the Lenkurt Demodulator

1966

ISSUES

LENKURT ELECTRIC San Carlos, California, U.S.A.

SUBSIDIARY OF GENERAL TELEPHONE & ELECTRONICS GT&E

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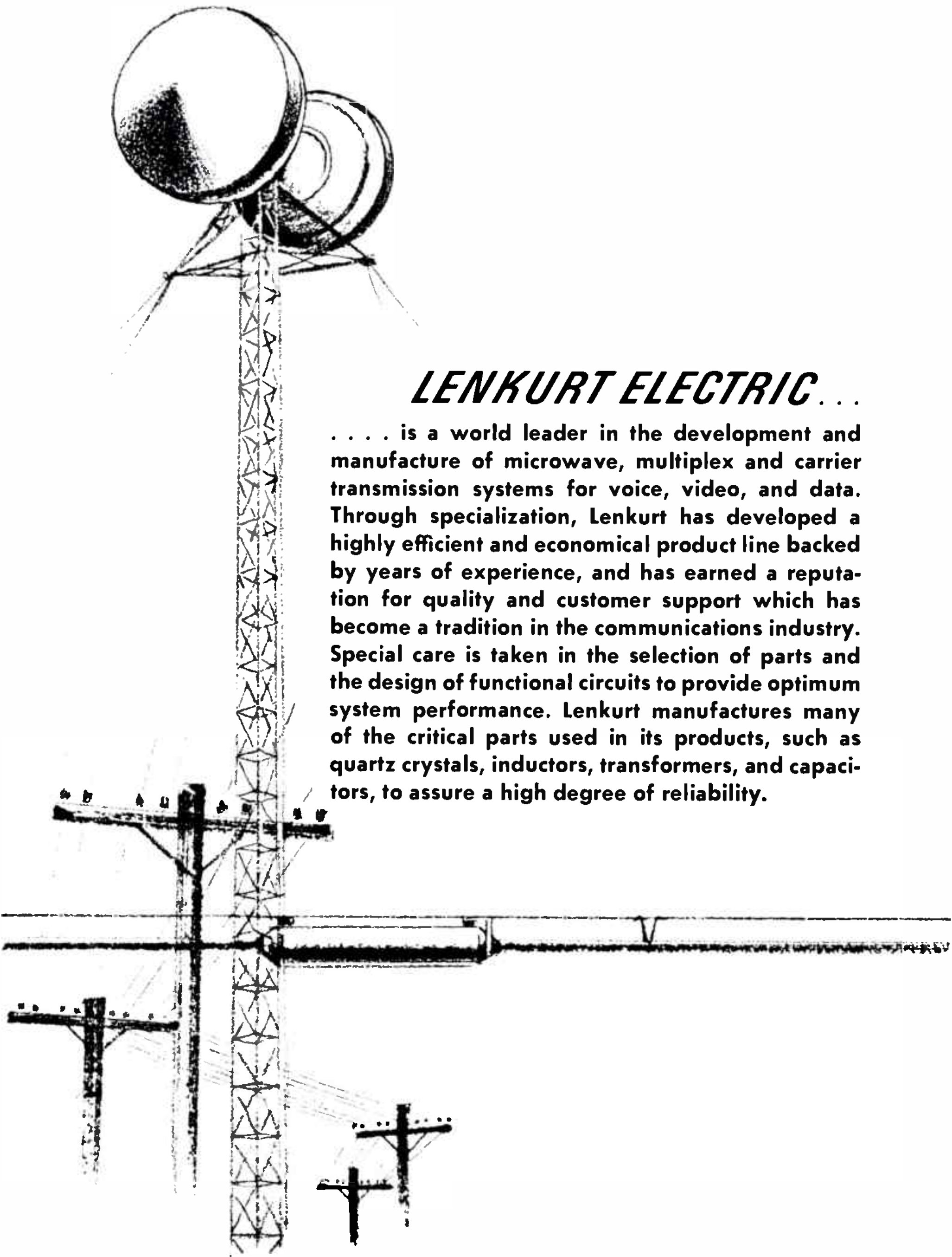
PREFACE

The Lenkurt Demodulator is an informative technical periodical published monthly and circulated without charge to technicians, engineers, and managers of companies or government agencies who operate communications systems, and to educational institutions. Each issue features an interesting and instructive article dealing with the subject of telecommunications.

This **special complimentary volume** is a collection of the twelve issues for 1966. If you would like to receive the **Demodulator** regularly, a subscription card is attached inside the rear cover for your convenience.

