent, the efficiency in generating the nth harmonic can not exceed $1/n^2$;

(C) The rest of the power at the fundamental frequency is converted into direct current.

In the case of a lossless resistive doubler, for instance, the maximum possible efficiency would be 25 percent; the rest of the input power (75 percent) would be converted into direct current. In the case of a tripler, the best efficiency would not exceed 11 percent.

2.2 Nonlinear Reactances

The nonlinear reactance as a harmonic generator differs in several significant respects from the nonlinear resistance.

(A) It does not rectify; that is, a sinusoidal voltage applied to the reactance gives rise only to different harmonic currents, with amplitudes depending on varactor characteristic and operational conditions. The biasing direct voltage created by the varactor when operated without external bias is due to the nonlinear resistance that is part of the P-N junction.

(B) The efficiency of a lossless reactance, operated as a frequency multiplier, can reach the theoretical limit of 100 percent. All power accepted at the fundamental frequency is released at one or several harmonics. This is confirmed by the Manley and Rowe equations [8], which in the case of a frequency multiplier assume the simple form:

$$P_{1,0} = \sum_{m=2}^{\infty} - P_{m,0}. \qquad (3)$$

In case the circuit configuration favors only one harmonic frequency, the full input power appears at that particular frequency, independent of the reactance characteristic, circuit, and load conditions. The load impedances for all undesired harmonics, however, must be made zero, infinite, or purely reactive.

(C) The fact that the varactor accepts real power does not mean that it has an input impedance in the common sense of the word. It has, however, a complex ratio of input voltage to input current termed conversion impedance that does not exist physically; rather, it describes the ability of the varactor to accept real power at the fundamental frequency and is largely a function of the chosen operating conditions.

The same remarks apply to the output impedance of the varactor. For these reasons, the performance of a varactor cannot be represented by an equivalent diagram. As will be seen, certain precautions can be taken to ensure maximum power output or best efficiency. However, these can hardly be described as matching procedures, in the common sense of the word.

3. *P-N* Junction

The P-N junction exhibits two important properties that lead to basically new applications: the sharp-reverse-breakdown effect and the electrically controllable capacitance.

Physically, the diode can be represented by the equivalent circuit shown in Figure 1. Here, $C_{(v)}$ is the voltage-sensitive junction capacitance; $R_{(v)}$ is the voltage-sensitive diode resistance; R_S is the junction series resistance or spreading resistance that is primarily responsible for the losses in the varactor. C_1, R_1,

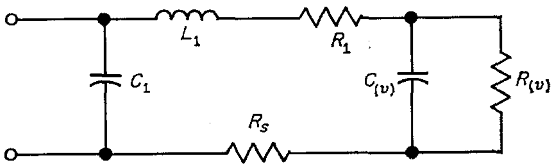
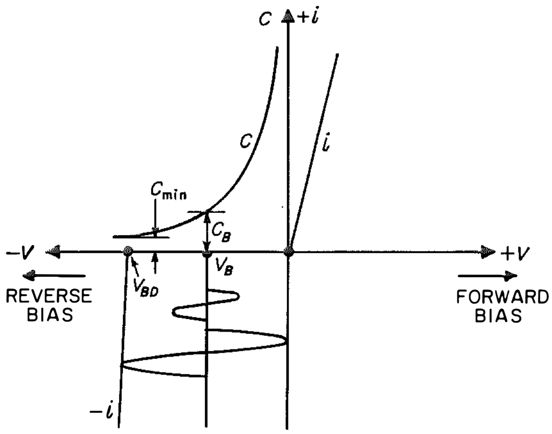


Figure 1—Equivalent diagram of a P-N junction.

Figure 2—Characteristics of a varactor.



and L_1 are case capacitance, losses, and lead inductance.

While the case components C_1, R_1, L_1 are not without influence on the proper operation for the varactor at microwave frequencies, the truly important parameters are the series resistance R_S and the capacitance-versus-voltage sensitivity of the junction capacitance $C_{(v)}$.

The cutoff frequency f_{co} is defined as the frequency at which the absolute value of the capacitive reactance of $C_{(v)}$ is equal to the series resistance R_S. In this case, the static value of $C_{(v)}$ at breakdown voltage is taken.

$$f_{co} = \frac{1}{2\pi C_{min} R_S}. \qquad (4)$$

The Q factor of the junction represents the ratio of the capacitive impedance of C_{min} and the series resistance R_S.

$$Q = \frac{1}{2\pi f C_{min} R_S}. \qquad (5)$$

Cutoff frequency or Q factor, together with the capacitance-voltage relationship, suffice to define a varactor for most microwave applications. The Q factor, which becomes equal to unity when the varactor is operated at cutoff frequency, is a complete measure of the efficiency of the diode element. Only high-Q diodes can ensure efficient operation at high frequencies, which means that the product fR_SC_{min} must be minimized. To ensure this is the most important problem of diode manufacture.

The static characteristics of the varactor diode are depicted in Figure 2. The capacitance-versus-voltage characteristic obeys the law $C_{(v)} = KV^{-\theta}$, θ being equal to 1/2 for the abrupt junction, and 1/3 for the graded junction. The steepness and the absolute value of the capacitance variation depends mainly on the background doping level of the P-N junction. Recent investigations [6] have shown that higher capacitance-voltage sensitivity can be achieved by using impurity profiles that have an abrupt transition from P to N at the junction, but a decreasing impurity concentration when moving away from the junction. Higher capacitance-voltage sensitivity means a greater coefficient θ; this is favorable for the application of varactors as frequency multipliers.

To operate the varactor in the purely capacitive mode, the instantaneous voltage applied to the P-N junction should remain within the limits set by V_{BD}, the breakdown voltage, and zero. For full exploration of the capacitance-voltage curve, both the bias voltage V_B and the amplitude of the alternating voltage should be chosen equal to $V_{BD}/2$ (voltage limitation). For simplicity of associated equipment, automatic bias has been adopted throughout all experimental work.

A second (thermal) limitation is set by the losses in the spreading resistance R_S, which may not exceed a safe value determined by the heat transfer properties of the varactor.

Should the losses in R_S grow beyond this point, the internal contact potential of the P-N junction will be affected. The contact potential varies from about 0.7 volt at 25 degrees to about 0.4 volt at 150 degrees centigrade. A change in the contact potential will result in a change of the capacitance-voltage characteristic of the varactor, which in turn will affect the tuning and cause a drop in output power and efficiency.

4. Varactor

4.1 General

The following is a study of the simplest configuration of varactor and power source. The varactor is assumed lossless; the power source has zero internal impedance and maintains a purely sinusoidal voltage across the varactor at the fundamental frequency. No resistances, filters, or reactances are present in the current path. No resonance effects take place.

The incremental capacitance is given by

$$C_{(v)} = \frac{dq}{dV} = KV^{-\theta}. \tag{6}$$

The incremental charge is

$$q = K\int V^{-\theta}dV = \frac{K}{1-\theta}V^{1-\theta} \tag{7}$$

where the integration constant is set equal to zero.

Assuming that the voltage applied to the varactor is

$$V = V_B + V\sin\omega t$$

the charge on the nonlinear capacitor will be

$$q = \frac{K}{1-\theta}(V_B + V\sin\omega t)^{1-\theta} \tag{8}$$

where V_B is the bias voltage and V is the amplitude of the sinusoidal voltage at the fundamental frequency.

The current through the varactor is given by

$$i = \frac{dq}{dt} = \frac{K\omega V\cos\omega t}{(V_B + V\sin\omega t)^{\theta}}. \tag{9}$$

The constant K can be determined from the boundary conditions. At breakdown voltage V_{BD} the varactor capacitance assumes its minimum value C_{min}, hence

$$K = C_{min}V_{BD}{}^{\theta}$$

and the general expressions for capacitance, charge, and current are

$$C = C_{min}V_{BD}{}^{\theta}(V_B + V\sin\omega t)^{-\theta} \tag{10}$$

$$q = \frac{C_{min}}{1-\theta}V_{BD}{}^{\theta}(V_B + V\sin\omega t)^{1-\theta} \tag{11}$$

$$i = \frac{C_{min}V_{BD}{}^{\theta}\omega V\cos\omega t}{(V_B + V\sin\omega t)^{\theta}}. \tag{12}$$

Setting $\theta = \frac{1}{2}$ for the abrupt P-N junction and $V = \xi V_B(V_B = V_{BD}/2)$, where ξ for operation in the purely capacitive mode approaches but never reaches unity, (10), (11), and (12) appear in the following form

$$V = V_B(1 + \xi\sin\omega t)$$

$$C = \frac{2^{1/2}C_{min}}{(1 + \xi\sin\omega t)^{1/2}} \tag{13}$$

$$q = 2^{1/2}C_{min}V_{BD}(1 + \xi\sin\omega t)^{1/2} \tag{14}$$

$$i = \frac{\xi\omega C_{min}V_{BD}}{2^{1/2}}\cdot\frac{\cos\omega t}{(1 + \xi\sin\omega t)^{1/2}} \tag{15}$$

where C, q, and i are expressed in terms of breakdown voltage and minimum capacitance, whereas the driving voltage is expressed in terms of the bias voltage.

These equations permit an insight into the mechanism of operation of the varactor.

With the driving voltage varying between approximately zero and V_{BD}, the charge on the varactor oscillates, without changing sign between a constant minimum value q_{min} (at $\omega t = 3\pi/2 + n\cdot 2\pi$) and maximum value $q_{max} \cong 2\cdot C_{min}V_{BD}$, which occurs at $\omega t = \pi/2 + n\cdot 2\pi$. The capacitance varies between C_{min} (at $\omega t = (\pi/2) + n\cdot 2\pi$), and C_{max}, which is reached at $\omega t = (3\pi/2) + n\cdot 2\pi$, and the absolute value of which depends on ξ. For $\xi = 0.95$, C_{max} is equal to $6.3C_{min}$. Thus, minimum charge value coincides with maximum capacitance value. Both occur when the voltage across the varactor is a minimum.

The current waveform closely resembles a saw tooth with zero values at $\omega t = \pi/2$ and $\omega t = 3\pi/2$. As expected, the current is symmetrical with respect to the time axis (that is, no rectifying effect takes place) and exhibits a steep transition from negative to positive values at the point $\omega t = 3\pi/2$. Further treatment of (15) reveals that both positive and negative current maxima are situated symmetrically with respect to the point $3\pi/2$. The exact phase angle at which current maximum occurs is a function of ξ and is given by

$$(\omega t)_{max} = \sin^{-1} \frac{(1 - \xi^2)^{1/2} - 1}{\xi} \qquad (16)$$

for $\xi = 0.9$, for instance, $(\omega t)_{max} = \sin^{-1}(-0.63) = -39$ degrees, and 180 degrees + 39 degrees; for $\xi = 0.95$ the phase angle is $(\omega t)_{max} = -46$ degrees and 180 + 46 degrees. Thus, with ξ approaching unity, the maximum points approach the zero transition point from both sides, which results in an increase in steepness of the saw-tooth curve.

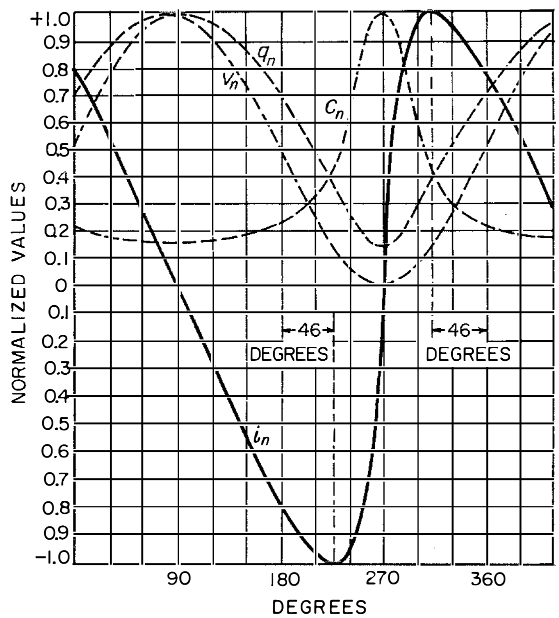


Figure 3—Normalized values of voltage, capacitance, charge, and current for $\xi = 0.95$.

Figure 3 shows the normalized values of these quantities for $\xi = 0.95$.

$$V_n = \tfrac{1}{2}(1 + \xi \sin \omega t)$$

$$C_n \cong 0.16(1 + \xi \sin \omega t)^{-1/2}$$

$$q \cong 0.71(1 + \xi \sin \omega t)^{1/2}$$

$$i \cong 1.25 \frac{\cos \omega t}{(1 + \xi \sin \omega t)^{1/2}}.$$

4.2 Circuit Analysis of a Series Doubler

This analysis is based on the following assumptions.

(A) Purely sinusoidal voltages are applied to the varactor at frequencies ω and 2ω. The harmonics appear in the charge q and the current i.

(B) Ideal conditions prevail, that is, complete harmonic separation, no power flow at any undesired harmonic, and no losses either in tuned circuits or in the varactor.

Based on the Taylor theorem, the alternating-current charge on the varactor can be expressed as a function of the applied voltage in the following manner.

$$q = \sum_{K=1}^{\infty} \alpha_K V^K \qquad (17)$$

where α_K is the Taylor coefficient of the Kth term of the Taylor expansion, and V is the applied voltage across the varactor.

In the general case, the applied voltage as well as the resulting charge can be expressed as a sum of sinusoidal voltages.

$$\sum_{\substack{n=-\infty \\ n\neq 0}}^{n=+\infty} Q_n \exp[jn\omega t] = \sum_{K=1}^{\infty} \alpha_K \left\{ \sum_{\substack{n=-\infty \\ n\neq 0}}^{n=+\infty} V_n \exp[jn\omega t] \right\}^K \qquad (18)$$

where V_n, V_{-n} and Q_n, Q_{-n} are complex conjugate vectors rotating in opposite directions at an angular velocity equal to $n\omega$.

This equation is general. Physically, it signifies that a charge component Q of frequency

$l\omega$ is built up by contributions of all those voltage components whose products yield a voltage component of frequency $l\omega$. An infinite number of such components exist, even in the case where the number of voltage components across the varactor is restricted to 2. Two restrictions reduce (18) to a form that, although less general, describes the voltage and charge relations in the series doubler with sufficient accuracy.

(A) The voltage across the varactor consists of components at the fundamental frequency and at only the second harmonic.

(B) In the case of a frequency doubler, the most significant contribution to the second-harmonic charge component is due to the square term of the Taylor series.

(C) Only current components at frequencies ω and 2ω will be selected since the external circuit configuration will assign zero-impedance paths to all higher-order currents.

The resulting equation is dealt with in Appendix 7.1. It leads to the following expression of the conversion conductance

$$G_c = \frac{\omega^2 C_B{}^2}{G_L} \cdot \frac{\theta^2}{4} \cdot \left(\frac{V_{in}}{V_B} \right)^2 \qquad (19)$$

where

C_B = capacitance at the bias point
θ = capacitance nonlinearity coefficient
G_L = real part of the load admittance
V_{in} = amplitude of the input voltage, and
V_B = bias voltage.

Or, the conversion resistance that appears across the input terminals of the varactor

$$R_c = \frac{1}{G_c} = \frac{1}{\omega^2 C_B{}^2 R_L} \cdot \frac{4}{\theta^2} \cdot \left(\frac{V_B}{V_{in}} \right)^2 \qquad (20)$$

where $R_L = 1/G_L$ is the load resistance, and $C_B = 2^{1/2} C_{min}$.

Equation (19) and the general equations for efficiency and output power derived in Appendix 7.2 allow an insight into the significant parameters that ensure efficient operation.

It can be seen from (39) that the efficiency is a maximum when both load conductance and conversion conductance are large compared to the loss conductances in secondary and primary circuits, respectively. In this case, output power at frequency 2ω will equal input power at frequency ω, and the efficiency will approach 100 percent.

While the choice of a reasonably large load conductance should not present major difficulties, the attainment of a large conversion conductance is not always possible. Examination of (19) reveals the following ways to increase G_c.

(A) By increasing the nonlinearity coefficient θ, a function of varactor physics and manufacturing.

(B) Given θ and selecting G_L from secondary-circuit considerations (mainly $G_{S2} \langle G_L \rangle G_{T2}$), the only two parameters left are the ratio V_{in}/V_B, and the varactor conductance ωC_B at the bias point.

(C) V_{in}/V_B shall not exceed unity if the varactor is to be operated in the purely capacitive mode. This requirement is met by automatic bias operation.

(D) Finally, ωC_B cannot be increased beyond a certain point without affecting the loss conductance of the varactor. This can be seen from the following.

In a first approximation, the varactor spreading resistance R_S can be represented in shunt configuration as an equivalent loss conductance that must be compared to G_L and G_c in the secondary circuit and in the primary circuit, respectively. From

$$G_L > G_{S2}$$

or

$$G_L > (2\omega)^2 C_B{}^2 R_S \qquad (21)$$

it can be seen that it becomes increasingly difficult, with increasing frequency, to keep the load conductance larger than the equivalent loss conductance of the varactor in the secondary circuit. The use of varactors with low capacitances at the bias point can alleviate this difficulty only to a certain point. Small varactor capacitances require careful circuit design to reduce the effect of stray capacitances that reduce the effective nonlinearity of the device and consequently degrade its performance. The same considerations, applied to the primary circuit, yield

$$G_c > G_{S1}$$
$$G_{S1} = (\omega C_B)^2 R_S$$

or

$$G_L < \frac{\theta^2}{4} \cdot \frac{1}{R_S}. \tag{22}$$

Equations (21) and (22) define a rather small margin of G_L values within which efficient varactor operation can be obtained.

$$(2\omega)^2 C_B{}^2 R_S < G_L < \frac{\theta^2}{4} \cdot \frac{1}{R_S}. \tag{23}$$

Assuming that the losses in the tank circuits are at least one order of magnitude lower than the losses in the varactor, (39) can be written

$$\eta = \frac{G_c}{G_c + G_{S1}} \cdot \frac{G_L}{G_L + G_{S2}}. \tag{24}$$

Examination of (24) reveals that a maximum of η can be obtained by selecting $G_L = \omega C_B \theta$. Introducing $\theta = \frac{1}{2}$ for the abrupt junction, and the cutoff frequency from (4), the efficiency can be written as

$$\eta = \left[\frac{1}{1 + 11.2(f/f_{co})} \right]^2.$$

This equation is plotted in Figure 4 with f_{co} as a parameter and f as the independent variable. It can be seen that efficiencies above 50 percent can be obtained only when the operating frequency is between one and two orders of magnitude below the cutoff frequency.

5. Tuning

The voltage-dependent capacitance of the varactor can give rise to some difficulties when the varactor is operated with automatic bias. This can be seen from the following: Figure 5 represents the radio-frequency voltage V_{RF} across the varactor (curve a) as a function of the bias voltage V_B. This is the familiar response curve of a tuned circuit, symmetrical in the vicinity of its peak point, but otherwise departing from the vertical axis of symmetry. In the same figure, the diode effect is represented by a straight line defining the relationship between radio-frequency voltage and bias voltage. The dependence of the varactor capacitance on bias voltage, C versus V_B, is also shown.

Curve a shows three points of intersection with the straight line. Of these, only points A and B are stable, whereas C is a point of unstable equilibrium. In the region where the radio-frequency response curve shows a greater radio-frequency voltage than the self-bias line, the operating point will shift toward higher bias until a new point of intersection is reached; and conversely, where the radio-frequency response curve shows values lower than the self-bias line, the operating point will shift toward lower bias values until another stable point of intersection is reached. Only at points A and B will radio-frequency voltage and varactor capacitance be in stable equilibrium. Any perturbation about these points will be encountered by a restoring effect.

A normal tuning process consists in gradually approaching the peak radio-frequency voltage across the varactor by varying one reactive component in the circuit. The peak voltage must necessarily lie on both the radio-frequency response curve and the self-bias line and will produce a bias voltage equal to one-half the breakdown voltage of the varactor. In case the tuning process is started at or below point A on curve a with a large inductance, point C is gradually approached by decreasing the inductance, and thus increasing radio-frequency and bias voltage. Increasing the bias voltage, however, reduces the varactor capacitance so that the tuning process is accelerated by a varactor capacitance variation in the right direction. An additional self-tuning takes place that depends on the diode effect, the Q of the radio-frequency response curve, and the varactor capacitance-bias-voltage characteristic; it can be described as a regeneration process where effect and cause are interrelated and sustain each other. This effect is insignificant so long as tuning takes place far from the resonant peak. However, as tuning progresses beyond point C, it increases rapidly and culminates in a sudden jump that indicates that the stable point B, which lies beyond the maximum of the radio-frequency response curve, is reached. This point will, however, rarely correspond to the desired operating condition. The self-bias produced may be smaller or greater than the required value, and this will result in insufficient power output or overload. Consequently, subsequent adjustment of the applied radio-frequency voltage by means of a variable capacitor in series with the varactor, for instance, and retuning of the tank circuit will be needed to attain a stable operating point with the desired bias voltage V_B. This is shown on curve b.

The same line of reasoning shows that no regenerative effect will take place between points A and C, if resonance is approached by a gradual increase of a variable inductance. The resulting decrease of the capacitance of the varactor will counteract the tuning process. The system will be stabilized by what may be called an inherent negative feedback. Tuning beyond point C, however, will result in the same effect as in the first case described.

The following equations can be used to describe the tuning process.

The total incremental increase in radio-frequency voltage across the tuned circuit is

$$\Delta V_{RF} = \Delta V_{RF}' + \Delta V_{RF}''$$
$$= \frac{dV_{RF}}{dL}\Delta L + \frac{dV_{RF}}{dC}\Delta C \quad (25)$$

where L and C are tank inductance and varactor capacitance, respectively. $\Delta V_{RF}''$ is

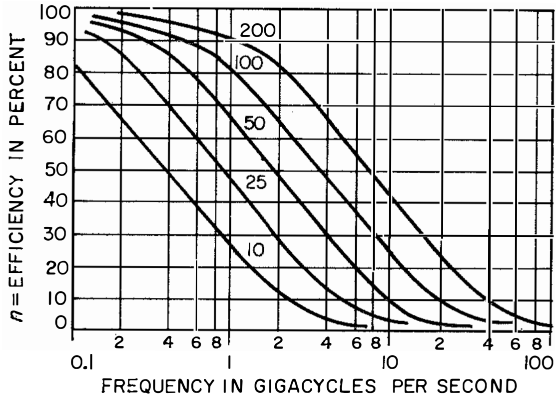


Figure 4—Efficiency of a series doubler as a function of frequency for the indicated values of cutoff frequency in gigacycles per second.