EDITOR

The Lenkurt Demodulator



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. . . . is a world leader in the development and manufacture of microwave, multiplex and carrier transmission systems for voice, video, and data. Through specialization, Lenkurt has developed a highly efficient and economical product line backed by years of experience, and has earned a reputation for quality and customer support which has become a tradition in the communications industry. Special care is taken in the selection of parts and the design of functional circuits to provide optimum system performance. Lenkurt manufactures many of the critical parts used in its products, such as quartz crystals, inductors, transformers, and capacitors, to assure a high degree of reliability.

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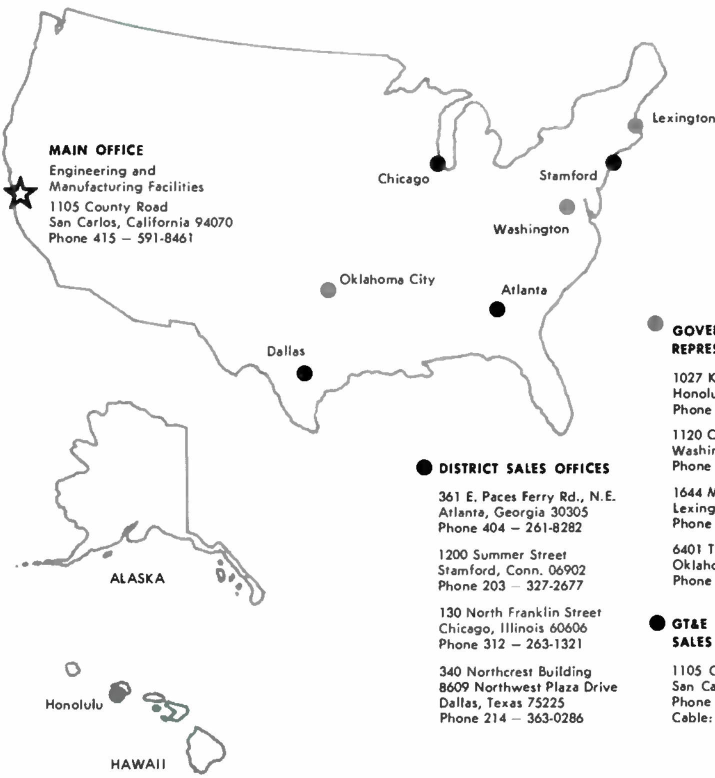
CUSTOMER SERVICES

Many valuable services are offered to assure that customers receive all of the benefits and economies of Lenkurt's high-quality products. Each of these services has been specially designed to help customers plan, install, operate, and maintain the wide selection of microwave, multiplex, carrier, and data transmission systems engineered and manufactured by Lenkurt.

ADDITIONAL INFORMATION

In addition to the **Demodulator,** Lenkurt has many other types of publications and descriptive literature. Included are product brochures (P4's) which briefly describe the technical characteristics and applications of Lenkurt's major systems. Also, comprehensive technical publications and drawings are available, containing detailed equipment engineering considerations.

If you are interested in receiving more information about Lenkurt products and services, please contact one of our sales offices or Government engineering representatives, or write directly to our Main Office.



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JANUARY, 1966
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the Lenkurt Demodulator

Worldwide Defense Communications...

In the last five years the Department of Defense and its Defense Communications Agency have been implementing two worldwide communications networks, known as AUTOVON *and* AUTODIN. *As the needs of the world change, so do the responsibilities of the communications industry. Close cooperation between the Federal Government and private enterprise has been necessary to successfully carry out such a project.*

This article briefly reviews the needs for such a communications system, and gives an overall view of its functions and capabilities.

At a time when the United States military and diplomatic forces are spread around the globe in proportions greater than at any time in history, communication is decidedly *the* most important factor in establishing and maintaining centralized control. The military tactic of selective response — retaliation in proportion to the enemy's action — has placed a tremendous burden on national communications agencies. They must gather and assimilate pertinent information, transmit it reliably to centralized locations, and disseminate directives instantly to field forces, all in time limits measured in minutes. Actions, diplomatic and military, must share a high degree of worldwide coordination, with decisions in one corner of the world in concert with those in every other part of the world.

Traditionally, communications have been centralized within each branch of the Armed Forces. The Army, Navy, and Air Force have maintained their own individual global networks serving their own needs. Policy coordination existed at higher levels of command, with directives filtering down to field commands through separate military channels.