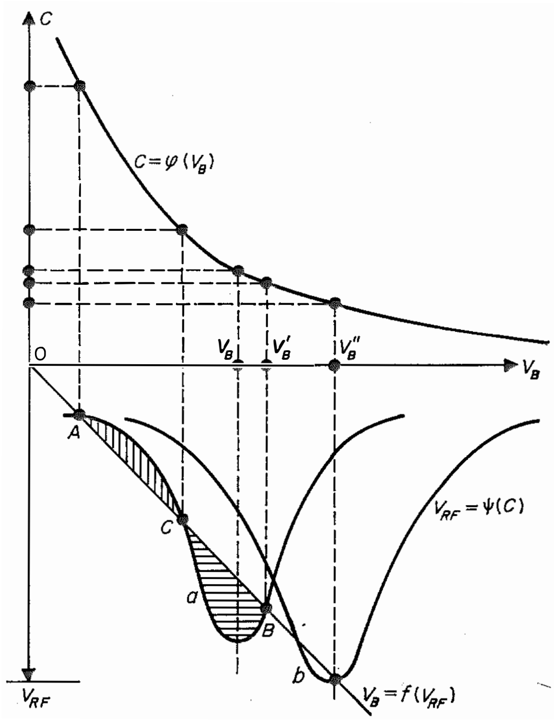


Figure 5—Tuning characteristics of a varactor.

caused by $\Delta V_{RF}'$ and can be stated as follows.

$$\Delta V_{RF}'' = \frac{dV_{RF}}{dC} \cdot \frac{dC}{dV_B} \cdot \frac{dV_B}{dV_{RF}} \Delta V_{RF}'. \quad (26)$$

Sign and absolute value of the right-hand side of this equation depend on the derivatives of three curves in Figure 5, that is,

$$V_{RF} = \psi(C)$$
$$C = \varphi(V_B)$$
$$V_B = \phi(V_{RF}).$$

A positive sign will indicate that the inductance tuning is sustained by the capacitive effect.

Substituting from

$$V_{RF} = \frac{1}{\omega C} \cdot \frac{E}{\left[R^2 + \left(\omega L - \frac{1}{\omega C}\right)^2\right]^{1/2}} \quad (27)$$

with

$$Q = \frac{1}{\omega_r CR}$$
$$1 = \omega^2 L C_r$$
$$V_{RF} = EQ$$

(27) yields

$$V_{RF} \cong \frac{V_{RF}}{\left[1 + Q^2\left(\frac{C - C_r}{C_r}\right)^2\right]^{1/2}}$$

$$\frac{dV_{RF}}{dC} = \frac{-Q^2(C - C_r)V_{RF\,max}}{C_r^2\left[1 + \frac{Q^2}{C_r^2}(C - C_r)^2\right]^{3/2}}$$

$$\frac{dV_B}{dV_{RF}} = \xi$$

$$\frac{dC}{dV_B} = -\theta \cdot K \cdot V_B^{-1-\theta}.$$

Hence,

$$\Delta V_{RF}'' = \frac{Q^2(C - C_r)\cdot\theta\cdot K V_B^{-1-\theta}\cdot\xi}{C_r^2\left[1 + \frac{Q^2}{C_r^2}(C - C_r)^2\right]^{3/2}} \Delta V_{RF}'. \quad (28)$$

The sign of this equation is negative, since resonance is always approached from $C > C_r$. Hence, the tuning process is sustained when dV_{RF}/dL is negative also, that is, when decreasing inductance leads to resonance.

Equation (28) contains other information. The voltage increase will give rise, by virtue of the mechanism described, to an additional increase of radio-frequency voltage, $\Delta V_{RF}'''$, and this process will maintain itself and progress toward a stable point, if

$$\frac{|\Delta V_{RF}'''|}{|\Delta V_{RF}''|} > 1. \quad (29)$$

Evaluating (28), first with $\Delta V_{RF}'$ at the corresponding point of operation, then with $\Delta V_{RF}''$ at the new operating point, will reveal whether (29), and hence the condition for regeneration, is satisfied.

6. Circuit Complexity

To obtain satisfactory harmonic separation in a frequency doubler, the following tuning controls must be provided.

(A) One variable inductance in each circuit to tune out the reactance of the varactor at the fundamental frequency and at the desired harmonic.

(B) One trap in each circuit to provide isolation from the undesired harmonic voltages or currents.

(C) In case a large frequency range must be covered, a matching device in both the primary and the secondary circuits.

This amounts to six tuning controls. Omitting any one of these tuning controls will degrade the operation. In particular, omitting the traps will not allow complete current separation. Power at the fundamental frequency will be dissipated in the load resistance, and power at the generated harmonic will cause losses in the internal resistance of the power source. In addition, in the case of a series configuration, the varactor terminals will not appear short-circuited for all other harmonics. Consequently, current flow at these frequencies through the load and the internal resistance of the generator will increase the losses. Omitting the tuning inductances, on the other hand, will not allow the two circuits to resonate properly. This will affect the input conductance, degrade the efficiency, and create abnormal matching conditions. Finally, omitting the matching controls will confine efficient operation and adequate power output to a restricted frequency range.

In microwave exciters, large multiplication coefficients are required. Assuming operations in the X band, for instance, and a prime frequency source of 50 to 60 megacycles per second followed by transistor stages that bring the frequency up to 150 to 200 megacycles per second, a multiplication factor of 64 is needed. This amounts to a chain of 6 doublers, which in turn sets the number of tuning controls equal to 36. The obvious way to reduce this number of controls is the use of higher-order frequency multipliers. Two quadruplers and one doubler, for instance, would yield a multiplication factor of 32, and 18 controls would be required instead of 36. The quadrupler in a simple circuit configuration, however, consisting of only two tuned circuits, would operate at lower efficiency; in many applications this is preferable to an excessive number of tuning controls. Introducing idler circuits to increase efficiency is, from this point of view, also a poor choice. In fact, a quadrupler with an idler circuit tuned to the second harmonic is equivalent to two doublers making use of the same varactor diode, and almost as complex as two separate doublers.

Figure 6 shows a frequency multiplier in which efficiency has been sacrified to reduce the number of tuning controls. Input frequency is in the range of 443.75 to 531.25 megacycles per second, and the output frequency is 7100 to 8500 megacycles per second. The multiplier makes use of two quadruplers in a series configuration; it consists of three tunable coaxial cavities and a piece of waveguide with an adjustable short. Cavity 1 receives the input signal through loop coupling; it is tunable from 443.75 to 531.25 megacycles per second by movable short (1). A varactor crystal (*MA-4345C*) is mounted at the end of the inner conductor, which reaches through an opening into the second cavity and thus establishes a capacitive link to the second cavity. The coupling is controlled by micrometer (2). Cavity 2 is tunable over the range 1775 to 2125 megacycles per second (control 3). The second and third cavities are connected by aperture coupling. The center conductor of the third cavity bears the second varactor (*MA-4343A*), which reaches (control 4) into the piece of waveguide *RG-51/U* with a movable short that is tunable from 7100 to 8500 megacycles per second, but which cannot be seen in the photograph. The third cavity is tunable by a

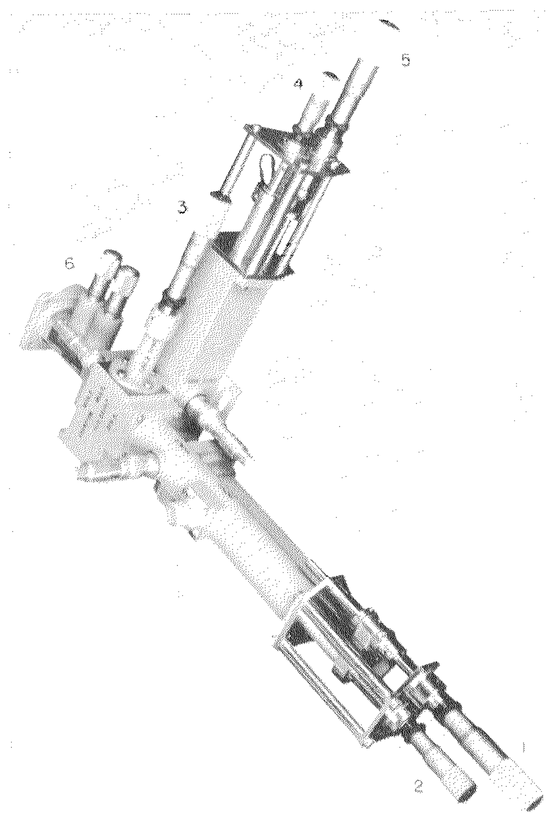


Figure 6—Frequency multiplier from about 500 to 8000 megacycles per second. There are 6 tuning controls, 5 of which may be seen. Controls (6) are for the band-pass filter in the output waveguide.

moving short (control 5). A two-cavity band-pass filter ensures additional attenuation of the undesired harmonics and is adjusted by controls (6).

The whole assembly has 6 tuning controls excluding the band-pass filter. Its dimensions are 4.5 inches (11.4 centimeters) by 6 inches (15.2 centimeters) by 20 inches (50.8 centimeters). The electrical performance is given in Figure 7.

7. Appendix

7.1 Series Doubler

$$\sum_{u=-1,-2}^{+1,+2} Q_n \exp[jn\omega t] = \alpha_1 \sum_{u=-1,-2}^{+1,+2} V_n \exp[jn\omega t] + \alpha_2 \{ \sum_{u=-1,-2}^{+1,+2} V_n \exp[jn\omega t]\}^2. \quad (30)$$

Separating fundamental and second-harmonic terms

$$Q_1 \exp[j\omega t] + Q_{-1} \exp[-j\omega t] = (\alpha_1 V_1 + 2\alpha_2 V_{-1} V_2) \exp[j\omega t] + (\alpha_1 V_{-1} + 2\alpha_2 V_1 V_{-2}) \exp[-j\omega t]$$

$$Q_2 \exp[j2\omega t] + Q_{-2} \exp[-j2\omega t] = (\alpha_1 V_2 + \alpha_2 V_1^2) \exp[j2\omega t] + (\alpha_1 V_{-1} + \alpha_2 V_{-1}^2) \exp[-j2\omega t].$$

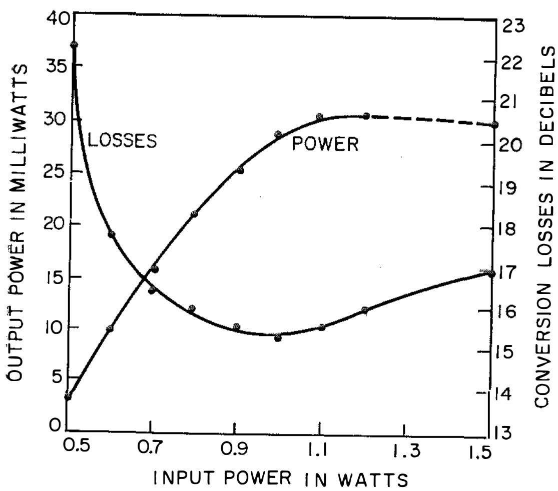


Figure 7—Output power and efficiency of a frequency multiplier operating at 7.8 gigacycles per second.

Fundamental and second-harmonic current components are obtained by differentiating the Q vectors with respect to time. Letting V_1 be V_{-1}, which amounts to defining the time origin, and eliminating the time dependence in the preceeding equations, the following relationships among complex quantities are obtained.

$$\begin{aligned} I_1 &= j\omega(\alpha_1 V_1 + 2\alpha_2 V_1 V_2) \\ I_{-1} &= -j\omega(\alpha_1 V_1 + 2\alpha_2 V_1 V_{-2}) \end{aligned} \quad (31)$$

$$\begin{aligned} I_2 &= j2\omega(\alpha_1 V_2 + \alpha_2 V_1^2) \\ I_{-2} &= -j2\omega(\alpha_1 V_{-2} + \alpha_2 V_1^2). \end{aligned} \quad (32)$$

Introducing the load constraint

$$-I_2 = V_2 |Y_L| \exp[j\varphi] \quad (33)$$

where $Y_L \exp[j\varphi]$ is the load admittance, into (32) yields

$$I_2 = \frac{j2\omega\alpha_2 V_1^2 |Y_L| \exp[j\varphi]}{|Y_L| \exp[j\varphi] + j2\omega\alpha_1}. \quad (34)$$

Resonating the secondary circuit, that is

$$\mathcal{I}m\{|Y_L| \exp[j\varphi]\} = -j2\omega\alpha_1$$

annuls the imaginary part in the denominator and the secondary current appears in the following form.

$$I_2 = \frac{j2\omega\alpha_2 V_1^2 |Y_L| \exp[j\varphi]}{\mathcal{R}e\{|Y_L| \exp[j\varphi]\}}. \quad (35)$$

Substituting (33) and (35) into (31) leads to

$$I_1 = j\omega\alpha_1 V_1 + \frac{4\omega^2\alpha_2^2 V_1^3}{\mathcal{R}e\{|Y_L| \exp[j\varphi]\}}. \quad (36)$$

Then, the input or conversion conductance Y_c is

$$Y_c = \frac{I_1}{V_1} = j\omega\alpha_1 + \frac{4\omega^2\alpha_2^2 V_1^2}{\mathcal{R}e\{|Y_L| \exp[j\varphi]\}}. \quad (37)$$

Resonating the primary circuit at the fundamental frequency will cancel the imaginary part of Y_c; introducing G_L for the real part of the load admittance, (37) can be written.

$$G_c = \frac{4\omega^2\alpha_2^2 V_1^2}{G_L}. \quad (38)$$

One last step remains to be done: the computation of the Taylor series coefficients α_1 and α_2; α_2 will determine the numerical value of G_c, whereas α_1 determines the admittance that will have to be resonated in the primary and secondary circuits.

The Taylor expansion of the charge q as a function of the applied voltage can be written

$$q(V) = q(V_B) + \alpha_1(V - V_B) + \alpha_2(V - V_B)^2 + \cdots$$

where

$$\alpha_1 = \frac{1}{1!}\frac{dq_{(V)}}{dV}\bigg|_{V=V_B}$$

$$\alpha_2 = \frac{1}{2!}\frac{d^2q_{(V)}}{dV^2}\bigg|_{V=V_B}.$$

Making use of (7)

$$\alpha_1 = C_B$$
$$\alpha_2 = -\tfrac{1}{2}\cdot\theta\cdot C_B\cdot V_B^{-1}.$$

With this expression for α_2 and since the driving or input voltage $V_{in} = 2V_1$, (38) appears in the following final form

$$G_c = \frac{\omega^2 C_B^2}{G_L}\cdot\frac{\theta^2}{4}\cdot\left(\frac{V_{in}}{V_B}\right)^2. \quad (19)$$

7.2 Efficiency and Power Output

An elementary circuit analysis of a varactor multiplier shows the effects of varactor losses and of the external circuit properties on the efficiency of the chosen assembly.

The following assumptions are made.

(A) The varactor is resonated at the fundamental frequency and at the desired harmonic. Power flow is restricted to only these two frequencies.

(B) The fundamental and harmonic currents are confined to strictly determined paths, and losses in the respective circuits are due to currents at only these two frequencies.

Under these conditions the multiplier circuit can be split into two separate circuits as in Figure 8. The first circuit operates at the fundamental frequency and contains the power source with its internal conductance G_1, the loss conductance of the tuned circuit G_{T1}, the loss conductance of the varactor diode at the fundamental frequency G_{S1}, and the "conversion conductance" G_c, which is the real part of the input admittance and constitutes the only useful element in the primary circuit.

The secondary circuit operates at the chosen harmonic frequency and contains the load conductance G_L, the loss conductance of the tank circuit G_{T2}, and the loss conductance of the varactor diode at the chosen harmonic G_{S2}.

7.2.1 *Primary Circuit*

Input power

$$P_1 = \frac{I_1^2}{G_{T1} + G_{S1} + G_c}.$$

Power absorbed by the conversion conductance

$$P_c = \frac{I_1^2 G_c}{(G_{T1} + G_{S1} + G_c)^2}.$$

Efficiency of the primary circuit

$$\eta_1 = \frac{P_c}{P_1} = \frac{G_c}{G_{T1} + G_{S1} + G_c}.$$

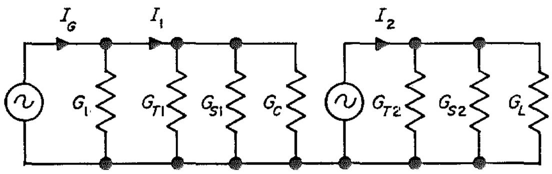


Figure 8—Equivalent circuit for a frequency multiplier.

7.2.2 Secondary Circuit

Input power

$$P_2 = \frac{I_2^2}{G_{T2} + G_{S2} + G_L}.$$

Output power

$$P_L = \frac{I_2^2 G_L}{(G_{T2} + G_{S2} + G_L)^2}.$$

Efficiency of the secondary circuit

$$\eta_2 = \frac{P_L}{P_2} = \frac{G_L}{G_{T2} + G_{S2} + G_L}.$$

Over-all efficiency

$$\eta = \eta_1 \eta_2 = \frac{G_L G_c}{(G_{T1}+G_{S1}+G_c)(G_{T2}+G_{S2}+G_L)}. \quad (39)$$

These equations show that the over-all circuit efficiency and output power at the selected harmonic depend on the loss conductances, the load conductance, and the conversion conductance. While the varactor losses are determined primarily by fabrication methods, the conversion conductance depends on operating conditions.

8. References

1. D. B. Leeson and S. Weinreb, "Frequency Multiplication with Nonlinear Capacitors—A Circuit Analysis," *Proceedings of the IRE*, volume 47, pages 2076–2084; December 1959.

2. C. W. Mueller and R. D. Gold, "High-Frequency Varactor Diodes," *RCA Review*, volume 21, pages 547–557, December 1960.

3. J. H. Forster and R. M. Ryder, "Diodes can do Almost Anything," *Bell Laboratories Record*, volume 39, pages 3–9; January 1961.

4. G. Luettganau, P. H. Dirnbach, and M. V. Griffin, "High Power at 1000 Megacycles Using Semiconductor Devices," Pacific Semi-Conductors, Inc.

5. Kenneth M. Johnson, "Large-Signal Analysis of a Parametric Harmonic Generator," *IRE Transactions on Microwave Theory and Techniques*, Volume MTT-8, pages 525–532; September 1960.

6. G. Adam, Standard Telephones and Cables Ltd; Laboratory Technical Report 600926.

7. C. H. Page, "Harmonic Generation with Ideal Rectifiers," *Proceedings of the IRE*, volume 46, pages 1738–1740; October 1958.

8. T. M. Manley and H. E. Rowe, "Some General Properties of Nonlinear Elements," Part 1, *Proceedings of the IRE*, volume 44, pages 904–913; July 1956.

9. T. Utsunomiya and S. Yuan, "Theory, Design and Performance of Maximum Efficiency Variable Reactance Frequency Multipliers," Columbia University, Technical Report T-1/164.

9. Acknowledgments

The author is indebted to Mr. L. Pollack and to Mr. W. Glomb for their critical review of this paper and for many helpful suggestions; to Mr. H. Goldman and Mr. E. Imboldi for discussions of various points during the course of this work.

W. Janeff was born in Burgas, Bulgaria, in October 1920. He received a Diplom Ingenieur degree from the Technische Hochschule in Vienna, Austria, in 1942 and a Ph.D. degree in 1945 from the Technische Hochschule in Dresden, Germany.

He joined the engineering staff of ITT Federal Laboratories in 1959 and is now a senior engineer in the space communication laboratory. Dr. Janeff is a member of the Association des Ingenieurs Docteurs de France and of the Institute of Electrical and Electronics Engineers.

Institute of Radio Engineers Celebrates 50th Anniversary

Founded in 1912, the Institute of Radio Engineers published in May 1962 an unusually interesting commemorative issue of the Proceedings of the IRE celebrating its 50th anniversary.

Following a factual presentation of the history of the Institute, over 50 of its Fellows contributed to a predictive symposium on communications and electronics in 2012. Henri Busignies, Vice President and General Technical Director of International Telephone and Telegraph Corporation, wrote in two fields, "Communication and Navigation" and "Electronic Spectro-Analysis of Chemical Compounds."

The major part of the editorial contents consists of 28 sections each prepared with the assistance of the appropriate Professional Group of the Institute and covering its field. They are abundantly rich in historical and tutorial material. In the section on aerospace and navigation is a paper on "The Air Traffic Control Equipment Subsystem—Present and Future" by P. C. Sandretto of the United States Defense Group of the International Telephone and Telegraph Corporation, and in the section on communication systems is one on "Radio Receivers—Past and Present" by Christopher Buff, Vice President and Chief Engineer of American Cable & Radio Corporation.

It is interesting to note that on 1 January 1963, the Institute of Radio Engineers and the 78-year-old American Institute of Electrical Engineers will combine and become the Institute of Electrical and Electronics Engineers.

Contribution to Studies of Overflow Traffic

J. CHASTANG

Compagnie Générale de Constructions Téléphoniques; Paris, France

1. Simple Overflow Problem

1.1 General Equations for Distribution: Adaptation of Riordan Method

Firstly consider a primary group of x trunks to which a random traffic a is offered and, secondly, a secondary group of y trunks to care for the traffic rejected by the primary group, that is, the overflow traffic. We have to calculate the probabilities $\theta(0), \theta(1), \cdots, \theta(n)$ of occupation of $0, 1, \cdots, n$ unspecified paths (and only this number) among the total number y of them, these probabilities being independent of the state of the primary group.

If we let $f(m, n)$ be the probability of m paths being busy in the lower group of x trunks and n being also busy among the y circuits of the upper group, the equations of state giving the probabilities $f(m, n)$ are

$$\left.\begin{aligned}(a+m+n)f(m,n)-(m+1)f(m+1,n)-(n+1)f(m,n+1)-af(m-1,n)&=0\\ (a+x+n)f(x,n)-af(x,n-1)-(n+1)f(x,n+1)-af(x-1,n)&=0\\ (x+y)f(x,y)-af(x,y-1)-af(x-1,y)&=0.\end{aligned}\right\}\quad(1)$$

This last equation having been introduced to take into account the fact that we have to consider a limited number of overflow trunks, instead of an unlimited group as Riordan has done. [1]. The factorial moment-generating function defined by

$$M(m,t)=\sum_{k=0}^{y}M_k(m)\frac{t^k}{k!}=\sum_{n=0}^{y}f(m,n)(1+t)^n \quad (2)$$

satisfies the following equations.

$$\left.\begin{aligned}(a+m)M(m,t)+t(D/dt)M(m,t)-(m+1)M(m+1,t)-aM(m-1,t)&=0\\ (-at+x)M(x,t)+t(D/dt)M(x,t)-aM(x-1,t)+af(x,y)t(1+t)^y&=0.\end{aligned}\right\}\quad(3)$$

In a way similar to Riordan's, the following recurrences may be found.

$$\left.\begin{aligned}(a+m+k)M_k(m)-(m+1)M_k(m+1)-aM_k(m-1)&=0\\ (x+k)M_k(x)-akM_{k-1}(x)-aM_k(x-1)+a\frac{k!}{y!}\binom{y}{k-1}M_y(x)&=0.\end{aligned}\right\}\quad(4)$$

The introduction of the new generating function

$$G_k(u)=\sum_{m=0}^{x}M_k(m)u^m \quad (5)$$

leads to the same equations

$$(a+k-u)G_k(u)+(u-1)(D/du)G_k(u)=0 \quad (6)$$

and by integration

$$G_k(u)=ce^{au}(1-u)^{-k} \quad \text{with} \quad c=M_k(0).$$

The quantities $M_k(0)$ in this case are of course different from the corresponding quantities in the case of an unlimited overflow. For their calculation, we could also use the last of (6), but in fact, as will be seen later, we shall only be interested in the values of $M_k(x)$. By replacing $M_k(x-1)$ in (4) by its value

$$M_k(x-1)=\sigma_k(x-1)M_k(0)=\frac{\sigma_k(x-1)}{\sigma_k(x)}\cdot M_k(x)$$

we obtain

$$\left[x+k-a\frac{\sigma_k(x-1)}{\sigma_k(x)}\right]M_k(x)=akM_{k-1}(x)-a\frac{k!}{y!}\binom{y}{k-1}M_y(x). \quad (7)$$

We recall that

$$\sigma_k(x)=\sum_{j=0}^{x}\binom{k-1+j}{k-1}\frac{a^{x-j}}{(x-j)!}$$

and that the quantities σ satisfy a good number of relations of which we shall use the following.

$$\left.\begin{aligned} &(m+k)\sigma_k(m)-a\sigma_k(m-1)=k\sigma_{k+1}(m) \\ \text{hence}\quad & \\ &k\sigma_{k+1}(m)=(m+k-a)\sigma_k(m)+a\sigma_{k-1}(m) \\ &\sigma_{k+1}(m)=\sigma_{k+1}(m-1)+\sigma_k(m). \end{aligned}\right\} \tag{8}$$

By use of the first relation for $m = x$, we arrive at the required recurrence

$$M_{k-1}(x)=\frac{1}{a}\cdot\frac{\sigma_{k+1}(x)}{\sigma_k(x)}\,M_k(x)+\frac{M_y(x)}{(y-k+1)!}. \tag{9}$$

This relation has already been given by Bech [2] and Le Gall [3] under slightly different forms.

1.2 Probabilities of Overflow Distribution

Having thus adapted Riordan's method to the particular case of a limited number of overflow trunks, we shall now calculate the distribution probabilities $\theta(n) = \sum_{m=0}^{x} f(m, n)$.

A simple result, which appears to be formulated here for the first time in studies of this question, is the following

$$\theta(n) = (a/n)f(x, n-1). \tag{10}$$

In fact, by summing m in (3), we obtain

$$aM(0,t) + t\frac{dM(0,t)}{dt} - M(1,t) = 0$$

$$(a+1)M(1,t) + t\frac{dM(1,t)}{dt} - 2M(2,t) - aM(0,t) = 0$$

$$(a+2)M(2,t) + t\frac{dM(2,t)}{dt} - 3M(3,t) - aM(1,t) = 0$$

...

$$(a+x-1)M(x-1,t) + t\frac{dM(x-1,t)}{dt} - xM(x,t) - aM(x-2,t) = 0.$$

$$(-at+x)M(x,t) + t\frac{dM(x,t)}{dt} - aM(x-1,t) = af(x,y)t(1+t)^y$$

...

$$-atM(x,t) + t\sum_{m=0}^{x}\frac{dM(m,t)}{dt} = af(x,y)t(1+t)^y$$

as $$\sum\frac{dM(m,t)}{dt} = \theta(1) + 2(1+t)\theta(2) + 3(1+t)^2\theta(3) + \cdots + y(1+t)^{y-1}\theta(y).$$

We deduce $n\theta(n) = af(n, n-1)$ by equating the coefficients of the powers of t.

We have thus

$$\theta(y) = (a/y)f(x, y-1)$$
$$\theta(y-1) = (a/y-1)f(x, y-3)$$
$$\ldots$$
$$\theta(1) = a\cdot f(x, 0)$$
$$\theta(0) = 1 - \theta(1) - \theta(2) - \cdots - \theta(y).$$

To calculate the quantities

$$f(x, y-1),\ f(x, y-2), \text{ et cetera,}$$

we shall use the equations of the factorial moments.

(*To Top of Right-Hand Column*)

$$y!f(x, y) = M_y(x)$$
$$(y-1)!f(x, y-1) + y!f(x, y) = M_{y-1}(x)$$
$$(y-2)!f(x, y-2) + (y-1)!f(x, y-1) + \frac{y!}{2!}f(x, y) = M_{y-2}(x),$$

et cetera.

The quantities $M_k(x)$ are given by (9), which we shall write from now on as

$$M_{k-1}(x) = \frac{S_{k+1}}{a}M_k(x) - \frac{M_y(x)}{(y-k+1)!}$$

with the notation

$$S_{k+1} = \frac{\sigma_{k+1}(x)}{\sigma_k(x)}.$$

Thus we obtain

$$M_{y-1}(x) = \left(1 + \frac{S_{y+1}}{a}\right)M_y(x)$$
$$M_{y-2}(x) = \left(\frac{1}{2!} + \frac{S_y}{a} + \frac{S_y\cdot S_{y+1}}{a^2}\right)M_y(x)$$
$$M_{y-3}(x) = \left(\frac{1}{3!} + \frac{1}{2!}\frac{S_{y-1}}{a} + \frac{S_{y-1}\cdot S_y}{a^2} + \frac{S_{y-1}\cdot S_y\cdot S_{y+1}}{a^3}\right)M_y(x)$$
$$M_{y-4}(x) = \left(\frac{1}{4!} + \frac{1}{3!}\frac{S_{y-2}}{a} + \frac{1}{2!}\frac{S_{y-2}\cdot S_{y-1}}{a^2} + \frac{S_{y-2}\cdot S_{y-1}\cdot S_y}{a^3} + \frac{S_{y-2}\cdot S_{y-1}\cdot S_y\cdot S_{y+1}}{a^4}\right)M_y(x)$$

et cetera.

The equations giving the probabilities $\theta(n)$ are thus

$$\left.\begin{aligned}
\theta(y) &= S_{y+1}E_{x+y}(a) \quad \text{as} \quad f(x, y) = E_{x+y}(a)\\
\theta(y-1) &= \frac{y}{a}[S_y\cdot S_{y+1} + a(S_y - S_{y+1})]E_{x+y}(a)\\
\theta(y-2) &= \frac{y(y-1)}{a^2}\left[S_{y-1}\cdot S_y\cdot S_{y+1} + aS_y(S_{y-1} - S_{y+1})\right.\\
&\qquad\left. + \frac{a^2}{2!}(S_{y-1} + 2S_y + S_{y+1})\right]E_{x+y}(a)
\end{aligned}\right\} \quad (11)$$

et cetera.

They may be written in the following way, which may be clearer.

$$\left.\begin{aligned}
\theta(y) &= \frac{\sigma_{y+1}}{\sigma_y}\cdot E_{x+y}(a)\\
\theta(y-1) &= \frac{y}{a}\left[\frac{\sigma_{y+1}}{\sigma_{y-1}} + a\left(\frac{\sigma_y}{\sigma_{y-1}} - \frac{\sigma_{y+1}}{\sigma_y}\right)\right]E_{x+y}(a)\\
\theta(y-2) &= \frac{y(y-1)}{a^2}\left[\frac{\sigma_{y+1}}{\sigma_{y-2}} + a\left(\frac{\sigma_y}{\sigma_{y-2}} - \frac{\sigma_{y+1}}{\sigma_{y-1}}\right) + \frac{a^2}{2!}\left(\frac{\sigma_{y-1}}{\sigma_{y-2}} - 2\frac{\sigma_y}{\sigma_{y-1}} + \frac{\sigma_{y+1}}{\sigma_y}\right)\right]E_{x+y}(a)
\end{aligned}\right\} \quad (12)$$

et cetera.

The above expressions become more and more complex but the most useful $\theta(y)$, $\theta(y-1)$, $\theta(y-2)$ are relatively simple especially if we can work out a simple way of obtaining the quantities S_{y+1}, S_y, S_{y-1}, et cetera.

1.3 Approximate Calculation of S

We have $\quad S_1 = (1/E_x(a))$

and $\quad S_2 = x + 1 - a + aE_x(a)$

and according to the recurrent relation (8)

$$kS_{k+1} = x + k - a + (a/S_k). \qquad (13)$$

Thus if the number of the lines of the overflow group is not too high, it is quite easy to calculate all the values, each one in turn, starting from the first.

But when, on the contrary the number y is high enough to result in a long and tedious computation, it seems easier to calculate directly the quantities S_{y+1}, S_y, et cetera, by making use of an approximation.

(*To Top of Right-Hand Column*)

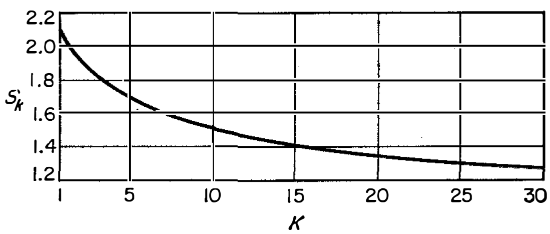


Figure 1.

If we make S_k satisfy (13), it is clear from Figure 1 that its curve quickly assumes a flat shape with an asymptote of 1, so that an approximation of S_{k+1} may be derived from

$$(k+\tfrac{1}{2})S_{k+1}^2 - (x+k+\tfrac{1}{2}-a)S_{k+1} - a = 0. \qquad (14)$$

This equation has a positive root that gives a good approximate value of S_{k+1}.

1.4 Numerical Application

We shall examine a particular example using the following data.

Primary group $x = 12$ lines

Overflow group $y = 20$ lines

Offered traffic $a = 21$ erlangs.

The exact values of S_k calculated by (13) are given in Table 1 and plotted in Figure 1.

TABLE 1
VALUES OF S_k

S_1	2.10825	S_{11}	1.48771	S_{21}	1.33205
S_2	1.96085	S_{12}	1.46506	S_{22}	1.32215
S_3	1.85482	S_{13}	1.44449	S_{23}	1.31287
S_4	1.77395	S_{14}	1.42600	S_{24}	1.30415
S_5	1.70950	S_{15}	1.40904	S_{25}	1.29593
S_6	1.65683	S_{16}	1.39359	S_{26}	1.28818
S_7	1.61247	S_{17}	1.37931	S_{27}	1.28085
S_8	1.57478	S_{18}	1.36617	S_{28}	1.27390
S_9	1.54190	S_{19}	1.35397	S_{29}	1.26731
S_{10}	1.51329	S_{20}	1.34263	S_{30}	1.26105

The exact values of the occupation probabilities are

$$\theta(20) = 1.33205 \times 0.00593 = 0.00793; \quad E_{32}(21) = 0.005953$$

$$\theta(19) = \frac{20}{21}(1.78845 + 0.22226)0.005953 = 0.01140$$

$$\theta(18) = \frac{20 \times 19}{21 \times 21}(2.42150 + 0.61804 + 0.16577)0.005953 = 0.01644$$

$$\theta(17) = \frac{20 \times 19 \times 18}{21 \times 21 \times 21}(3.3082 + 1.3028 + 0.5393 + 0.1851)0.005953 = 0.02345.$$

The approximate values of S given by (14) are

$$S_{21} = 1.3308, \quad S_{20} = 1.3413, \quad S_{19} = 1.3527, \quad S_{18} = 1.3649$$

hence

$$\theta(20) = 1.3308 \times 0.005953 = 0.0079$$

$$\theta(19) = \frac{20}{21}(1.7850 + 0.2205)0.005953 = 0.0114$$

$$\theta(18) = \frac{20 \times 19}{21 \times 21}(2.4146 + 0.6169 + 0.1984)0.005953 = 0.0166$$

$$\theta(17) = \frac{20 \times 19 \times 18}{21 \times 21 \times 21}(3.2957 + 1.2992 + 0.5621 + 0)0.005953 = 0.022.$$

As may be seen, this approximation may be considered to be satisfactory.

It should be noted that values of S_{19}, S_{18}, et cetera, drawn from the recurrent relation (13) by making use of the approximate value of S_{20} obtained from (14) are less accurate than the corresponding values directly obtained from (14).

1.5 Variance of Overflow Distribution

The mean of the distribution $\bar{m}$ equals $aE_x(a) - aE_{x+y}(a) = \alpha - \beta$ if we represent by α the overflow traffic and by β the traffic in the last instance.

The overflow distribution variance v equals $\sum_0^y n^2\theta(n) - \bar{m}^2$.

$$\begin{aligned}
n^2\theta(n) &= anf(x, n-1) = a(n-1)f(x, n-1) + af(x, n-1) \\
\sum n^2\theta(n) &= a[M_1(x) - yf(x,y) + M_0(x) - f(x,y)] \\
&= a[M_1(x) + M_0(x) - (1+y)f(x,y)]
\end{aligned}$$

$$\begin{aligned}
M_1(x) &= E_x(a) \\
M_1(x) &= (a/S_2)[M_0(x) - f(x,y)]
\end{aligned}$$

hence after reduction

$$\sum n^2\theta(n) = \frac{a(\alpha - \beta)}{x + 1 + \alpha - a} + \alpha - \beta(1+y)$$

and

$$v = \frac{a(\alpha - \beta)}{x + 1 + \alpha - a} + \alpha - \beta(1+y) - (\alpha - \beta)^2.$$

1.6 General Equations for Factorial Moments

By fully solving (9), it is found

$$M_y(x) = \frac{a^y \sigma_0(x)}{\sigma_{y+1}(x)} \cdot \frac{\sigma_{y+1}(x)}{\sigma_{y+1}(x) + a\sigma_y(x) + \frac{a^2}{2!}\sigma_{y-1}(x) + \cdots + \frac{a^y}{y!}\sigma_1(x)}$$

$$M_{y-1}(x) = \frac{a^{y-1}\sigma_0(x)}{\sigma_y(x)} \cdot \frac{\sigma_{y+1}(x) + a\sigma_y(x)}{\sigma_{y+1}(x) + a\sigma_y(x) + \frac{a^2}{2!}\sigma_{y-1}(x) + \cdots + \frac{a^y}{y!}\sigma_1(x)}$$

$$\cdots$$

$$M_1(x) = \frac{a\sigma_0(x)}{\sigma_2(x)} \cdot \frac{\sigma_{y+1}(x) + a\sigma_y(x) + \cdots + \frac{a^{y-1}}{(y-1)!}\sigma_2(x)}{\sigma_{y+1}(x) + \cdots + \frac{a^y}{y!}\sigma_1(x)}.$$

If we represent by $\mu_y(x)$, $\mu_{y-1}(x)$, $\cdots$, $\mu_0(x)$, the moments of the unlimited overflow distribution, $M_y(x)$ may be written

$$M_y(x) = \mu_y(x) \frac{1}{1 + \frac{\mu_y(x)}{\mu_{y-1}(x)} + \frac{1}{2!}\frac{\mu_y(x)}{\mu_{y-2}(x)} + \frac{1}{3!}\frac{\mu_y(x)}{\mu_{y-3}(x)} + \cdots + \frac{1}{y!}\frac{\mu_y(x)}{\mu_0(x)}}. \tag{15}$$

This expression appears as a generalization of the Erlang equation.

2. Composition of Overflow Traffic from 2 Primary Groups

Let us consider 2 primary groups of trunks with the following characteristics.

x trunks for the first group offered traffic (poissonian) a

x' trunks for the second group, offered traffic (poissonian) b.

We shall denote by $f(m, m', n)$ the probability for simultaneously having m and m' trunks busy in the primary group and also n trunks busy in the overflow group.

2.1 Unlimited Overflow Group. Moments of the Unlimited Overflow Distribution

The engagement of trunks in the overflow group results from the calls rejected by both primary groups. Let us write $n = \nu + \nu'$. The distributions ν and ν' are independent in the case of the unlimited overflow. The factorial moments of the resulting overflow distribution are the factorial moments of a sum of 2 independent variables. It is known that the order k moment, for the occupation situation (m, m') in the primary groups reads

$$M_k^{(n)}(m, m') = \mu_k^{(\nu)}(m) \cdot \mu_0^{(\nu')}(m') + \binom{k}{1} \mu_{k-1}^{(\nu)}(m) \cdot \mu_1^{(\nu')}(m')$$
$$+ \binom{k}{2} \mu_{k-2}^{(\nu)}(m) \cdot \mu_2^{(\nu')}(m') + \cdots + \mu_0^{(\nu)}(m) \cdot \mu_k^{(\nu')}(m'). \tag{16}$$

If we denote by $\mu_k^{(\nu)}(m)$ and $\mu_k^{(\nu')}(m)$ the factorial moments of order k of the distributions ν and ν', respectively.

For instance, the second moment has the following value

$$M_2(m, m') = \frac{a^2\sigma_0(x)\sigma_2(m)}{\sigma_3(x)\sigma_2(x)} \cdot \frac{\sigma_0'(m')}{\sigma_1'(x')} + 2a\frac{\sigma_0(x)\sigma_1(m)}{\sigma_2(x)\sigma_1(x)} \cdot b \cdot \frac{\sigma_0'(x)\sigma_1'(m')}{\sigma_2'(x)\sigma_1'(x')} + \frac{\sigma_0(m)}{\sigma_1(x)} \cdot b^2 \cdot \frac{\sigma_0(x')\sigma_2'(m')}{\sigma_3'(x')\sigma_2'(x')}$$

the polynomials σ and σ' having to be taken with their respective values in each group.

2.2 Limited Overflow Group

Let us now assume the overflow group to be limited to a number of trunks y. The equations of state analogous to (1) become

$$(a+b+m+m'+n)f(m,m',n)-(m+1)f(m+1,m',n)-(m'+1)f(m,m'+1,n) \\ -(n+1)f(m,m',n+1)-af(m-1,m',n)-bf(m,m'-1,n)=0. \quad (17)$$

If $m'=x'$

$$(a+b+m+x'+n)f(m,x',n)-(m+1)f(m+1,x',n)-f(m,x',n+1) \\ -af(m-1,x',n)-bf(m,x'-1,n)-bf(m,x',n-1)=0. \quad (18)$$

For $m=x$ and $m'=x'$

$$(a+b+x+x')f(x,x',n)-(n+1)f(x,x',n+1)-af(x-1,x',n) \\ -bf(x,x'-1,n)-(a+b)f(x,x',n-1)=0. \quad (19)$$

Finally for $n=y$, the term with $(n+1)$ disappears. If now $m=x$, the term with $(m+1)$ disappears in (18) as does the quantity a in the first term.

If lastly $m=x$, $m'=x'$, and $n=y$, (19) reduces to

$$(x+x'+y)f(x,x',y)-af(x-1,x',y)-bf(x,x'-1,y)-(a+b)f(x,x',y-1)=0. \quad (20)$$

Let us once more denote

$$M(m,m',t)=\sum_{k=0}^{y} M_k(m,m')\frac{t^k}{k!}.$$

It is found that the generating function of factorial moments, the function M, satisfies the following equations.

$$\left.\begin{aligned}
&(a+b+m+m')M(m,m',t)+t(d/dt)M(m,m',t)-(m+1)M(m+1,m',t)\\
&\qquad -(m'+1)M(m,m'+1,t)-aM(m-1,m',t)-bM(m,m'-1,t)=0\\
&(a+m+x'-bt)M(m,x',t)+t(d/dt)M(m,x',t)-(m+1)M(m+1,x',t)\\
&\qquad -aM(m-1,x',t)-bM(m,x'-1,t)+bt(1+t)^y f(m,x',y)=0.\\
&(b+x+m'-at)M(x,m',t)+t(d/dt)M(x,m',t)-(m'+1)M(x,m'+1,t)\\
&\qquad -aM(x-1,m',t)-bM(x,m'-1,t)+at(1+t)^y f(x,m',y)=0.\\
&[x+x'-(a+b)t]M(x,x',t)+t(d/dt)M(x,x',t)-aM(x-1,x',t)\\
&\qquad -bM(x,x'-1,t)+(a+b)t(1+t)^y f(x,x',y)=0.
\end{aligned}\right\} \quad (21)$$

By making use of a process analogous to that of the first part, we obtain the recurrent relations applying to the factorial moments:

$$(a+b+m+m'+k)M_k(m,m')-(m+1)M_k(m+1,m')-(m'+1)M_k(m,m'+1) \\ -aM_k(m-1,m')-bM_k(m,m'-1)=0. \quad (22)$$

$$(b+x+m'+k)M_k(x,m')-(m'+1)M_k(x,m'+1)-bM_k(x,m'-1) \\ -aM_k(x-1,m')-akM_{k-1}(x,m')+\frac{k!}{y!}\binom{y}{k-1}aM_y(x,m')=0. \quad (23)$$

$$(a + m + x' + k)M_k(m, x') - (m + 1)M_k(m + 1, x') - aM_k(m - 1, x') - bM_k(m, x' - 1) - bkM_{k-1}(m, x') + \frac{k!}{y!}\binom{y}{k - 1} bM_y(m, x') = 0. \quad (24)$$

$$(x + x' + k)M_k(x, x') - bM_k(x, x' - 1) - aM_\kappa(x - 1, x') - (a + b)kM_{\kappa-1}(x, x') + \frac{k!}{y!}\binom{y}{k - 1} (a + b)M_y(x, x') = 0. \quad (25)$$

2.3 Loss Probabilities

The loss probability to be applied to the traffic a offered to the first group is the probability for the lines of both this primary group and the overflow group being engaged in totality. This probability is $\sum_{m'=0}^{x'} f(x, m', y)$ that we shall write in brief form as $f(x, y)$ and we have

$$f(x, y) = \frac{1}{y!}\sum_{m'=0}^{x'} M_y(x, m') = \frac{1}{y!} M_y(x).$$

We are thus led to sum up (23) and (25) over all possible values of m' and obtain

$$(x + k)M_k(x) - aM_k(x - 1) - akM_{k-1}(x) - bkM_{k-1}(x, x') + \frac{k!}{y!}\binom{y}{k - 1} aM_y(x) + \frac{k!}{y!}\binom{y}{k - 1} bM_y(x, x') = 0. \quad (26)$$

and in particular

$$(x + y)M_y(x) - aM_y(x - 1) - ayM_{y-1}(x) - b_yM_{y-1}(x, x') + ayM_y(x) + b_yM_y(x, x') = 0.$$

If one compares the latter equation to the last of (4) and notes that such quantities as $M_{k-1}(x, x')$ and $M_y(x, x')$ are comparatively small, one is led to expect that an equation analogous to (15) may give an approximate value of $M_y(x)$, provided the quantities $\mu(x)$ are given the actual values resulting from (16) of Section 2.2. The greater the number of trunks of the primary groups, the better should be the approximation, but on the other hand the result may not be so good in the case of overflow groups built up of a large number of lines.

2.4 Verification of Approximate Equation

The numerical results published hitherto are very few and concern only primary groups of rather small size. This is due to the difficulty of solving the important set of linear equations involved in any method, despite the substantial simplifications proposed by Le Gall [6] for the handling of the matrix calculations.

G. Neovius [4] has treated the case of $x = x' = 3$ for various traffic values: $a = b = 1, 2, 3, 4,$ and 5 erlangs. A comparison of the exact results and the approximate values given by the proposed equation are shown in Table 2, in which E_y denotes the loss to be applied to the primary traffic (traffic offered).

TABLE 2
COMPARISON OF EXACT AND CALCULATED LOSS TO PRIMARY TRAFFIC

a	E_1 in Percent		E_2 in Percent		E_3 in Percent	
	Exact	Approximate	Exact	Approximate	Exact	Approximate
1	1.73716	1.750	0.39562	0.402	0.07678	0.0785
2	11.54760	11.684	5.62963	5.764	2.43797	2.5225
3	24.56570	24.754	16.39860	16.663	10.21233	10.465
4	35.93526	36.095	27.70477	27.968	20.53455	20.842
5	44.91999	45.039	37.33561	37.547	30.30786	30.435

2.5 Case of Several Primary Groups

The generalization of (16) for the case of more than 2 primary groups is self-evident in the same way that M_2, for instance, corresponds to a symbolic power $(a + b)^2$ in the case of 2 primary groups, the same quantity corresponds to the symbolic power $(a + b + c + d)^2$ in the case of 4 primary groups.

G. Bretschneider [7] has published some results concerning 4 primary groups of $x = x' = x'' = x''' = 4$ lines for traffic values $a = b = c = d = 0.75$, 2.25, 3, 4.50, and 10 erlangs, and an overflow group of $y = 6$ lines. Comparative results for $a = 2.25$ and 3 erlangs are also given in Table 3.

TABLE 3
LOSS PROBABILITY

a	Exact	Approximate
2.25	0.15425×10^{-2}	0.1675×10^{-2}
3	0.15539×10^{-1}	0.1676×10^{-1}

The approximate equations, satisfactory as they be, fail however to give a better accuracy than the approximate method of R. I. Wilkinson. Nevertheless, they may be used to investigate the important problem of determining the loss that is to be assigned to each element of traffic among those that compose the total resulting overflow.

2.6 Remarks

In the same way as Poisson's equation gives a lower, though valuable approximation for the Erlang equation, we may also consider the probability $f(x, y)$ of having x busy lines in the concerned primary group and simultaneously y calls in the unlimited overflow group. The latter probability is given by

$$f(x, y) = \frac{1}{y!}\Big[M_y(x) - M_{y+1}(x) + \frac{1}{2!}M_{y+2}(x) - \frac{1}{3!}M_{y+x}(x)\cdots\Big]$$

in which for instance

$$M_y(x) = \sum_{m'=0}^{x'} M_y(x, m') = \frac{a^y\sigma_0(x)}{\sigma_{y+1}(x)} + \binom{y}{1}\frac{a^{y-1}\sigma_0(x)}{\sigma_y(x)}\cdot\frac{b\sigma_0'(x)}{\sigma_1'(x')} + \binom{y}{2}\frac{a^{y-2}\sigma_0(x)}{\sigma_{y-1}(x)}\cdot\frac{b^2\sigma_0'(x')}{\sigma_2'(x')} + \cdots.$$

The series giving $f(x, y)$ converges slowly in most cases and requires a long calculation. It gives a result that is, by far, less accurate than the result of the method of Section 2.4 even in the case of overflow groups of very-low loss probabilities.

2.7 Remarks

By summing up $(22)\cdots(25)$ over all possible values of m and m', it is found for the M_k

factorial moment of distribution of the overflow group limited to y lines

$$M_k = a\left[M_{k-1}(x) - \frac{k!}{y!}\binom{y}{k-1} M_y(x)\right] + b\left[M_{k-1}(x') - \frac{k!}{y!}\binom{y}{k-1} M_y(x')\right].$$

This result is nothing more than a generalization of the results of Section 1.2, which permits us to write

$$y\theta(y) = af(x, y-1) + bf(x', y-1)$$
$$(y-1)\theta(y-1) = af(x, y-2) + bf(x', y-2).$$

3. References

1. R. I. Wilkinson, "Theories for Toll Traffic Engineering in the USA," *Bell System Technical Journal*, volume 35, pages 421–514; March 1956.

2. N. Z. Bech, "Méthode de calcul de la perte dans des systèmes de grading et de circuits de débordement," *Teleteknik*, volume 5; December 1954.

3. P. Le Gall, "Les trafics téléphoniques et la sélection conjuguée en téléphonie automatique," *Annales des Télécommunications*, volume 13, numbers 7–12; 1958.

4. G. Neovius, "Artificial Traffic Trials Using Digital Computers," *Ericsson Technics*, volume 11, number 2; 1955.

5. E. Brockmeyer, "The Simple Overflow Problem in the Theory of Telephone Traffic," *Teleteknik*, volume 4; December 1954.

6. P. Le Gall, "Le trafic de débordement," *Annales des Télécommunications*, volume 16, numbers 9 and 10; September and October 1961.

7. G. Bretschneider, "L'importance des méthodes exactes pour les calculs simulatifs et approximatifs destinés a la détermination de la probabilité de perte." Presented at the Third Teletraffic Congress in Paris, 1961.

J. Chastang was born in Mauriac, Cantal, France, on 23 November 1903. In 1926, he received the diploma of the Ecole Polytechnique of Paris.

He served in the Signals Branch of the French Génie Militaire. From 1932 until 1941, he was head of the technical department of Public Telephone Service in the French mandated territories of Syria and Lebanon. From 1941 to 1943, he was with the French Posts, Telegraphs, and Telephones Administration as a transmission engineer and, as such, belonged to the Direction des Recherches et du Contrôle Technique.

Mr. Chastang is now in charge of traffic studies for Compagnie Générale de Constructions Téléphoniques.

Interregister Multifrequency-Code Signalling for Telephone Switching in Europe

M. DEN HERTOG
ITT Europe; Brussels, Belgium

Important developments in the technique of signalling used in telephone switching have taken place during the past few years in Europe and will have a far-reaching effect on the basic principles underlying switching methods as well as on the operating facilities that will be provided for both national networks and for international connections in Europe. Since agreement on these methods has now been reached among many of the leading telephone administrations in Europe, a description of these developments, their basic philosophy, and their implications is thought to be timely.

1. History

The need for fast and reliable signalling methods for telephone switching was recognized independently around 1956 or 1957 by at least three organizations working on new or improved switching systems in Europe.

At this time, the Netherlands administration considered the use of multifrequency-code signalling for the transfer of numerical information on their toll network as an essential requirement for reducing connection time. In Belgium a fast signalling method was needed for international subscriber's toll dialling to signal the identity of the calling subscriber, during the establishment of the connection, via national toll lines to recording equipment that was to be provided for automatic message accounting at the centrally located international toll office. More than one manufacturer was engaged in the development of crossbar switching equipment in which a fast signalling system operating via 2-wire interoffice junctions would provide means for simplifying the common control equipment. For example, at the Bell Telephone Manufacturing Company in Antwerp, Belgium, plans were conceived for using a uniform code of multifrequency signals for transmitting all numerical and other information needed in establishing telephone connections and for signalling between different stages of selection in the same central office or in different offices, both on a national and international scale.

The issuance of technical recommendations on signalling methods for international toll lines being in the province of the Comité Consultatif International Télégraphique et Téléphonique, with which international body many administrations had collaborated to design two recommended signalling systems for use on international toll lines in Europe and the Mediterranean basin, it was recognized that any system proposed for international use would have to be submitted to this body for discussion.

Informal discussions with several European administrations showed, however, a widespread desire to consider improved methods of signalling with emphasis on their use for national systems. Late in 1957 the Netherlands administration took the initiative to call a conference on these questions; all administrations and the leading manufacturers of telephone equipment in Western Europe were invited. This conference took place in March 1958 and was attended by representatives of 8 administrations, 3 large manufacturing organizations, and furthermore by observers from France, the United Kingdom, and in the person of its director by the Comité Consultatif International Télégraphique et Téléphonique. Contributions were made by the administrations of the Netherlands, Denmark. the United Kingdom, and by each of the manufacturers.

A second meeting, again on the initiative of the Netherlands administration, took place in September 1959 at Brussels between the administrations of the common-market countries, and a third, at the invitation of the European Conference of the Administrations of Post and Telecommunications in October 1960 at Paris, in which 22 countries participated.

The voluminous documents that had been prepared at Brussels for further study could not, however, be fully discussed at Paris, at which delegates were present from many countries that had not previously participated in these studies, and no decisions were reached.

Since then, discussions among various administrations and manufacturers* have led to a crystallization of requirements and to a large degree of agreement on the signalling code to be used. At a meeting at Munich in March 1962, which was attended by delegates of the administrations of Belgium, Denmark, Italy, The Netherlands, Norway, Sweden, Switzerland, and Western Germany, at the invitation of the latter, general agreement was reached on a code of signals and the consequent methods of operation. A complete specification on this system will be prepared for discussion at a (it is hoped) final meeting in the autumn of 1962, to which other administrations will also be invited.

2. General Principles

The general principle on which the plan is based is that the signals used in the establishment and supervision of telephone connections will be divided into two categories.

2.1 Line Signals

Line signals make use of signal transmitters and receivers associated with each individual line or speech channel. They comprise all signals that must be exchanged between the line terminations when no common-control or register equipment is connected at either or both of these terminations. The signals shown in Table 1 form part of this category; their definitions and functions are explained in Section 5.2.

As all of these signals, if provided, make use of the line signalling equipment they may be transmitted in various manners as determined by the characteristic properties of this signalling equipment and have the same signal composition as employed for corresponding signals on existing line equipment on which all signals are line signals. In other words, existing junction and toll line equipment may be used unchanged, whether these lines are provided with various types of in-band voice-frequency signalling or out-of-band signalling, whether the signalling system is of the continuous or pulse (spurt) type, whether the toll lines are used for one-way or both-way operation, whether signal discrimination is obtained by signal length or frequency, or whether the lines or junctions are of the 2-wire or 4-wire type.

TABLE 1
LINE SIGNALS

Forward Signals	Backward Signals
Seizure	Acknowledgment of Seizure*
Clear Forward	Answer or Reanswer
Forward Transfer*	Clear Back
Toll Offering*	Release Guard*
Reringing*	Blocking*
	Acknowledgment of Optional Forward Signals*

* Optional

The only requirement the line signalling equipment should meet to be used with the new system is that the signal receivers must not respond to frequencies between 500 and 2000 hertz which requirement is fulfilled by modern equipment meeting recommendations of the Comité Consultatif International Télégraphique et Téléphonique.

It should be observed that, in comparison with methods employing line signals exclusively, fewer line signals are needed with the new methods, so that it may be possible to simplify the relay sets associated with the line terminations for new equipment. Further, the new methods, employing a combination of line-and-register signals, do not in any way influence the introduction of changes in existing line signalling codes if such were found desirable to increase reliability and/or transmission speed of line signals.

* H. Pausch, A. Pfau, and F. Pfleiderer, "Multifrequency Code Dialing Method for Worldwide Telephone Traffic," *NTZ Communications Journal,* volume 1, number 1, page 13; 1962.

As line signals are independent of the particular subject of register signals to be dealt with here, they will not be considered further in the following section.

2.2 Register Signals

Register signals make use of signal transmitters and receivers associated with the common control equipments or registers that are temporarily engaged during the establishment of a connection. They comprise all signals exchanged between these common circuits. Although these signals are transmitted via the speech channels by which these common circuits are interconnected during certain periods in the establishment of the connections, no mutual interference can take place between the register and line signals as they employ different frequencies.

The following signals may form part of this category.

Forward Signals

Numerical signals

A. Constituting number of called line
B. Constituting number of calling line
C. Indicating call to operator
D. Indicating call to routine test equipment

End of sending

Affirmation that echo suppressor is needed

Rejection of request for information

Category of caller

A. Subscriber without preference
B. Subscriber with preference
C. Operator
D. Test call
E. Data transmission

Backward Signals

Proceed-to-send with

A. Next digit
B. Digit transmitted last but 1
C. Digit transmitted last but 2
D. Digit transmitted last but 3
E. First digit of international country code
F. First digit of national number

Request for transmission of

A. Local number of calling line
B. National number of calling line
C. Code of originating country
D. Category of caller
E. Need for echo suppressor
F. Subsequent figures with dial pulsing

End-of-selection

A. Without further information on condition of called line
B. Preparative to transmitting information on condition of called line

Condition of called line

A. Free with call charging
B. Free without call charging
C. Free with holding of malicious callers
D. Busy
E. Out-of-order
F. Changed number
G. Not used

Congestion

A. In toll switching equipment
B. In local switching equipment

Dead level

It should be understood that the above list of register signals shows only the possibilities incorporated in the system. All common equipments do not have to be arranged for transmitting or receiving all of these signals. A number of them are for optional use, such as the signals relating to transmission of the number of the calling line, which are to be provided only if the local switching equipment is arranged for identifying the calling line.

The signals relating to the category of the caller and to the condition of the called line need be provided similarly only if the local switching equipment (and possibly the national toll dialling system) are capable of providing corresponding electric signals.

Other signals shown will be required only by the common equipments located at a relatively small number of particular types of offices, for example a number of signals are used only on international connections and will be employed only by registers controlling the establishment of international connections. Some other signals will be required only for national toll connections and will be provided only at the national toll registers. None of these two classes of signals will be required at the registers of local offices.

This will be discussed further hereafter.

2.3 Signals for Interregister Signalling

It follows from the above that these so-called register signals are used for the transmission of digits and associated orders for establishing connections, as well as for transmitting information on the condition of certain equipments engaged in the connection. It is, therefore, necessary to transmit information in both a forward and backward direction relative to the direction in which the connection is set up. For the signals in each direction, a self-checking multifrequency code is used, in which each signal is composed of 2 frequencies out of a group of 6.

In this manner, 15 signal combinations are available for each direction. Arrangements may be made so that in both directions—by the transmission of appropriate "shift" signals—some or all of these signals may be given new meanings during different stages of the establishment of the connection, so that the total number of signal meanings that may be transmitted is larger than 15 in each direction.

As it is desirable for signalling systems to be effective on both 2-wire and 4-wire circuits, separate groups of 6 frequencies are employed for each direction, so that a total of 12 frequencies is provided. In each group the 6 frequencies are spaced at intervals of 120 hertz and the space between the 2 groups has been fixed at 240 hertz with a highest frequency of 1980 hertz; the lowest frequency that may be used thereby becomes 540 hertz. The following frequency allocation has been accepted.

Signals in forward direction use 1380 through 1980 hertz.

Signals in backward direction use 540 through 1140 hertz.

A complete list of register signals is given in Table 2. Each signal comprises 2 of the frequencies indicated by the 2 crosses in the corresponding frequency columns. The code is of the additive type. In the table, a broken line separates the 2 frequencies used for each signal. The frequency indicated to the left of this line corresponds to an index figure for each frequency, shown at the top, and that shown at the right of the line corresponds to a weight figure shown at the bottom of the table. The code number or numerical value of each signal may be found by adding the values of the index and weight figures indicated.

2.4 Compelled Operation

A system has been adopted in which one signal in the forward direction and one signal in the backward direction are always used in combination, so that the length of each of these signals is determined by the other. This has been called the compelled method of operation for this reason. The exchange of signals always commences with a forward signal that has no definite length but continues until a backward signal is received that is transmitted on receipt and in acknowledgment of the forward signal. In its turn, the backward acknowledgment signal continues until the termination of the forward signal is recognized. The next forward signal may be transmitted from the moment the termination of the backward signal is recognized at the originating end. The principle of this operation has been indicated diagrammatically in Figure 1.

The fact that signals in the two directions of operation are available permits the adoption of this compelled type of operation, which has the following advantages.

A. The length of each signal and the intervals between signals need not be specified accurately, as is the case with systems using pulses and intervals of predetermined duration without acknowledgment of each signal. This avoids the use of apparatus capable of providing and accepting these pulses within prescribed limits, as well as maintenance to ensure continued operation within these limits.

B. The system is insensitive to receiver distortion and will work as long as the receiver responds to the received signal. This renders the performance specification of the receiver less stringent and permits operation over a wider range of signal levels than if the receiver would perform only within narrow limits of distortion.

C. The arrangement is less sensitive to disturbances in the transmission media than noncompelled systems. Neither short openings nor the presence of other frequencies is able to disturb a signal, since its transmission is continued until it has been properly received and acknowledged. Once a signal has been recognized, it will be recorded so that no disturbance may be caused by subsequent irregularities in the transmission equipment during the time the signal continues. It should be noted that the codes are self-checking, for only when 2 frequencies are received will a signal be accepted. Signals of 1 frequency or of more than 2 frequencies will be ignored.

TABLE 2
DENOMINATION AND COMPOSITION OF REGISTER SIGNALS

Code	Numerical Value of Code		Frequency in Cycles per Second (Hertz)					
		Forward	1380	1500	1620	1740	1860	1980
		Backward	1140	1020	900	780	660	540
		Index	f0	f1	f2	f3	f4	f5
			Rank Frequency					
15	4 + 11						x	x
14	3 + 11					x		x
13	2 + 11				x			x
12	1 + 11			x				x
11	0 + 11		x					x
10	3 + 7					x	x	
9	2 + 7				x		x	
8	1 + 7			x			x	
7	0 + 7		x				x	
6	2 + 4				x	x		
5	1 + 4			x		x		
4	0 + 4		x			x		
3	1 + 2			x	x			
2	0 + 2		x		x			
1	0 + 1		x	x				
							Base Frequency	
		Weight	0	1	2	4	7	11

2.5 Link-by-Link and End-to-End Signalling

With the type of signalling described above, use may be made of either of 2 different principles for the exchange of signals between registers, which are known as "link-by-link" and "end-to-end" signalling, respectively. *Link-by-link signalling* implies that the register at each, except the last, office involved in a connection must receive all figures of the called number, including those needed for controlling the selecting operations at that office and other figures that may be needed only at offices engaged later in the connection. Either all of these figures or the latter part mentioned must be repeated (re-transmitted) from each to the next engaged register during the process of establishing the connection.

With *end-to-end signalling,* the register at a transit office accepts only that few of the initial digits of the called number that indicate the direction to be selected at that transit office. After the necessary number of figures have been received at the register of the transit office and the call has been extended in the wanted direction, this register will be released immediately. The numerical information required at the next office will be transmitted from a "leading" register at a preceding office, which will remain attached until the connection is completed. This leading register may be either at the office nearest to the calling subscriber or be located for example at a toll center acting as a pivotal point for a number of local areas.

2.6 Leading Registers

By a leading register is meant a register that is located relatively near the calling subscriber and that remains connected to control the connection process until the called line has been selected. This register will receive the complete numerical information identifying the called subscriber and by means of significant backward signals it will be ordered to transmit any part of this information to any other office that may be involved in the establishment of the connection.

For certain connections, the leading register will be the local register at the office to which the calling subscriber is connected. This is notably the case for calls that remain within the local area or for calls to other areas in which no office of higher order will be involved.

In other cases, the leading register may be located at the toll office serving one or a number of local areas. In such a case, the originating local register in the office to which the calling

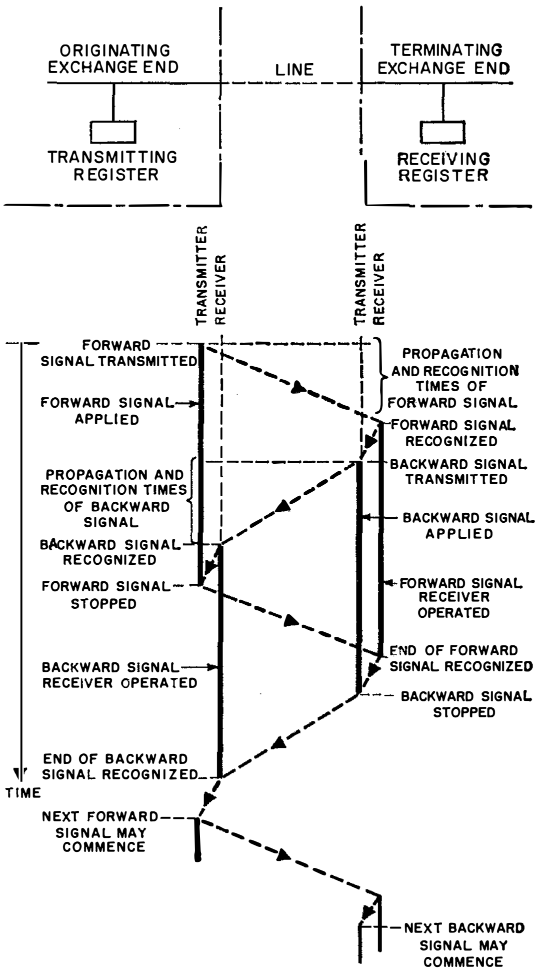


Figure 1—Principles of compelled signalling.

subscriber is connected will transmit all numerical information to the leading register in the toll office, so that all operations that are required for establishing a connection through a toll network may be controlled by the latter register. It is therefore unnecessary that the registers at the local offices be loaded with all the complications required for the establishment of toll connections.

For international connections, the leading register may be located at other offices that are of a higher order than the toll office serving the calling subscriber. Such so-called international registers may be located at regional centers or at one or more international toll offices. Again, the purpose of concentrating these registers at a few offices of higher order is to simplify the registers of lower order, so that the complications for international connections will be concentrated at a relatively small number of registers.

In this case also, all numerical information will be transferred from the originating local register to the leading register at the regional or international toll office. It will be evident that for this transfer of numerical information it may be necessary to transmit the signals through one or more transit offices and that end-to-end signalling may take place from the register at the originating local office to the leading register. For the establishment of the connection beyond the office of the leading register, the latter register takes control and the signalling may again be an end-to-end operation starting from the leading register.

It is not mandatory to employ end-to-end register signalling in the strict sense of the word between the first and last offices engaged in a long-distance connection. Such connections may be made in 2 or more sections with end-to-end signalling taking place within each section. Thus, each section may contain one or more transit offices the registers of which receive only a part of the numerical information and are released immediately after the connection has been extended to the next office.

2.7 Advantages of End-to-End Signalling

The advantages of end-to-end signalling when applied in the manner described in Section 2.6 are as follows.

2.7.1 The toll transit registers will be few in number because of their short holding time and of a simple type as they need receive only a restricted number of digits and need not retransmit any digits. If the transit traffic is small, the functions of the toll transit registers may be combined with those of the terminating registers in an economical manner.

2.7.2 The total number of operations for the transfer of numerical information in a connection with end-to-end signalling involving one or more toll transit exchanges, will be less than in a system with link-by-link signalling. For example, assuming a connection including two toll transit offices for which the complete called number comprises 10 digits, of which only 3 are required for completing the connection at a toll transit office, the total number of digits that must be transmitted with link-by-link signalling is 10 on the first two links and 7 on the last link, or 27 in all; with end-to-end signalling, this number is 3 on the first 2 links and 7 on the last link, or 13 in total. This will tend to increase the speed of operation and to reduce the error liability.

2.7.3 With end-to-end signalling, the leading register always remains connected until the connection has been completed. On receipt of a signal indicating that the call cannot be completed because of the state of the called line, the leading register will immediately release that part of the connection beyond its own office. Another possibility is to use the signal indicating the free condition of the called line for the connection of ringing tone to the caller from the originating office, or from the office at which the leading register is located, when the called office provides a type of ringing tone to which the caller is not accustomed, for example, on international connections.

2.7.4 Similarly as for 2.7.3, the leading register is available for receiving signals indicating congestion or any need for alternative routing. It may from the absence or presence of the latter signal determine the route of the call, which is particularly important in international connections if revenues are to be shared by different administrations. It also permits the leading register to receive reorder signals so that a different route may be taken from the office at which the leading register is located, in case of congestion at a further transit office through which the call was first routed.

2.7.5 The waiting time between the moment the last digit has been dialled and the moment the called line has been reached may be reduced by starting selection at the leading register before all numerical information has been received. With link-by-link signalling, this would not be possible without holding all transit registers engaged in the connection until all numerical information had been received and retransmitted, which would very considerably increase their holding time. With end-to-end signalling, it is sufficient to wait only until enough digits have been received to permit the transit registers to complete their operation and to release immediately thereafter.

2.7.6 End-to-end signalling facilitates operation under the following conditions.

2.7.6.1 In case of nonuniform numbering of subscribers within local areas. Subscribers numbers with either 5 or 6 digits are sometimes employed in one area or even in one office.

2.7.6.2 For the interworking of local areas having a different number of digits for their subscribers numbers. Various areas in one national network may have local subscribers numbers ranging from 3 to 7 digits. Also, the total number of digits to be dialled for a national toll call may vary for different areas.

2.7.6.3 If provision has to be made for dialling into private automatic branch exchanges by supplementary digits, resulting in a total number of digits larger than that for regular subscribers numbers.

All these cases present the problem of how to determine when all digits have been received at the originating and transit registers and when these registers have completed their functions, so that they may disconnect. This holds true particularly in the case of automatic service, because subscribers have not the possibility afforded to operators to transmit an end-of-sending signal. The automatic equipment must determine itself when the numerical information is complete.

In the cases of irregular numbering mentioned above, it becomes very complicated, if not impossible, to do this at the originating (leading) or transit registers, because in principle these registers must control the operation to many areas, each of which may have a different numbering pattern. It is, however, relatively easy, and in many cases quite simple, to determine the moment the full selection information has been received at the terminating area, because either the switching equipment serving this area may already provide a signal for this purpose from the final selection stage to the terminating register, or number discriminators may be associated with these registers that are able, in a rather inexpensive and rapid way, to determine the number of digits required for completing the connection within the terminating area alone.

It may now be seen that end-to-end signalling presents an easy and straightforward solution to the problem mentioned by a combination of the following circumstances.

Firstly, with end-to-end signalling, transit registers need not receive and retransmit the complete numerical information and may release immediately after having received a few initial digits and after having completed the required selection at that transit office. There is, therefore, no problem of indicating the receipt of all numerical information or of the completion of the selection, as far as the transit registers are concerned.

Secondly, end-to-end signalling is based on the condition that a leading register remains connected until the connection has been completed and that backward signals may be transmitted to this register at any time until this moment.

Thirdly, the fact that the receipt of the last numerical information or the completion of selection may conveniently be determined at the terminating office, leads to the concept of using backward signals from the terminating to the leading register to signal this fact. The leading registers will not need to determine themselves that no more numerical information is to be expected.

Fourthly, in case the leading register is preceded by another part of the connection controlled by the originating register, as explained in Section 2.6, the leading register may retransmit the backward signal it receives to the originating register.

It should be observed that this backward signal may contain further information, such as on the state of the called line, as already explained under 2.7.3.

2.8 Signals Employed

The meanings of the accepted signals are shown in Tables 3 and 4 for the forward and backward register signals, respectively. The following may serve as an explanation of these tables.

2.8.1 *Signals in Forward Direction*

Table 3 shows that each of the 15 signals available in the forward direction may have two different meanings as given in groups *I* and *II*, the normal meaning being that for group *I*. Group *II* will become effective on receipt of a particular backward signal, as will be explained later.

2.8.1.1 *Group I*

The first 10 signal numbers in group *I* are allotted to the 10 numerical digits, and will in general be used for the transmission of numerical information such as that in the called number. Signal *11* is used only for semiautomatic international operation where it may be necessary for an outgoing international operator to select an operator at the distant country, for example, for handling delayed or person-to-person calls. This signal may be followed by other numerical signals indicating the particular operator's position wanted. Such additional numerical indications may be omitted if any free position is acceptable. In either case, the numerical information is always followed by the transmission of signal *15,* which indicates that no further digits are to be accepted. Signal *15* will be transmitted by the operator who will depress an end-of-sending key for this purpose. It will be evident that subscribers who are permitted to establish automatic connections via the same international lines that are accessible to operators cannot reach the operator's positions for delayed or person-to-person calls because subscribers cannot make use of code signals *11* and *15*. The operation as here described corresponds to that covered by signals *11* and *12* in the recommendations of the Comité Consultatif International Télégraphique et Téléphonique for semiautomatic international operation.

Signal *12* is transmitted in the forward direction from a register when it receives a backward signal from a distant office indicating a request that cannot be complied with. This may be the case, for example, when the distant office sends a backward signal asking the originating or leading register to transmit the identity or the category of the calling subscriber and the automatic equipment preceding this originating or leading register is not able to furnish this information. Other cases in which forward signal *12* may be used will be discussed later.

TABLE 3

LIST OF FORWARD REGISTER SIGNALS

Signal Number	Group I	Group II, Category of Caller (Responding to A-3 and A-5)	
1	digit 1	Subscriber	For national service (2)
2	digit 2	Subscriber with preference	
3	digit 3	Maintenance equipment	
4	digit 4	Reserve	
5	digit 5	Operator's position	
6	digit 6	Data transmission	
7	digit 7	Subscriber	For international service
8	digit 8	Reserve	
9	digit 9	Reserve	
10	digit 0	Operator's position	
11	Access to operator's positions (1)	Spare for optional use	For national service (3)
12	Demand rejected (responding to A-9, A-10, or A-13)	Spare for optional use	
13	Access to maintenance equipment (1)	Spare for optional use	
14	Insert echo suppressor	Spare for optional use	
15	End of sending	Spare for optional use	

(1) These signals may be followed by other numerical signals and by signal *I-15* or by *I-15* alone.
(2) The international outgoing registers must convert signals *II-1* to *II-4*, *II-6* into *II-7*, and *II-5* into *II-10*.
(3) These signals will be converted by international outgoing registers into signal *II-7*.

TABLE 4

LIST OF BACKWARD REGISTER SIGNALS

Signal Number	Group *A*	Group *B*
1	Send next digit	Prepare for malicious call tracing
2	Send next digit but one (4)	Changed number
3	Switch over to *B*-signals (1)	Called line busy
4	Congestion (4)	Congestion
5	Transmit category of caller (1)	Dead line or dead level
6	Switch through speech path	Called line free, with call charging
7	Send last digit but two	Called line free, no call charging
8	Send last digit but three	Called line out-of-order
9	Spare for optional national use	Spare for optional national use (3)
10	Spare for optional national use	Spare for optional national use (3)
11	Send first digit of international country code	Reserve
12	Send first digit of national number	Reserve
13	Identification: send first digit of international country code (2)	Reserve
14	Inform whether or not echo suppressor is needed	Reserve
15	Reserve	Reserve

(1) Must be responded to by a forward signal of group *II*.
(2) Repeat signal *A-13* = send next digit of calling line's number.
(3) On international connections, signals *B-9* and *B-10* must initiate the connection of an information tone to the caller, similarly to signal *B-5*.
(4) On lines from outlying offices preceding leading registers and on which 4 frequencies for *A* signals without *B* signals are employed, signal *A-4* may initiate the connection of information tone and signal *A-2* may be used to initiate the transmission of a predetermined digit.

Signal *13* is a numerical signal that can be transmitted only from test equipment; it initiates the selection of other test equipment at the distant end so that the transmission conditions of the connection may be tested. If necessary, this signal may be followed by further numerical signals to indicate which of several different types of test equipment should be selected.

The only forward signal of group *I* not discussed so far is *14,* which has been provided in case it is necessary to call for the insertion of echo suppressors. The exact manner in which this signal and the corresponding backward signal will be used has not yet been determined.

2.8.1.2 *Group II*

All the forward signals of group *II* are concerned with the particular category of a caller. The first 6 signals in this group have been reserved for national use and identify 5 different categories of callers with one signal in reserve. These forward signals will not be transmitted on international connections. If a request for information on the category of a caller is required on an international connection, one of signals *7* through *10* will be transmitted. The leading register controlling international connections is able to distinguish the fact that an international connection is concerned and will, therefore, send the appropriate signal. This leading register must obtain the necessary information on the category of the caller from the register at the originating office and this latter register will be able to send only one of the signals *1* to *6*. On receiving this information from the originating register, the leading register will convert signals *1, 2, 3, 4,* and *6* into the international signal *7,* and will convert signal *5* into international signal *10*. Signals *11* to *15* have not been assigned but are reserved for only national service and may be employed by each administration for such purposes as may be required in its country.

A few words may be said here regarding the need for providing these category-of-caller signals. A distinction between calls originating from *subscribers* and from *operators* may be needed for various reasons. In certain national networks an operator, by sending offering signals, may through-connect to a busy called line to inform the busy subscriber that a national or international toll call is in preparation for him. In such cases, it is, therefore, necessary that a connection to a busy subscriber's line be not broken down by the leading register if it receives a backward signal indicating the busy condition. In order that subscribers or other unauthorized callers will not be able to effect the offering operations, it is useful to receive an indication at the terminating office that offering should be permitted. Accordingly, when a called line is found busy, a backward signal will be transmitted initiating the transmission of the forward signal indicating the category of the caller and only if this indicates an operator's call will the terminating equipment respond to offering signals. It will not respond when such offering signals have been imitated by subscribers. Another reason for distinguishing between operators and subscribers is that in cases of overload or congestion, arrangements may be provided so that preference is given to operator's calls, and a congestion signal is always given to subscribers even if free outlets become momentarily available.

Arrangements are provided also to signal *two different classes of the calling subscribers,* of which only those having preference will be able, in cases of emergency, to obtain certain connections.

A further category-of-caller signal has been provided to indicate calls originating from *subscriber stations at which data-transmission equipment is connected.* Such a signal may be needed in the future to apply the proper tariffs for data-transmission connections.

On international connections, a distinction is provided only between operator's and subscriber's calls.

2.8.1.3 *Optional Extension of Forward Signals*

Apart from the signals indicated in Table 3, administrations are free to provide in their national network additional forward signals not in this list. Thus the forward signals of group I may receive additional meanings on receipt of a particular backward signal. For example, in certain countries it may be necessary to transmit the identity of the calling subscriber to recording equipment for automatic message accounting concentrated at certain toll offices. Toll calls on which information for automatic message accounting is to be provided, will cause the transmission of a backward signal that changes the meaning of the forward signals in such a manner, for example, that signals *1* to *10* no longer signify the digits of the called subscriber's number but those of the calling subscriber's number. In case such a request for identification of the calling subscriber's number is required at originating offices where this request cannot be complied with (for example, when no automatic identification equipment has been provided to determine the identity of the calling subscriber), this request may be answered with signal *12,* which in this case will indicate that no identification is possible, as a result of which such a toll call may be refused.

2.8.2 *Signals in Backward Direction*

From the above description of the forward signals, it will be obvious that a minimum of 10 signals is always needed; they require the use of combinations of a minimum of 5 frequencies in the forward direction. In most cases, as soon as one supplementary signal has to be provided, the use of the 6th frequency will be required.

Contrary to this condition, the number of signals in the backward direction does not always require the use of as many as 5 or 6 frequencies. In many cases, the necessary number of signal combinations may be provided by only 4 frequencies, such as when the number of backward signals does not exceed 6 combinations. Table 4 takes into account the desire to reduce the number of frequencies employed for backward signals wherever possible. The signals have been grouped so that either 4, 5, or 6 frequencies may be used according to the number of combinations required, which will be 6, 10, and 15, respectively, in these 3 cases. The first thing to be observed about the table for backward signals is, therefore, that it has been divided by broken lines into these 3 subgroups of signals.

Furthermore, in a manner similar to that for forward signals, Table 4 has two columns of different meanings for each of the signals from *1* to *15.* Signals indicating only one of these columns, or both, may be employed. This implies that several different possibilities exist, the simplest being to use only the 6 signals in the first subgroup for the meanings given in *A*. This simplest case may be extended in either of two ways, namely, by extending into subgroups 2 and 3 in the same vertical column or by adding signals to indicate a transfer to the meanings shown in the second column, group *B*. This same principle may be applied to the case of the 10 *A* signals shown in the first column. Supplementary signals in this case may be obtained either by adding the last 5 signals in group *A* or by making use of signals *1* to *10* in group *B*. It will, therefore, be seen that a large number of combinations of signals is possible that are all compatible with one another, so that a restricted number of signals may be used in part of the offices and a more-extended number in other offices.

The number of signals that may be employed at any one office is determined by the characteristic operating features of the local switching equipment. For the simplest possible type of local switching equipment, which does not provide any electric criterion on the condition of the called line, it is sufficient to provide only signals *1* to *6* of group *A,* which even then do not all need to be used. Local switching equipment that is arranged to provide a restricted number of different electric signals to indicate the condition of the called line, may also employ the first group of 6 signals by adding the second meanings shown for group *B* of these signals. Additional features may be provided in each of these two cases by making use of 10 signals, as will be explained hereafter.

Signals *11* through *15* appear only in group *A* and have been reserved for use on international connections exclusively, so that normally none of the registers in local switching equipments will have to provide for transmitting or receiving these signals.

The purpose of each of the signals shown in Table 4 will now be explained by describing how some typical connections are established. It will be evident that the backward signals are very important because they exercise a necessary control on the operations to be performed.

3. Typical Cases of Operation

3.1 Interoffice Connection in Local Multioffice Area

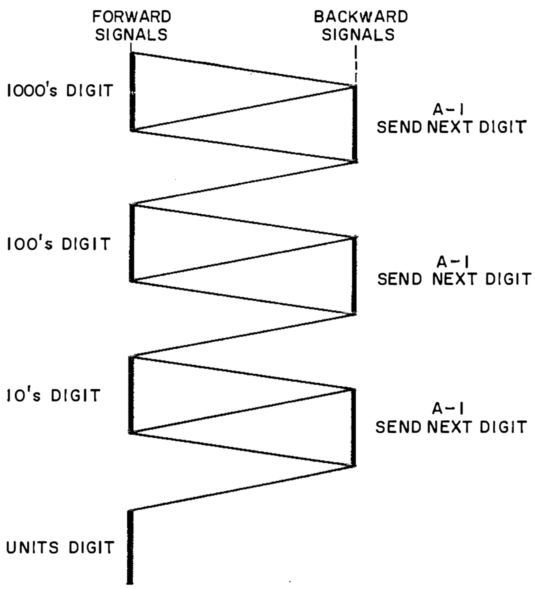


Figure 2—Transmission of 4 digits from one local exchange to another.

3.1.1 *Direct Connection between Two Local Offices*

The subscribers' numbers used in a local area have an initial part identifying the local office to which the subscriber is connected and a second part identifying each particular subscriber of this office. With offices having the usual 10 000-line capacity, this latter part consists of 4 digits. The initial part, which is the office code, may have from 1 to 3 digits depending on the size of the area. In the simplest case, when a direct junction interconnects the two offices concerned in the connection, the selection at the originating office is controlled by the office code; only the last 4 digits of the subscriber's number need be transferred to the wanted office. The operation in this case is shown in Figure 2.

As soon as the junction to the wanted office has been seized, a signal is transmitted thereon which initiates the connection to the incoming end of this junction of an incoming register at the wanted office. Concurrently, the register at the originating office will connect to the junction the combination of frequencies indicating the value of the thousands figure, and this signal will be extended to the incoming register as soon as it is connected. On recognizing this first digit, the incoming register transmits to the originating register backward signal *1* of

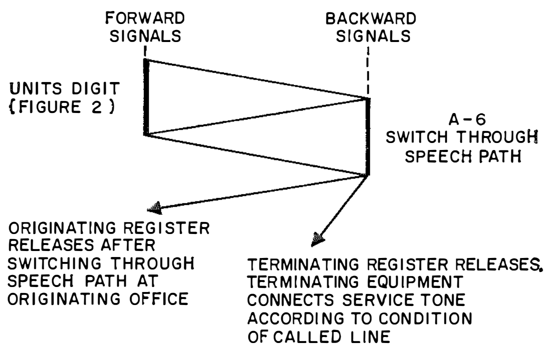


Figure 3—End of selection without signalling condition of called line.

group *A* (signal *A-1*), which has the dual purposes of informing the originating register that the first digit has been properly received, which causes it to stop transmission of the first digit, and also of signalling to the originating register that the next operation required is the transmission of the following digit. The termination of the forward signal in its turn, by being recognized at the incoming register, causes the backward signal to stop. Recognition of this at the originating register permits it to commence transmitting the next digit. It will be obvious that this transmission will take place only when this digit has already been dialled by the calling subscriber. In this manner, all 4 digits of the number wanted at the distant exchange will be transmitted by mutual control of forward and backward signals, and each time the next digit must be transmitted, the forward signal will be acknowledged by the transmission of the backward signal *A-1*.

This situation changes at the moment the units digit has been transmitted to and is recognized at the incoming register, and different operations may now take place according to the possibilities provided at the terminating office. The simplest case is when the terminating office is *not capable of giving any indication of the condition of the called line.* As shown in Figure 3, recognition by the incoming register of the units digit will immediately be acknowledged by signal *A-6,* which causes the originating register to release immediately and the speech path to be closed at the originating office. Also, the register at the terminating office will disconnect and the incoming junction will be connected through to the selector equipment so that the service tone indicating the condition of the called line (ringing, busy, or any other tone) may be transmitted to the calling subscriber from the terminating office.

A somewhat-more-complicated condition exists when the terminating equipment is capable of *providing electric criteria on the free and busy conditions of the called line.* With reference to Figure 4, the units digit will be acknowledged by the transmission of the backward signal *A-3,* instructing both originating and incoming registers to use the meanings given in group *II* of Table 3 and group *B* of Table 4. When the signal *A-3* has stopped, the originating register proceeds immediately with the transmission of one of the signals shown in group *II* of Table 3, depending on the origin of the call. In this case, where a call is assumed to be originated by a subscriber, either signal *II-1* or *II-2* will be transmitted. Assuming that it will be possible within a very short interval after the units figure has been received to select the called line and determine its condition, the forward signal

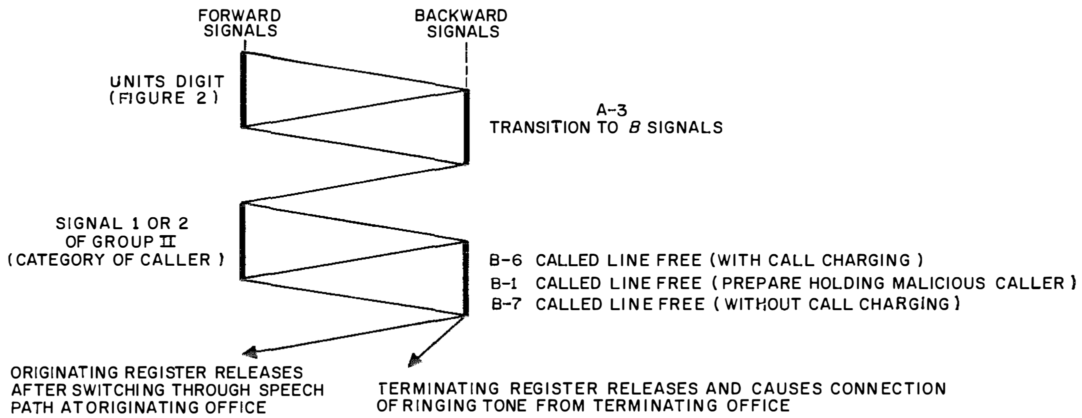


Figure 4—End of selection when called line is free.

II-1 or *II-2* may within a very short time be acknowledged by one of the signals in group *B* of Table 4, indicating the condition in which the called line has been found. If the called line is free and answering should cause the call to be charged, signal *B-6* must be transmitted. Both originating and terminating registers will disconnect on termination of the *B-6* signal, and the switching equipment will connect the speech path through at both offices so the calling subscriber may hear the ringing tone from the terminating office. Signals *B-1* and *B-7* have this same effect but initiate supplementary operations at the originating office, signal *B-1* causing the equipment at this office to prepare for holding the connection under the control of the called subscriber to permit the tracing of a malicious caller, and signal *B-7* operates a relay to prevent the call from being charged when the answer signal is received.

Signal *B-3* will be transmitted when the line of the called subscriber is busy, as indicated in Figure 5. In this case, the switching equipment may be arranged so that on termination of the signal the originating register causes busy tone to be connected to the calling subscriber and releases the junction to the distant office, as a consequence of which all equipment engaged at the distant office will be set free. Signals *B-2, B-5, B-8, B-9,* and *B-10* are available for other conditions of the called line that do not permit the connection to be completed to the called subscriber. All of these signals may cause the register at the originating office to set free the connection to the distant office and connect to the calling subscriber either a special tone (so-called information tone) or a recorded message to inform him why the connection cannot be completed.

If the connection cannot be completed because of congestion in the switching equipment at the terminating office, the incoming register will transmit the signal *B-4,* congestion, to acknowledge one of the forward signals of group *II* that is transmitted in response to signal *A-3* as shown in Figure 6. This causes the originating register to release the junction to the distant office and to connect an appropriate tone signal to the caller.

If the terminating selector equipment is of a type in which *a few seconds may elapse before the called line may be reached* after the receipt of the units figure, it is not desirable to ac-

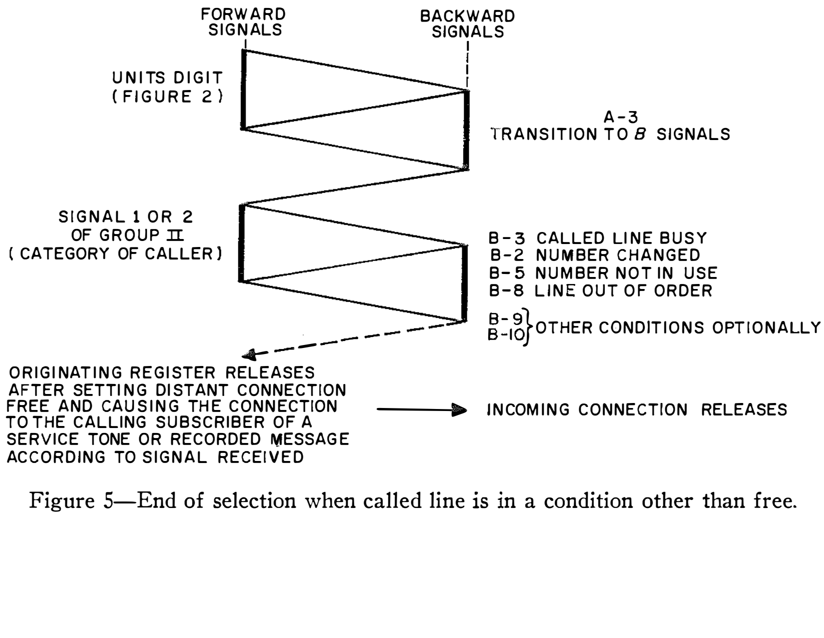


Figure 5—End of selection when called line is in a condition other than free.

knowledge the units figure by signal *A-3* as described above, because this would be immediately responded to by the transmission of the signal *II-1* or *II-2,* which would have to remain connected until it can be acknowledged by a *B* signal after the completion of selection. Although this is not objectionable in a local connection as now considered, it becomes objectionable when a connection is extended via a toll line employing carrier equipment in which the amount of signalling energy should be made the minimum possible. For this reason, the units digit will not be acknowledged by either signal *A-3* or *A-6* but by signal *A-1,* which invites the originating register to transmit the next digit. See Figure 7. Since, however, no next digit is available, no forward signal will be transmitted during the period the terminating selector equipment needs to complete the selection. After selection and the condition of the called line has been determined, further operations are then initiated by the incoming equipment, which will send a pulse of predetermined duration. If this pulse is of signal *A-3,* the operation might be continued in the same manner as described above for the cases in which the backward signal *A-3* was transmitted immediately on receipt of the units digit. In this case, however, it would be necessary, when the called line is found free, to transmit two cycles of signals before this free condition can be signalled, namely, firstly by the cycle commencing with backward signal *A-3* and secondly by the cycle consisting of a forward signal of group *II* and the backward signal *B-6*. This may be objectionable, because for a quick reply, the signalling operations

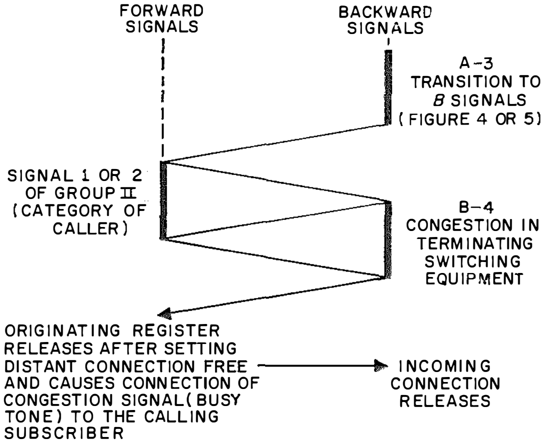


Figure 6—Congestion in terminating exchange switching equipment.

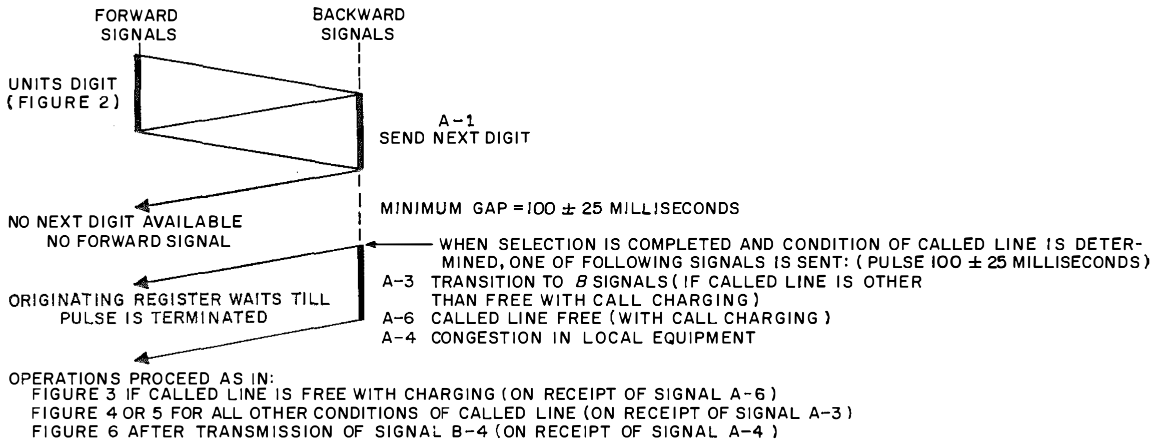


Figure 7—Suspension of signalling after units digit if selection is delayed.

would not be completed before the called subscriber has challenged the caller. Therefore, when it can be determined that the line is free, the free condition in this case will not be signalled by signal *B-6,* but by a pulse of signal *A-6,* which is then transmitted instead of the pulse of signal *A-3* referred to above. In this way, it is ensured that the speech path may be closed with the minimum of time after the free condition of the called subscriber has been determined. In the case of congestion in the terminating switching equipment, a pulse of signal *A-4* will be transmitted instead of *A-3,* signal *A-4* having the same significance as signal *B-4.*

3.1.1.1 *Interoffice Junctions Providing Access to 2 or More Units of 10 000 Lines*

It is obvious that when two or more office units of 10 000 lines each are located in the same building and are served by common groups of junctions, the originating register, on seizing a junction to the wanted office, will first have to transmit the ten-thousands digit to the incoming register. All other operations as described in the former case remain the same.

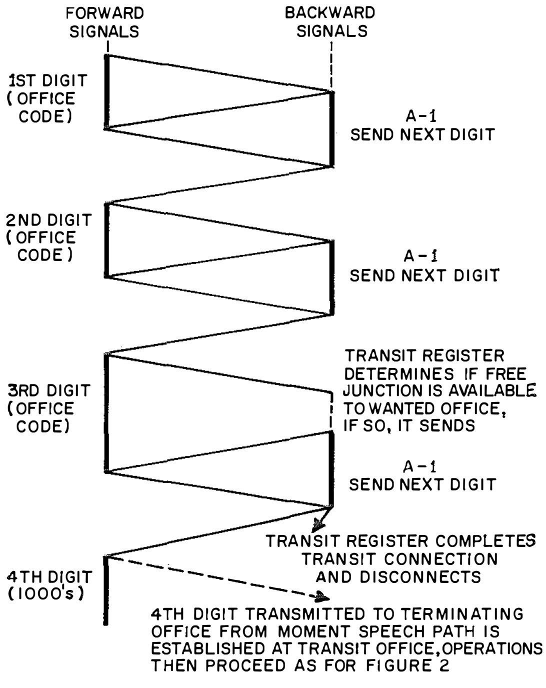


Figure 8—Operation at transit office if junction to the wanted office is available.

3.1.2 *Transit Connection in Local Area*

When two offices in the same local area are interconnected via a transit office, the operations are as follows.

The register at the originating office will determine from the office-code part of the called subscriber's number that the connection must be routed through a transit office and will cause the selection of a free junction to that office. It will then proceed by transmitting the first digit of the called subscriber's number, as shown in Figure 8. The register at the transit office will accept this first digit and by successive transmissions of backward signal *A-1* it will consecutively initiate the transmission of those digits from the originating register that are required at the transit office to determine the wanted office. The transit register will, before acknowledging the last of these digits, determine if a free junction to the wanted office is

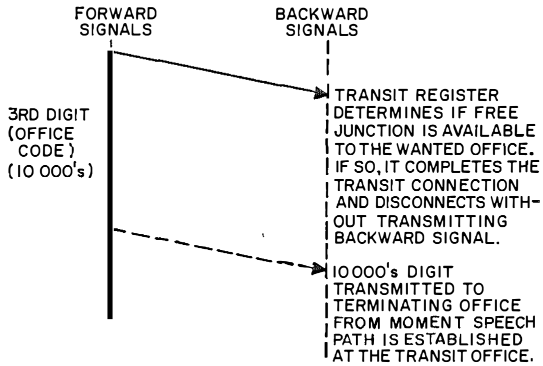


Figure 9—Modified operation of Figure 8 if the last digit of the office code is required as the first digit at the terminating office.

available. If so, it will acknowledge this digit with signal *A-1,* which will cause the originating register to prepare for transmitting the thousands figure. This will be extended to the wanted office as soon as the selection at the transit office has been completed. The register at the transit office will disconnect after it has connected the speech path through at the transit office and on having recognized the termination of the transmission of the last digit of the office code, so that this cannot be switched through to the terminating office. Figure 8 shows the case of an office code consisting of 3 digits.

In case the terminating office is connected via a group of junctions providing access to more than 10 000 lines, it will be necessary to transmit the ten-thousands figure to this office. If this ten-thousands figure had already been received at the transit office because it was required at this office to determine the identity of the wanted office, the receipt of the ten-thousands figure at the transit office is not acknowledged by any backward signal, so that the transmission of this figure from the originating office continues and it will be extended to the terminating office as soon as the connection is switched through at the transit office. It is obvious that in this case the transit register will switch through without awaiting the termination of the ten-thousands digit, as in Figure 9.

3.1.3 *Alternative Routing from Originating Office*

Assuming the originating and wanted offices are interconnected by a group of direct junctions but that these junctions are fully occupied, the connection may be routed alternatively via a transit office.

The originating register will first attempt to select a free junction in the direct route. If successful, it will transmit the first digit of that part of the subscriber's number that is required at the terminating office (see Figure 2). If no free direct junctions are available, the register will attempt to select a free junction to the transit office. When successful, it will transmit the first digit of the called subscriber's number to this office as explained above for operation through a transit office (see Figure 8). Also all further operations will be as there described.

3.1.4 *Alternative Routing from First Transit Office through Second Transit Office*

The operations in this case will commence in the same way as described for the case of a call through a single transit office (Figure 8) up to the point where the register at a first transit office has received the last digit of the office code but before this has been acknowledged by a backward signal. The transit register will first attempt to select a direct junction to the wanted office and, if this is not available, it will next attempt to select an alternative route through a second transit office. When extending the connection to another transit office, the register at the first transit office has to ensure that the register at the second transit office receives the complete office code to determine the identity of the wanted office. To do this, the register at the first transit office will acknowledge the last digit of the office code by one of the signals, *A-2, A-7,* or *A-8,* depending on whether the office code characterizing the wanted office is of 2, 3, or 4 digits as shown in Figure 10. The transmission of one of these backward signals will cause the originating register to respond with the transmission of the first digit dialled by the calling subscriber and this is now transmitted via the first transit office to the register at the next transit office, which acts in exactly the same way as described for the case of a single transit office (see Figure 8).

3.1.5 *Congestion at Transit Office—Rerouting*

In case a connection cannot be completed at a transit office because of the unavailability of junctions both directly to the wanted office and to any other transit office through which the connection might alternatively be routed, the last digit of the office code will be acknowledged by signal *A-4*.

Receipt of this signal will cause the originating register to break down the connection to the transit office. Next, it will either cause the connection of an appropriate tone to the caller to inform him of the congestion condition or attempt to set up the connection from the originating office via a different route if such possibility exists. This is shown in Figure 11.

3.1.6 *Interworking between Offices with Different Operating Facilities*

The case may be assumed now that in a multi-office area some offices of older design are not able to provide or respond to electric signals indicating the condition of the called line and that others of more modern design incorporate this possibility. At the first-mentioned offices, the incoming registers associated with the incoming junctions from the latter type of offices will always acknowledge the receipt of the units figure with signal *A-6* as in Figure 3 independent of the called line's condition. This signal will at modern offices cause the disconnection of the originating register and the through connection of the speech path, so that the caller will hear the service tone (ringing, busy, et cetera) transmitted from the switching equipment at the terminating office.

On the other hand, the incoming registers at offices capable of providing an electric signal on the condition of the called line will normally transmit a signal *A-3* (transition to *B* signals and demand to transmit category of caller) to acknowledge the receipt of the units digit (Figures 4 and 5). When this signal is received at the originating register of an office that cannot respond appropriately to the various conditions signalled by the *B* signals and is also unable to provide a signal indicating the category of the caller, this register will in response to signal *A-3* transmit the forward signal *I-12,* which is a rejection of the demand for information.

As shown in Figure 12, this signal will be acknowledged by transmitting from the terminating register the signal *A-6,* which will act in the normal way to establish the speech path. The equipment at the terminating office will therefore not be released, even if the condition of the called line (busy, et cetera) prevents completion of the connection to the called subscriber, and the terminating equipment must be arranged to supply a service tone to signal this condition to the caller.

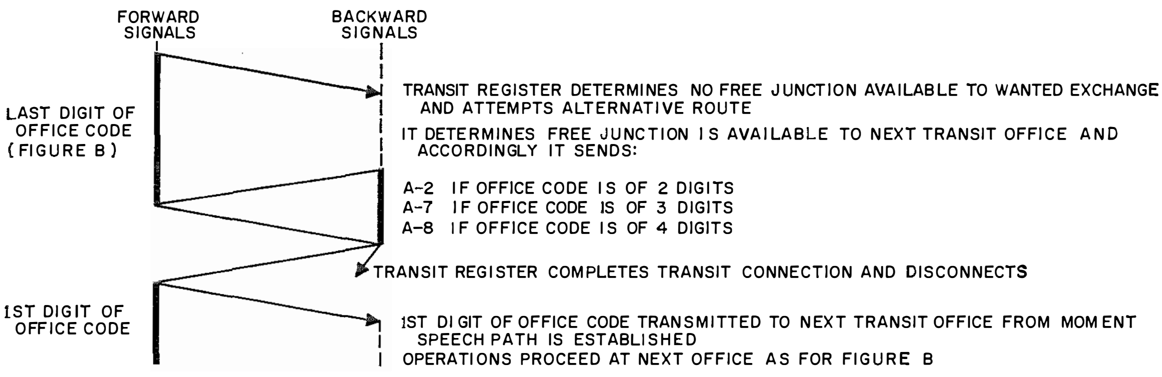


Figure 10—Alternative routing at transit office.

3.2 Operation in a National Toll Network

The methods of operation employed in establishing national toll calls are very similar to those described in conjunction with the operation between offices in a local network. The principal difference resides in the fact that the selection process in the toll network will be governed by the digits constituting the national toll code, instead of by the digits constituting the office code as described for the operation in local multioffice areas. In other words, the possibility is provided by this system to use identical methods of operation and identical signals for the transmission of numerical information on any type of connection within a national network.

3.2.1 *National Networks with Closed and Open Numbering of Network Sections*

A few words should first be said on the different methods used in various national networks to divide the total area into sections from a point of view of numbering subscribers. These various methods of numbering will influence to a certain extent the signals required for the establishment of connections between different local areas. Because the terminology used on this subject is not identical in various countries, a definition of the terminology used here has been given in the appendix, Section 5.2.

All national networks are characterized by different orders or ranks of switching centers used to establish connections among local areas. The lowest rank is that serving one local area or a few small localities considered as one area from the point of view of numbering. These switching centers will be called "toll end offices."

Furthermore, all national networks are characterized by the existence of a higher rank of switching center, which may be considered as the pivotal point of a network group, through which all traffic (with some exceptions) of this network group will be routed to and from other

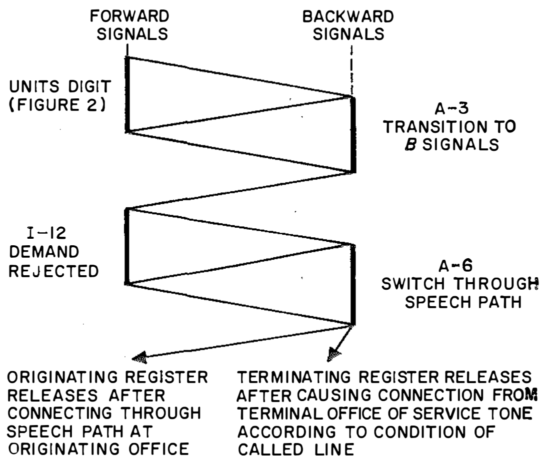


Figure 11—Congestion at transit office.

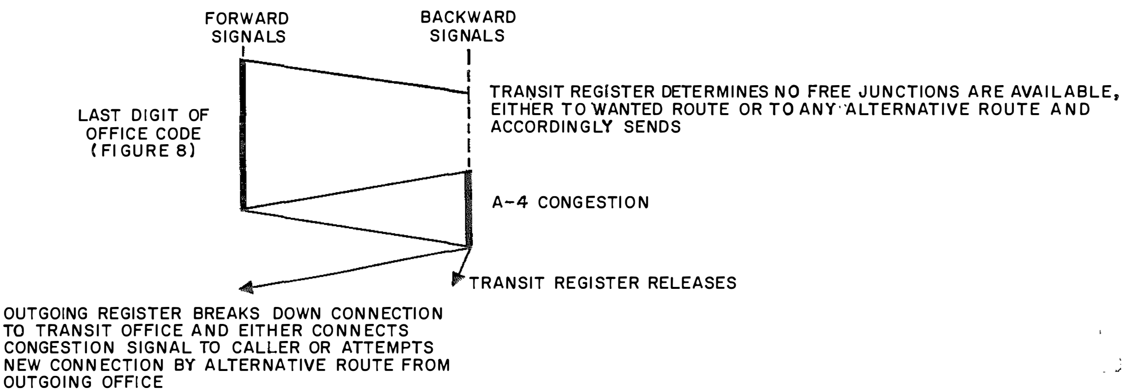


Figure 12—Procedure if the originating exchange cannot accept *B* signals.

similar network groups. In medium- and small-size countries, these pivotal switching centers constitute the highest rank of switching centers and many or all of them are interconnected by direct toll lines. The switching centers referred to here will be called "primary toll offices." (Very large countries will be divided into regions in which one of the primary toll offices acts as a regional center, all regional centers being directly interconnected, but the situation does not differ essentially from that in small- or medium-size countries.)

The network group served by a primary toll office is designated in some countries as a district or zone. In the following, such a network group will be called a "primary toll area." Its network in principle is of the star type with one or two intermediate ranks of switching centers between the primary toll office and the distant toll end offices located within the primary toll area.

In descending order, these switching centers will be called "secondary toll office" and "tertiary toll office" and the areas served by them "secondary toll area" and "tertiary toll area." The network within each secondary toll area and tertiary toll area (if existing) is again in principle of the star type, the toll end offices constituting the outer periphery of the integrated star-shaped network.

Evidently, offices of lower rank will be incorporated in each office of higher rank to serve the lower rank of area directly associated with each of these offices of higher rank.

From the point of view of numbering, there is a difference between some countries and others as concerns the area over which subscribers may obtain connection with one another by the use of only the local calling number, that is, the number used for connections within the local area.

A primary toll area is said to have open numbering when the use of local numbers is restricted to calls within each local area. In such a case, for establishing a connection between any two different local areas, even if they are within the same secondary or tertiary toll area, the local calling number of the wanted subscriber must be preceded by dialling the national "toll code" of the wanted toll end office, and each toll end office will have a distinct toll code.

A consequence of the use of open numbering is that the number of digits of the local calling number is determined independently for each local area according to the numbering capacity required for this area alone.

A primary toll area is said to have closed numbering when the use of local numbers is extended to calls for all subscribers within this area. This means that all subscribers within the primary toll area share a common block of numbers that will be divided among all local offices within the area according to the number of subscribers they have to serve or will be expected to serve in the future. Usually, all subscribers' numbers within this "closed numbering area" have an equal number of digits determined by the total numbering capacity required for the whole of the area, although the possibility of using nonuniform numbers is not excluded. In the case now considered, it will be evident that local calling numbers must be dialled for all calls completed within each primary toll area and that this number should be preceded by a national toll code only for calls between subscribers located in different primary toll areas. It further follows that only one toll code is needed for each primary toll area and these toll codes are usually assigned to the primary toll offices to identify the primary toll area served by them.

In the following sections (3.2.2 through 3.2.4), it will be shown that this difference in numbering influences the signals employed for making connections between different local areas.

In some countries, a numbering plan is employed that may be considered to be somewhat intermediary between the two cases described above, in that connection may be obtained between adjacent primary toll areas by using the local number preceded by only the last or the last two digits of the toll code instead of by the whole toll code. This is not further considered, as it may be regarded as a special case of the closed numbering plan referred to above.

3.2.2 *Connections within Primary Toll Area*

3.2.2.1 *Closed Numbering*

With the use of closed numbering for a primary toll area, the whole of this area may be considered from the point of view of switching and signalling as one large multioffice local network. As all connections employ only the local calling number, the operations proceed exactly as described in Sections 3.1.1 through 3.1.6 for a local multioffice area, and will normally take place under the control of leading registers located at the toll end office associated with the originating local area.

In some cases, however, it may be considered propitious to provide leading registers at the primary toll office or at a secondary or tertiary toll office on which the originating office depends for outgoing traffic. For connections switched through one of these offices of higher rank, the leading register will take control of that part of the connection at and beyond the office at which it is located.

This method of operation may be found particularly desirable to concentrate at the toll office, at which these leading registers are located, the equipment required for determining the tariff and for applying call charging on calls leaving these toll end offices.

It should be noted that in connection with primary toll areas with closed numbering as now considered, a particular terminology is in use and is defined in the terms used here under a separate heading in the appendix, Section 5.2.

3.2.2.2 *Open Numbering*

For any call leaving the local area, the calling subscriber must commence by dialling the national toll code of the wanted end office, comprising the national access code and a significant number identifying the wanted toll end office.

The national *access code* is used to obtain access from the originating local office to the *toll end office serving the originating local area.* In large local areas, the toll office may be in a separate building and reached from the originating local office either directly or via a local transit office. Any selection operation will be controlled by the toll access digit(s), using the principles of operation described for calls within a local area. In local areas served by a single local office, this may be located in the same building as, and be integrated with, the toll end office.

The significant number of the toll code is used to establish the connection to the wanted toll end office, which connection may be routed either via a direct toll line or via one or more of the primary, secondary, or tertiary toll offices (within the primary toll area, in the case now considered).

When a direct line is available to the wanted toll end office, the leading register controlling the establishment of the toll part of the connection (this will usually be the outgoing toll register located at the originating toll end office) recognizes this fact and will, having selected a line to the wanted end office, immediately transmit the first digit of the wanted subscriber's local number after which the operation will proceed exactly as described for a local call in Section 3.1.1.

When, however, the wanted toll end office is integrated with a toll office of higher rank, the selection of a direct line may not be followed by the transmission of the wanted local subscriber's number, as this line may have to serve also for connections to be switched in transit through this office.

In such a case, it is necessary to transmit to this office at least that part of the toll code that will permit its incoming registers to determine whether a terminating or a transit connection is involved. This number of digits varies with the numbering plan and with the rank of the toll office concerned.

The numbering plan will normally identify toll offices of higher rank by fewer digits of the toll code than those of lower rank; the number of digits usually increases by one for each rank of lower order.

In each case, the incoming register at the toll office to which connection is obtained will, by the successive transmission of signals *A-1,* call for as many significant digits of the toll code as is required to determine how the connection should be routed.

If the call terminates at the toll office concerned, the complete toll code must be received. After this, the transmission of the first digit of the local number is initiated by signal *A-1* and the connection proceeds as for a local interoffice connection (Section 3.1). Either the incoming toll register may perform the same functions as a local transit register in a local connection or it may delegate these functions to separate registers that are provided for handling terminating connections and are associated with outgoing junctions from the toll office to the local office equipment.

For a connection within the primary toll area that must be switched through one or more primary, secondary, or tertiary toll offices in this area, the operations at each of these offices preceding the primary toll office and including the latter will commence with the transmission of the first significant figure of the toll code followed by as many more requested digits as are necessary to determine the direction to be selected, which may be the terminating local area, another toll office of lower rank dependent on the office considered, another office of equal rank to which direct lines are provided, or an office of higher rank on which the office considered is dependent.

If the connection is directed to a toll office of higher rank, it is necessary that the leading register transmit to this office the information starting with the first of the significant digits of the toll code. This is brought about by acknowledging the last digit received at each office by signals *A-2, A-7,* or *A-8,* depending on whether 2, 3, or 4 digits were received.

In case the connection is directed to a toll office of equal or lower rank, it is usually unnecessary to transmit to this office all significant digits of the toll code; the digits identifying the area associated with the office of the next higher rank on which it is dependent may be omitted.

This again is brought about by the use of the appropriate proceed-to-send signals *A-1, A-2, A-7,* or *A-8* for the acknowledgment of the last digit received.

3.2.3 *Connections between Different Primary Toll Areas*

3.2.3.1 *Closed Numbering*

For calls leaving the primary toll area of the calling subscriber, the latter must commence by dialling the national toll code of the wanted primary toll area (or primary toll office). This code comprises one or two access digits and a significant number identifying the wanted primary toll office.

The *national access code* is used to obtain access from the originating local office to the *primary toll office serving the originating local area.* This part of the called number controls the connection through the toll end office serving the local area and if necessary through a secondary and a tertiary toll office to the primary toll office. At all of these offices, receipt of the toll access code alone will cause the selection of the shortest possible route to the primary toll office. Furthermore, this same numerical information will also be transmitted first to the incoming register at the primary toll office, where it serves to select the outgoing toll equipment, which comprises toll registers acting as leading registers for handling the connection on the toll network beyond the primary toll office.

It will be obvious that each of the offices involved in establishing the connection up to this point, by omitting the transmission of an acknowledgment signal if the access code is only one digit and by using signal *A-2* for acknowledgment of the second access digit if two access digits are employed, will cause the transmission of the access digit(s) in the manner described. The incoming register at the primary toll office is the last to receive this information and will release after it has caused the attachment of a leading toll register.

This procedure is employed so that the same switching and line equipment available for interconnecting the toll end offices with the primary toll office may serve both of the categories of connections dealt with here and in Section 3.2.2.1, which use different numerical information for controlling the connection.

The leading toll register at the primary toll office now initiates transmission from the originating register of all numerical information commencing with the first of the significant digits of the toll code. This first digit is transmitted because the incoming register at the primary toll office acknowledged the (last) digit it received with signal *A-1*.

The establishment of the connections via the toll network that interconnects the primary toll centers is controlled by the leading toll register and the significant part of the toll code, which will be transmitted to any other primary toll office involved in the connection, to select the shortest possible route to the wanted primary toll office. This part of the operation may be compared with that taking place in a large local multioffice area. If the primary toll offices are interconnected by direct lines, the operation is similar to that described in Section 3.1.1 for a direct connection between two local offices. If one or more of the primary toll offices may act as toll transit offices for primary toll offices not directly interconnected, operation is similar to that described in Section 3.1.2 for a transit connection in a local area. Also alternative routing or rerouting may take place in the toll network as described for similar circumstances in Sections 3.1.3 through 3.1.5 for a local area. However, in all of these cases, the selection will be controlled by the significant part of the toll code, instead of by the office code as described in connection with local calls.

3.2.3.2 *Open Numbering*

The description given in Section 3.2.2.2 will apply to the establishment of connections between different primary toll offices, taking into consideration that all of these offices are of the highest rank. This implies that at all of these offices only enough significant digits of the toll code will be received (commencing with the first of these digits) as are needed to determine whether the connection should terminate in the associated primary toll area or whether it must be switched in transit to another primary toll office (considering also that, in the same manner as described in Section 3.2.3.1, one or more of the primary toll offices may act as toll transit offices for primary toll offices not directly interconnected). It will also be evident that in this case, in the same manner as described in Section 3.2.3.1, the operation on the part of the toll network interconnecting the primary toll offices may be compared with that of a large local multioffice area, so that all operating possibilities may be provided as described in Sections 3.1.1 through 3.1.5 in connection with a local area.

As exactly the same types of signals are employed for the switching and line equipments within the primary toll areas as on the toll network interconnecting the primary toll offices, there is no reason, as in the case described in Section 3.2.2.1, to provide separate leading toll registers. The incoming registers may assume the function of leading registers for the type of connection now considered, without appreciable additional complication of these registers.

3.2.4 *Connections Terminating in Primary Toll Area*

3.2.4.1. *Closed Numbering*

The establishment of connections entering from other primary toll areas into a primary toll office will be completed in the terminating primary toll area under the control of incoming toll registers at the terminating primary toll office, which will perform the same functions as leading registers at this office charged with the completion of connections within the primary toll area. In the same manner as these leading registers, these incoming registers will receive the complete local number and control all subsequent operations within the terminating primary toll area.

So that the incoming register at the primary toll office will receive only the local subscriber's number and not the toll code, receipt of the last digit of the toll code at the preceding office will be recognized by signal *A-1,* which may be performed by the register at this preceding office based on its selection of a direct line to the terminating primary toll area.

In case the primary toll office acts as a toll transit office for connections between primary toll offices not directly interconnected or for the purpose of alternative routing between primary toll offices, it may be economical to combine the functions of the incoming toll registers with those of the toll transit registers, if the amount of toll transit traffic is insufficient to warrant provision of separate registers.

The use of a combined register for terminating and toll transit connections implies, however, that discriminative information be transmitted to this register to indicate whether the connection is terminating or in transit. Because this information may not be derived from the significant part of the toll code or from the local number, an obvious solution is to precede the toll code, in case of a toll transit connection, by the toll access code.

3.2.4.2 *Open Numbering*

For connections entering a primary toll office from other primary toll offices, the description given in Section 3.2.2.2 applies, taking into consideration that any office through which the connection must be routed subsequently will be of lower rank. Consequently, the part of the significant number of the toll code identifying the primary toll area need not be transmitted to subsequent offices.

3.2.5 *Control of Selection at Terminating Toll Area*

The control of the selection at the terminating toll area may be arranged in different ways depending on the facilities afforded by the switching equipment existing there.

If the equipment is capable of accepting multifrequency register signals, the operations will proceed directly under the control of the leading toll register at the originating toll office. An incoming register at the terminating office will then successively initiate the transmission from the leading register of the digits constituting the local office code. In all further respects, this incoming register also will act as a local transit register as shown in Figures 8 through 11, in which operations the leading toll register will perform the functions of an originating register.

If the equipment at the terminating area does not accept multifrequency signals, convertor registers will be needed to convert the multifrequency signals to the type of selective information required by the terminating switching equipment. These convertor registers may be located either at the terminating office to serve the whole of the area for all incoming calls or at individual offices if only some of the offices in the terminating area require different selective information. In the latter case, these convertor registers may handle toll calls in addition to incoming calls from other offices employing multifrequency signalling within the terminating area.

In each of these cases, the convertor register will cause the leading register to transmit all digits required to complete the connection by the successive transmission of acknowledgment signals *A-1,* until the units digit has been received. The units digit may be acknowledged by an appropriate backward signal from the convertor register, depending on the possibilities of the terminating switching equipment to signal the condition of the called line to this register, as explained in conjunction with Figures 4 through 7.

Another possibility consists in the use of the backward signal *A-9* or *A-10* from the last transit register in the connection that may accept multifrequency signals, for the purpose of informing the leading register that this has to transmit all remaining numerical information in a different form, for example, by decimal pulses. This will be possible only when all toll lines and junctions involved in the connection between the office at which the leading register is located and the terminating office are equipped for the transmission of decimal pulses.

3.3 International Connections

Disregarding certain differences that will be explained later, *the operating conditions with the new signalling methods as applied to international connections in many respects are comparable to those prevailing in a national toll network* employing a closed numbering system for each toll area.

Similarly to the assignment in a national network of so-called national toll codes to each primary toll area, which codes must be dialled ahead of the local subscriber's number only for connections between different primary toll areas, in the international network so-called international toll codes are assigned to each country and have to precede the national number only in case of international connections.

In a national network, each primary toll area is served by a primary toll office. All primary toll offices are interconnected by the national toll line network either with direct lines or via one or more of these primary toll offices serving as national toll transit offices. The selection at each of the toll offices involved in a connection through this network is determined by the significant part of the national toll code. In the international network, each country in principle is served by one principal "international toll office" via which access may be obtained to all subscribers within the country. These international toll offices are mutually interconnected by the international toll line network, either with direct lines or via one or more of them serving as international toll transit offices. The selection at each of the international toll offices involved in a connection is determined by the significant part of the international toll code dialled.

In a national toll network, the establishment of toll connections is under the control of leading national toll registers located at the originating toll office. Similarly, for international connections, leading international toll registers will control the establishment of the connection on the international network and will be preferably located at the originating international toll office.

Also comparable to a national system in which a larger number of register signals is provided for the exchange of information between toll registers than is employed by the local registers handling the connections within each toll area, the number of signals employed for international connections is larger than that used within national systems, as may be seen by referring to Tables 3 and 4.

Another point of resemblance is that in a national toll network distinction between local and toll calls is provided by dialling a national toll access code for the latter calls (that is, a figure or combination of figures not employed in the local numbering system) in a position similar to that for the local numbers; for international calls it is necessary to dial an international access code, after which the significant part of the international toll code may be dialled to signify the distant country to which the connection should be directed.

In some other respects, the comparison between national and international operating methods does not hold. A few cases will be mentioned.

3.3.1 *More Than One International Toll Office*

Some countries may have more than one international toll office to permit more economical routing to and from different adjacent countries than would be provided by a single international toll office. However, the same international toll code must be dialled for calls to such countries, independently of the incoming international toll office through which they will enter the country. A particular case influencing the signalling may arise if calls from other countries may enter through any of several of these incoming international toll offices, depending on the location of the called subscriber within the country of destination. Then, the registers at the outgoing international toll office of the neighboring country (or at the last international toll transit office as the case may be) have to determine from the significant part of the called subscriber's national toll code to which of the incoming international toll offices at the country of destination the call should be routed. Obviously, these registers receiving the last digit of the international toll code indicating the country of destination, must, before effecting a selection, call for a sufficient number of digits of the national toll code, which follows the international toll code, to determine the most-economical routing. After one of the incoming international toll offices at the country of destination has then been selected, it will normally be necessary to transmit to this office the complete national toll code. The backward signal *A-13* (send first digit of national number) will then be used for acknowledging the last digit received by the register of this incoming international toll office.

If the incoming international toll office serving the country of destination may also be used as an international toll transit office, it may be necessary to transmit to this office the international country code so it may determine that the call is terminating in the country concerned. The backward signal *A-11* (send first digit of international country code) is then employed by the register at the neighboring country for acknowledging the last digit received, so that the desired effect is produced.

3.3.2 *Special Requirements Applying Only to International Operation*

There are a number of special requirements applying only to international operation, such as recording at each originating country the charges and routing of calls for use in the division of revenues on international calls between different administrations. This does not affect the signalling methods, however. In certain cases, congestion signals transmitted from international transit toll offices to the originating international toll registers may signal the fact that an alternative route has been taken at a transit office. This permits a record of the routing to be kept at the originating end.

3.3.3 *Echo Suppressors*

Another question affecting signalling is the need *for the insertion of echo suppressors* in certain long-distance connections. Signals are given for this purpose in Tables 3 and 4.

Although no definite method of operation for the insertion of echo suppressors has been decided on, it should be noted that during the process of compelled end-to-end register signalling echo suppressors may not be used because they would prevent the simultaneous transmission of signals in both directions. Arrangements will, therefore, have to be provided to insert echo suppressors in the speech path if the routing information indicates need for them but only after the exchange of register signals is completed.

Furthermore, it may also be found necessary to disable echo suppressors after a connection has been completely established to permit data transmission with some form of error correction.

The most suitable location for echo suppressors would be at the outgoing and/or incoming international toll offices, at which the speech path is through connected only after the international registers are disconnected.

3.3.4 *Transborder Connection*

A particular type of international connection is the *transborder connection.* This is a relatively short connection between towns on opposite sides of an international border that is not routed through either of the principal international toll offices of the two countries concerned. To establish such connections, it is necessary that the register at the border town in the originating country examines the significant part of the international toll code and at least a part of the called subscriber's national toll code to determine whether or not the connection may be established via the transborder line. The fact that this is *not* possible and the call should be routed through the originating country's principal international toll office can be established only after a variable number of digits have been examined, namely, the international toll code alone or this code plus a variable number of digits of the national toll code that will provide the necessary distinction.

This may consequently require the repetition of a variable number of digits from the originating register, which must then be transmitted to the principal outgoing international toll register. This will be effected by the transmission of an appropriate backward signal from the register at the border town, when it has determined that no transborder connection is permitted.

3.4 Identification of Calling Line's Number

A number of optional signals have been included in the Tables of Signals that administrations are free to use in their own network for specific national purposes. Two of these signals are backward signals *A-9* and *A-10,* of which some use has already been described (see Section 3.2.5), if it is desired to change over from multifrequency-code signalling to dial pulsing at a certain point during the establishment of the connection.

Another purpose for these signals is to identify the calling subscriber's number as may be needed for automatic message recording and accounting on toll connections.

These signals permit the concentration of automatic message recording equipment at important toll offices and for the transmission to that toll office of the calling number via the toll lines used for establishing the toll connection from the originating office. The use of multifrequency-code signalling permits the calling number to be transmitted very rapidly while the switching operations take place at the toll office.

For this purpose, after the register at the toll office has received the last digit of the called number needed to effect its selection, it will acknowledge this digit by backward signal *A-9* or *A-10,* which requests transmission of the first digit of the calling line's local number or national number, respectively.

Two distinct signals are needed because in certain cases where the automatic message recording equipment is located at the primary toll office serving a primary toll area with closed numbering, it is sufficient to obtain the local number alone as all subscribers will have the same national toll code. In other cases, the recording equipment may serve several closed-numbering areas, for example, when this equipment serves for recording international calls exclusively and is located at the international toll office. It will then be necessary to obtain the complete national number of the called subscriber, less the access code.

Receipt of either *A-9* or *A-10* by the originating register will cause the calling number to be identified and will connect a forward signal indicating the value of the first digit of this number. The receipt of this and all following digits except the last of the calling number will be acknowledged by the same backward signal *A-9* or *A-10,* which thereby assumes the meaning of transmit next digit of calling number.

The last digit is acknowledged (if this is known to be the last digit at the receiving register) by backward signal *A-1*. This restores both the transmitting and receiving registers to the condition in which selection may be resumed and which prevailed before signal *A-9* or *A-10* was sent first, except that the originating register will be initiated to transmit the next digit of the called number. See Figure 13.

If it is not known at the receiving register how many digits are in the calling number, as in the case of nonuniform numbering, the transmitting register, after having received the signal *A-9* or *A-10* acknowledging the receipt of the last digit of the calling number, will send the forward signal *A-12* (demand rejected). This will in its turn be acknowledged by backward signal *A-1,* after which both registers are again in the condition in which selection may be resumed with the transmission of the next digit of the called number. See Figure 14.

On international calls it is possible by using backward signal *A-13,* to obtain in a similar manner from the originating country the significant number of the international toll code signifying this country, and, if desired, part or all of the national number of the calling subscriber. The maintenance staff may thus be informed of the origin of the connection.

4. Conclusion

Agreement has been reached among many of the leading telephone administrations in Europe on a flexible and adaptable system of signalling in the control of telephone switching equipment for national and international networks. It uses the same principles of operation throughout these networks and provides high operating speed and inherent reliability.

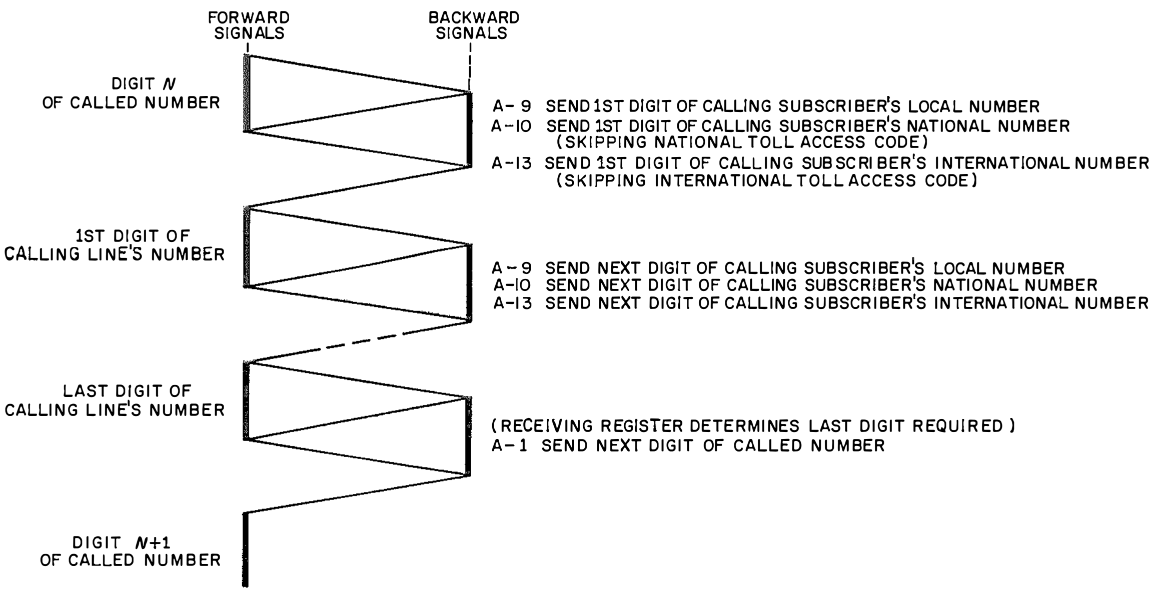


Figure 13—Identification of calling number if the last digit can be determined by the receiving register.

5. Appendix

5.1 Definitions and Functions of Line Signals

The definitions and functions of the line signals mentioned in Section 2.1 may be described similarly to those listed under the corresponding headings in "Recommendation Q 60" contained in the Red Book of the Comité Consultatif International Télégraphique et Téléphonique, volume 6, pages 93 through 98.

As some of the signals mentioned in Section 2.1 do not appear in the Red Book and others have a slightly different function, a complete list follows.

In these definitions the terms "outgoing termination" and "incoming termination" refer to that line termination, and the terms "forward signal" and "backward signal" refer to that direction of transmitting a signal to which these terms apply with respect to the direction in which the connection is established. This means that the terminations of both-way circuits may be either outgoing or incoming, depending respectively on whether they are located at the office from which the line is seized for a particular connection or at the office to which this connection is directed via the line.

Seizure (Forward Signal): This signal is transmitted on a line circuit after it has been engaged for a connection at its outgoing termination in order to switch the incoming termination into the "nonidle" condition and to initiate the connection to the incoming termination of an equipment capable of controlling the switching operations at the incoming office. It

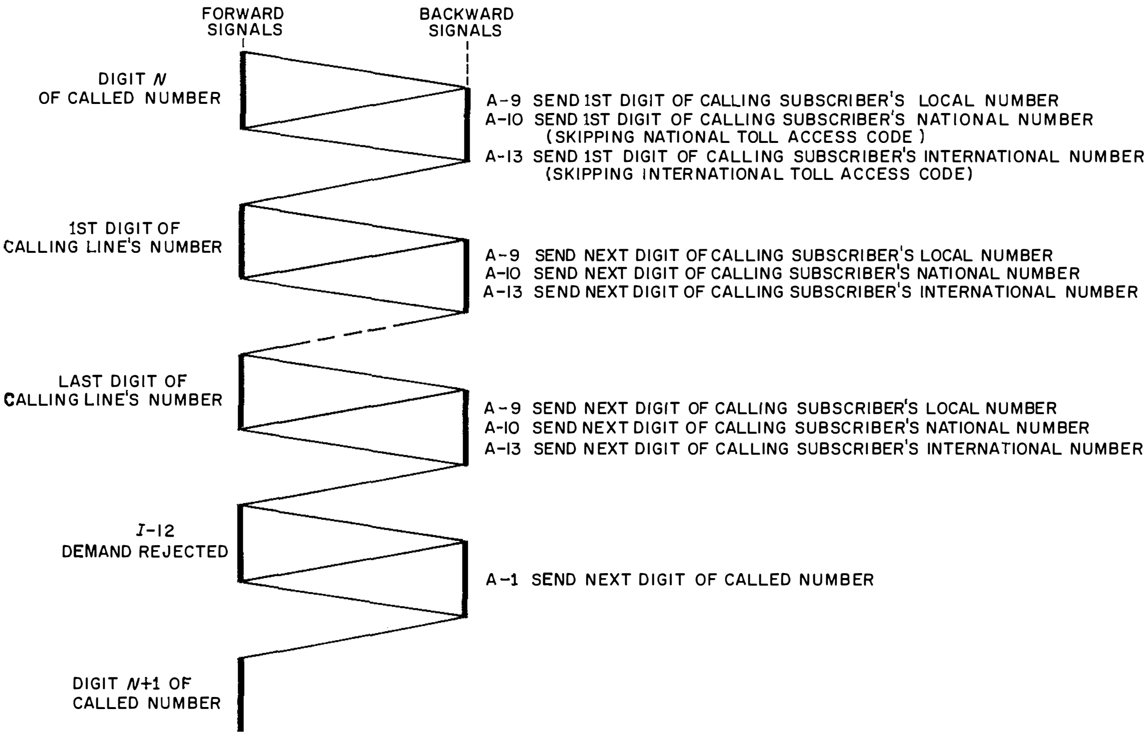


Figure 14—Identification of calling number if the last digit cannot be determined by the receiving register but can be by the transmitting register.

should be noted that no distinction is provided between terminating and transit connections by using two corresponding seizure signals, as it is considered preferable to provide this distinction by numerical signals.

Clear Forward (Forward Signal): This signal is transmitted from the outgoing to the incoming termination of a line circuit at the end of a connection, when (*A*) in the case of semiautomatic operation, the operator controlling the connection withdraws her plug from the line jack or accomplishes an equivalent operation, or (*B*) in the case of automatic operation, the calling subscriber replaces his telephone on the switch hook or accomplishes an equivalent operation as in the case of a subscriber's installation with supplementary telephone sets.

In automatic operation, this signal is automatically sent on receipt of certain backward signals indicating that the connection cannot be completed. This may be due to many reasons, such as: congestion in the switching or line equipment; called line being busy, out-of-order, et cetera; need for forced release because of the failure of the calling subscriber to clear within a predetermined interval subsequent to the receipt of the "clear back" signal; no reply, with calling subscriber failing to clear within a predetermined interval subsequent to the receipt of the signal indicating "end-of-selection" or "called subscriber free"; and abnormal operation at a transit center, when a "release guard" signal is received without a preceding "clear forward" signal.

At the termination of the "clear forward" signal, all switching equipment engaged in the connection must be liberated at both terminations of a line circuit, that is, at the outgoing, incoming, or transit office at which the line circuit terminates. (See, however, "Release Guard", below.)

Forward Transfer (Optional Forward Signal): This signal is transmitted in semiautomatic connections on international lines when the controlling operator at the outgoing intercontinental toll office wishes the assistance of an operator at the incoming international toll office. It will be transmitted from an outgoing line termination at the first-mentioned office to an incoming line termination at the last-mentioned office, without taking effect at transit offices through which the connection is switched and will serve normally to call for an assistance operator on a connection switched automatically through the incoming international toll office. If the connection at this office was switched manually by an operator, the signal will serve to recall this operator (operators with code *11* or *12*).

Toll Offering (Optional Forward Signal): This signal is transmitted in semiautomatic toll connections within a national network arranged for this type of service. It is used when the controlling toll operator, on finding a wanted number busy, and without breaking down that connection, wishes to communicate with this subscriber to inform him that a toll connection is in preparation for him. The toll offering condition may be undone by the transmission of the "end of toll offering" signal.

Reringing (Optional Forward Signal): This signal is transmitted in semiautomatic toll connections within a national network arranged for this type of service when the controlling toll operator wishes to rering a subscriber who had cleared after having been in communication with the operator previously. Normally, ringing is started by this signal, which is transmitted when the operator throws the ringing key, and is continued until, by restoring the ringing key, the ringing condition is undone. This may be produced by using a pulse signal for starting the ringing and another pulse signal "end-of-ringing" to stop the ringing, or alternatively, by using for the reringing signal a train of pulses that continues as long as the ringing key is thrown.

In particular cases, for example, when the toll switching circuits serving a local network are so arranged that the automatic ringing of the called subscriber on the selection of the called line is suppressed for certain types of operator's connections, these operators may initiate either manual or machine ringing by the use of the reringing signal.

Acknowledgment of Seizure (Optional Backward Signal): This signal is transmitted from the incoming to the outgoing termination of a line circuit immediately on receipt and recognition at the incoming termination of a seizure signal. It will be transmitted without awaiting the result of the switching operations (connection of a free register) that are initiated by the seizure signal and will merely serve to inform the outgoing termination that it may cease the transmission of the seizure signal. This signal is normally employed in conjunction with the seizure signal to provide for "compelled operation" by mutual control of these two signals.

Answer or Reanswer (Backward Signal): This signal is transmitted from the incoming to the outgoing termination of a line circuit to signal that the called subscriber has replied and, in general, to indicate the off-hook condition of the called line.

On semiautomatic connections, this signal will permit the controlling operator to supervise the answer of the called subscriber by visual means.

On automatic connections, this signal will initiate:

A. Commencement of charging for the connection by any appropriate means provided for this purpose, or of measuring the chargeable period.

B. Discontinuation of measuring the time interval before answering, which, if exceeded, would cause the connection to release forcibly (if provided).

C. Completion of the forward speech channel in certain cases.

Clear Back (Backward Signal): This signal is transmitted from the incoming to the outgoing termination of a line circuit to signal that the called subscriber has replaced his receiver on the switch hook, and, in general, to indicate the on-hook condition of the called line. In semiautomatic operation, this signal will permit the controlling operator to supervise the clearing of the called subscriber by visual means.

In automatic operation this signal will initiate:

A. Start of measuring a time interval, after which charging of the calling subscriber will be interrupted (or the measurement of the chargeable period will be discontinued) and the connection will be forcibly released if the calling subscriber had not cleared previously.

B. Interruption of the forward speech channel in certain cases.

Release Guard (Optional Backward Signal): This signal is transmitted from the incoming termination of a line circuit, either at an incoming or a transit office, to its outgoing termination, either at a transit or an outgoing office, immediately on the receipt and recognition at the incoming termination of a clear forward signal. It will be continued until the moment the incoming termination, in response to the clear forward signal, has been caused to switch into the idle condition, excepting from this condition the fact that the backward release guard signal is transmitted. (Which means that also the clear forward signal must have terminated.) The discontinuation of the release guard signal at this moment will cause both outgoing and incoming terminations to become simultaneously available for a next connection.

The presence of this signal will permit the clear forward signal to continue indefinitely, until its recognition at the incoming termination has been signalled to the outgoing termination by the release guard signal, so that compelled operation is thereby provided by the mutual control of these two signals.

Blocking (Optional Backward Signal): This signal is transmitted from an incoming to an outgoing termination of a line when it is desired at the incoming office to render the circuit inaccessible at its outgoing termination. The receipt of this signal at the outgoing termination in its idle condition causes the signalling equipment to mark the circuit "busy" to operators or automatic selectors normally having access to it. The blocking condition is different from a genuine busy condition in that it will produce an alarm at the outgoing termination after a predetermined delay, so that if the blocking condition is continued after this delay, the maintenance staff is warned to investigate this abnormal case and take appropriate measures. Normally, the blocking signal should be employed for a short period only, for example during routine testing of the incoming terminating equipment.

Acknowledgment of Optional Forward Signals (Optional Backward Signals): In conjunction with each of the forward signals mentioned in items 5.1.3 through 5.1.5, a backward acknowledgment signal may be optionally provided so that each pair of forward and backward signals thus constituted will provide for compelled signalling by mutual control.

5.2 Definitions of Terminology

Office: General designation of a switching center.

Local Office: Office to which the subscribers in one locality (or its equivalent) or part of a locality are connected.

Originating Office: Office to which the subscriber is connected who originates the connection considered.

Terminating Office: Office to which the called subscriber is connected for the connection considered.

Local Area: Area served by the office(s) of one locality.

Toll Office: Office employed for the establishment of connections between different local areas.

Primary Toll Office: Toll office of highest rank in a national toll network. Many or all of these offices are directly interconnected by toll lines, constituting the principal national long-distance toll line network.

Transit Toll Office: Primary toll office used for switching transit connections between other primary toll offices not directly interconnected or for alternative routing of connections between other primary toll offices interconnected by direct lines.

Primary Toll Area: Area for which a primary toll office is the pivotal switching center for connections to and from other similar areas, served by other primary toll offices.

Secondary Toll Office: Toll office of second rank within a primary toll area and that is directly dependent on a primary toll office for connections to and from other primary toll areas.

Secondary Toll Area: Area for which a secondary toll office is the pivotal switching center for connections to and from other offices of equal and higher rank.

Tertiary Toll Office: Toll office of third rank within a secondary toll area and that is directly dependent on a secondary toll office for connections to and from other toll offices of higher rank.

Tertiary Toll Area: Area for which a tertiary toll office is the pivotal switching center for connections to and from other areas of equal or higher rank.

Toll End Office: Toll office serving one local area. Toll end offices constitute the toll offices of lowest rank in a national toll network and are dependent on a toll office of the next higher rank (primary, secondary, or tertiary toll office depending on how many other ranks of toll offices exist), for connections to other toll offices of higher rank.

Originating Area: Area in which the connection considered is originated.

Terminating Area: Area in which the connection considered is terminated.

Originating Toll Office: Toll office that incorporates the leading registers controlling the establishment of the toll connection considered.

Terminating Toll Office: Toll office that incorporates the registers controlling the establishment of an incoming toll connection within the terminating area.

Single-Office Area: Locality (or its equivalent) served by one local office.

Multioffice Area: Locality served by more than one local office.

Local Number: Subscriber's number used for establishing connections between subscribers connected to the same office. This same number may be used by subscribers connected to offices other than that to which the called subscriber is connected, when these offices are located in the same closed numbering area as the terminating office.

Closed Numbering Area: Area comprising all offices the subscribers of which may mutually call one another by using the local number only.

Closed Numbering System: System of allotting subscriber's numbers in a national toll network according to which each closed-numbering area comprises several local areas. In most cases, this area may be identified with a primary toll area.

Open Numbering System: System of allotting subscriber's numbers in a national toll network according to which each closed-numbering area comprises one local area only.

National Number: Subscriber's number to be used for a connection within a national network when originated outside of the closed-numbering area in which the designated subscriber is connected. The national number is the local number designating a subscriber preceded by the national toll code, designating the closed-numbering area in which this subscriber is connected.

National Toll Code: Part of a national number preceding the local number and comprising the national toll access code and a significant number identifying the closed-numbering area in which the designated subscriber is connected.

National Access Code: Code of one or more digits by which national numbers commence and by which these numbers are distinguished from local and international numbers.

International Number: Subscriber's number to be used for a connection when originated outside of the national network in which the designated subscriber is connected. The international number is the national number less the national access code, preceded by the international toll code.

International Toll Code: Part of the international number preceding the national number less the national access code and comprising the international access code and a significant number identifying the country in which the designated subscriber is connected.

International Access Code: Code of one or more digits by which international numbers commence and by which these numbers are distinguished from local and national numbers.

International Toll Office: Office at which toll lines terminate interconnecting different national toll networks and through which the international traffic handled via these toll lines is switched to and from the whole or part of the country in which the office is located.

International Transit Toll Office: International toll office used for switching transit connections between other international toll offices not directly interconnected, or for alternative routing of connections between other international toll offices interconnected by direct lines.

Local Register: Register serving principally for the establishment of connections within a local area and provided at local offices.

Toll Register: Register provided at toll offices.

Outgoing Toll Register: Register at the toll end office serving the calling subscriber for a connection considered.

Incoming Toll Register: Register controlling the completion of connections within the terminating toll area.

Leading Register: Register located relatively near to the calling subscriber and charged with supervising the establishment of toll connections. This register may be located at the originating toll end office or at a toll office subsequently engaged. It will receive the full numerical information required for completing the connection at and beyond the toll office at which it is located and supply any part of this information to any subsequent office engaged, on receipt of appropriate backward signals. It will remain connected until it receives a backward signal informing it on the final result of the selective operations.

Leading National Toll Register: Leading register supervising the establishment of national toll connections. This may be located at the primary toll office serving the calling subscriber or at any preceding toll office.

Leading International Toll Register: Leading register supervising the establishment of international toll connections. This may be located at the international toll office serving the calling subscriber or at any preceding toll office.

Toll Line: Any type of transmission channel providing both-way speech possibilities between any two toll offices.

Terminology in use particularly in connection with primary toll areas with closed numbering (applied sometimes also to such areas with open numbering)

Rural Area: Primary toll area, but usually not including the local area directly associated with the primary toll office.

City Area: Local area associated directly with primary toll office.

Rural Main Office: Part of primary toll office required for switching connections betweer local areas within the primary toll area.

Toll Office: Part of primary toll office used foı switching connections between any local areε within the primary toll area on the one hanc and other primary toll areas on the other hand

Rural Office: Office equipment serving anʒ local area within the primary toll area, excepting that directly associated with a primary tol] office, and usually including both the local anc toll offices associated with that local area.

(Rural) Sector Center Office: Office equipment associated with a local area and comprising a secondary toll office.

(Rural) Subcenter Office: Office equipment associated with a local area and comprising a tertiary toll office.

(Rural) End Office: Office equipment associated with a local area and comprising a toll end office.

(Rural) Sector: Secondary toll area.

(Rural) Subsector: Tertiary toll area.

Rural Junction: Toll line within primary toll area.

Toll Line: Toll line interconnecting primary toll offices.

Tie Line: Toll line interconnecting two toll offices in nonhierarchical manner.

Martinus den Hertog was born in Utrecht, The Netherlands, in 1901.

He became a student engineer for Bell Telephone Manufacturing Company in 1921, working until 1925 as an equipment engineer. He then joined the circuit engineering group and served as its head from 1933 to 1953.

From 1953 to 1959, Mr. den Hertog was chief engineer and manager of the switching systems group. He became commercial manager of the company in 1959 and two years later was appointed assistant to the technical director of ITT Europe, where he has charge of the coordination of the development of switching systems.

United States Patents Issued to International Telephone and Telegraph System; November 1961–January 1962

Between 1 November 1961 and 31 January 1962, the United States Patent Office issued 36 patents to the International System. The names of the inventors, company affiliations, subjects, and patent numbers are listed below.

R. T. Adams, B. M. Mindes, and Z. G. Lyon, ITT Federal Laboratories, Post-Detection Diversity Combining System, 3 013 151.

L. D. Apt, ITT Kellogg, Dial Controlled Intercommunication Systems, 3 014 097.

A. H. W. Beck, Standard Telephones and Cables (London), Thermionic Cathodes, 3 013 171.

T. E. Beling, ITT Laboratories, System for Providing Short High-Amplitude Square Voltage Pulse, 3 015 777.

J. I. Bellamy, ITT Kellogg, Decade-Counting Device, 3 007 638.

T. G. Brown, Jr., ITT Laboratories, Decoder for Line Printer, 3 013 119.

P. Cheilik, ITT Federal Laboratories, Magnetic Read and Write System, 3 013 255.

B. Dal Bianco and M. Scata, Fabbrica Apparecchiature per Comunicazoni Elettriche (Milan), Electromagnetic Switch Apparatus, 3 014-102.

C. L. Day, Farnsworth Electronics Company, Method of Making Bowl-Shaped Fine-Mesh Screens for Electron Discharge Tubes, 3 017-687.

E. M. Deloraine and H. G. Busignies, International Telephone & Radio Manufacturing Corp., Radar Counter-Measure Radio Repeater, 3 015 096.

M. J. DiToro, ITT Federal Laboratories, Aural Detection Apparatus Comprising an Acoustical Delay Line Having External Feed-back Circuit, 3 013 212.

A. G. Field, Standard Telephones and Cables (London), Thermionic Valves, 3 013 176.

L. G. Fischer, ITT Federal Laboratories, Radio Range Receiver System, 3 013 264.

J. C. Gibson, ITT Kellogg, Primary-Secondary Link-Spread Crossbar Selector System, 3 017-465.

F. S. Gutleber, ITT Federal Laboratories, Diversity Receiving System Having Separate Phase-Angle Indicators, 3 013 150.

L. B. Haigh and T. W. Tuttle, ITT Federal Laboratories, Telecommunication Link, 3 013-111.

H. Herbig and E. Schmidt, Federal Telecommunication Laboratories, Transistor Oscillators, 3 015 696.

R. C. P. Hinton, B. Dzula, and A. Fettweis, ITT Federal Laboratories, Recording System, 3 013 253.

R. L. M. LeQueau, Compagnie Générale de Constructions Téléphoniques and Le Matériel Téléphonique (Paris), Telephone Systems, 3 008 010.

L. Lewin and Y. Klinger, Standard Telecommunication Laboratories (London), Maser-Type Oscillator, 3 013 217.

US Patents to ITT; November 1961–January 1962

F. P. Mason and R. G. Stemp, Creed & Company (Croydon), Facsimile Apparatus, 3 015 690.

B. Minakovic, Standard Telephones and Cables (London), Traveling-Wave Tubes, 3 013 177.

H. Reiner, Mix & Genest (Stuttgart), Transistor Double-Integrating Circuit, 3 013 160.

R. Scheidig, Standard Elektrik Lorenz (Stuttgart), Reed-Contact Thermo Relay, 3 008 019.

R. Schultz and J. Sullivan, Standard Telephones and Cables (London), Trigger Circuit, 3 015 736.

J. Schockaert, Bell Telephone Manufacturing Company (Antwerp), Carrier Supports for Sorting Machine, 3 015 379.

J. R. Simkovich, ITT Federal Laboratories, Radial Resonant Cavities, 3 013 230.

A. T. Starr, Standard Telecommunication Laboratories (London), Semi-Conductor Devices, 3 013 192.

R. Thyen, International Standard Electric Corporation, Circuit Arrangement for Encoding Devices, 3 015 805.

M. Toohig and C. L. Day, Farnsworth Electronics Company, Method of Making Charge-Storage Electrodes for Charge-Storage Tubes, 3 015 586.

A. W. Wallens, Creed & Company (Croydon), Teleprinter Communication Systems, 3 013 112.

J. Weber and C. Heck, Süddeutsche Apparate Fabrik (Nuremberg), Method of Making Ferrites, 3 015 538.

E. P. G. Wright, A. D. Odell, and C. Parker, Standard Telecommunication Laboratories (London), Electronic Equipment Practice 3 015 755.

E. P. G. Wright, Standard Telecommunication Laboratories (London), Data Processing Systems, 3 013 120.

E. P. G. Wright, Standard Telecommunication Laboratories (London), Data Processing Equipment, 3 013 251.

E. Zaratkiewicz and E. Smith, ITT Laboratories, Semi-Conductor Rectifiers and Method of Manufacture, 3 015 591.

Electronic Equipment Practice

3 015 755

E. P. G. Wright, A. D. Odell, and C. Parker.

A mounting rack and functional sub-unit assemblies in which all of the sub-units are identical. These units are shown as basic on-off switching units in the form of sub-panels. These sub-panels are provided with jack terminals for plugging into a prewired rack to provide complete functional equipment.

Method of Making Ferrites

3 015 538

J. Weber and C. Heck

Manganese-zinc mixed ferrites annealed in an atmosphere containing a small amount of oxygen and about ten percent sulphur dioxide. This process achieves a substantial improvement in magnetic properties of the ferrites, particularly a large increase in permeability.

Post-Detection Diversity Combining System

3 013 151

R. T. Adams, B. M. Mindes, and Z. C. Lyon

A diversity receiver post-detection combiner circuit for phase- or frequency-modulated waves provided with devices having a limiting action preceding the individual discriminators, to maintain equal gain in all the channels of the diversity receiving system.

Thermionic Cathodes

3 013 171

A. H. W. Beck

A thermionic dispenser cathode in the form of a relatively large block of sintered refractory metal powder mixed with emissive material, having a very small block of sintered emissive material integral with the larger block and serving to provide the emissive surface.

Electromagnetic Switch Apparatus

3 014 102

B. Dal Bianco and M. Scata

A magnetic-reed-type switch in which the switch envelope is mounted between two plates magnetically energized to normally operate the switch. A magnetic shield is positioned around the switch envelope and is movable out of shielding relation to control operation of the switch.

Principal ITT System Products

Telecommunication Equipment and Systems

Automatic telephone and telegraph central office switching systems
Private telephone and telegraph exchanges—PABX and PAX, electromechanical and electronic
Carrier systems: telephone, telegraph, power-line
Long-distance dialing and signaling equipment
Automatic message accounting and ticketing equipment
Switchboards: manual, central office, toll
Telephones: desk, wall, pay-station
Automatic answering and recording equipment
Microwave radio systems: line-of-sight, over-the-horizon
Radio multiplex equipment
Coaxial cable systems
Submarine cable systems, including repeaters
Data-transmission systems
Teleprinters and facsimile equipment

Military/Space Equipment and Systems

Aircraft weapon systems
Missile fuzing, launching, guidance, tracking, recording, and control systems
Missile-range control and instrumentation
Electronic countermeasures
Electronic navigation
Power systems: ground-support, aircraft, spacecraft, missile
Radar
Simulators: missile, aircraft, radar
Ground and environmental test equipment
Programmers, automatic
Infrared detection and guidance equipment
Global and space communication, control, and data systems
Nuclear instrumentation
Antisubmarine warfare systems
System management: worldwide, local

Industrial/Commercial Equipment and Systems

Distance-measuring and bearing systems: Tacan, DMET, Vortac, Loran
Instrument Landing Systems (ILS)
Air-traffic control systems
Direction finders: aircraft and marine
Ground and airborne communication
Data-link systems
Inverters: static, high-power
Power-supply systems
Altimeters
Flight systems
Railway and power control and signaling systems
Information-processing and document-handling systems
Analog-digital converters
Mail-handling systems
Pneumatic tube systems
Broadcast transmitters: AM, FM, TV
Studio equipment
Point-to-point radio communication
Marine radio
Mobile communication: air, ground, marine, portable
Closed-circuit television: industrial, aircraft, and nuclear radiation
Slow-scan television
Instruments: test, measuring
Oscilloscopes: large-screen, bar-graph
Vibration test equipment
Magnetic amplifiers and systems
Alarm and signaling systems
Telemetering
Intercommunication, paging, and public-address systems

Consumer Products

Television and radio receivers
High-fidelity phonographs and equipment
Tape recorders
Microphones and loudspeakers
Refrigerators, freezers
Air conditioners
Hearing aids
Incandescent lamps
Home intercommunication equipment
Electrical housewares

Cable and Wire Products

Multiconductor telephone cable
Telephone wire: bridle, distribution, drop
Switchboard and terminating cable
Telephone cords
Submarine cable
Coaxial cable, air and solid dielectric
Waveguides
Aircraft cable
Power cable
Domestic cord sets
Fuses and wiring devices
Wire, general-purpose

Components and Materials

Power rectifiers: selenium, silicon
Parametric amplifiers
Transistors
Diodes: tunnel, zener, parametric
Semiconductor materials: selenium, germanium, silicon
Capacitors: wet, dry, ceramic
Ferrites
Tubes: power, transmitting, traveling-wave, rectifier, receiving, thyratron
Picture tubes
Relays and switches: telephone, industrial
Magnetic counters
Resistors
Varistors
Fluorescent starters
Transformers
Quartz crystals
Crystal filters
Printed circuits
Hermetic seals
Magnetic cores

VOLUME 38 • NUMBER 1 • 1963