Post-war years saw the growth of numerous civil and Government communications channels criss-crossing the nation and the world. For example, the National Aeronautics and Space Administration (NASA) operates a specialized and complex worldwide network for continuous contact with astronauts in flight. Additionally, the State Department, the Federal Aviation Agency (FAA), the Interior Department, the Immigration Service and others find need for their own communications systems linking centralized control with numerous locations.

Then came another compounding factor — the need to transmit high speed data between many of these points. In a very few years computer technology grew to giant proportions, and telephone circuits were no longer simply paths for human conversation, but carriers of a new machine language made up of high speed, coded pulses. Equipment sophistication was forced to meet this new demand.

New Technology

Miniaturization, largely the result of intensified aerospace research for smaller and more reliable on-board electronics in spacecraft, added to the communication engineer's ability to create extremely small broadband components and fast switching methods. Transistors, diodes and others in the solid state family were added tools for building high capacity communications systems in small packages, with fewer heat problems and power needs.

At the same time, the communications industry attempted to keep up with the increasing demands for service and reliability by establishing more and more routes across the country, and laying thousands of miles of ocean cable to match the needs of around-the-world message delivery.

Soon it became increasingly obvious that the duplication of effort being made by the many agencies serving national needs were not only inefficient because of incompatibility and needless duplication, but were making it more and more difficult to maintain centralized control.

DCA Established

In May 1960, the Defense Communications Agency (DCA) was formed to coordinate military communications. At about the same time, concepts involving all Government agency communications were formulated into a plan known as the National Communi-

Figure 1. The Defense Communications Agency coordinates worldwide military communications through the AUTOVON *and* AUTODIN *networks.*

cations System. Under this plan, duplication was reduced in all communications systems under government control. As one system developed, no others were to be authorized for the same purpose. Supplementing this concept, the DCA's precise mission was to form a single, integrated Defense Communications Systems (DCS) from existing military networks.

To meet the long-haul, point-to-point telecommunications requirements of the Department of Defense, the DCS has been faced with problems of technological and geographic enormity. The system must be capable of providing instantaneous service almost any place in the world, withstanding both natural and man-made failures, moving large volumes of traffic, and satisfying the need for highly secure (classified) communications.

To accomplish this, two major networks are being implemented: AUTOVON (Automatic Voice Network) and AUTODIN (Automatic Digital Network). Where possible, such as in the Continental United States (CONUS), telephone and telegraph communications facilities are being leased from the common carriers. Overseas, the Government is building and operating most of its own facilities.

Existing military networks became the skeletons for the two DCS networks. AUTOVON, primarily a voice network, is built on portions of the Strategic Army Communications System (STARCOM), the Army's Switched Circuit Automatic Network (SCAN) and a switched network developed for the North American Air Defense Command (NORAD). Similarly, AUTODIN has used for its backbone the Air Force Data Communications Network (AFDATACOM). Both AUTOVON and AUTODIN will eventually provide a multitude of fast, computer-controlled automatic switching centers all over the world. Many of them are already in use.

AUTOVON

AUTOVON in the Continental United States was initiated with seven important switching centers — soon to become nine — and will continue to grow until there are 74 stateside locations (including as many as nine in Canada) and 23 overseas centers. Of those overseas, seven will be in the Pacific, one each in Alaska and Panama, and fourteen in the Europe-Mediterranean area (Figure 3). For the most part, the automatic switching centers will be located outside heavy industrial areas or other potential prime targets. Interlacing the switches will be numerous alternate routes for back-up and computer-controlled rerouting in case of failure or overload conditions.

The automatic switching centers for AUTOVON serve much the same function as the telephone company central office, providing circuit connections for all users, and interconnecting with a large number of trunks to other switching centers. The automatic centers offer many advantages over manual or semiautomatic techniques, including less restoral time in case of breakdown, and reducing the quantity of full-time allocated (or dedicated) circuits necessary. Automatic switching provides approximately 20 percent more capacity than previous methods, and survivability is three to four times greater.

AUTOVON Services

Four types of users will be connected to the AUTOVON switching system (Figure 2). General purpose subscribers will include most military installations, who will have service through their local PABX. Selected personnel will have direct access to the network through push button, touch tone telephone sets. Off-hook hot line service will be installed for certain command and control operations, and special grade service will permit use of some circuits for data and facsimile. In addition to these basic users, there will also be the capability of organizing special networks within AUTOVON for specific purposes. The automatic switches will be able to handle either the new tone dial signal from push button instruments, or conventional dial pulse signaling through the PABX.

A directly dialed connection through AUTOVON will be provided in about four seconds; difficult conditions may

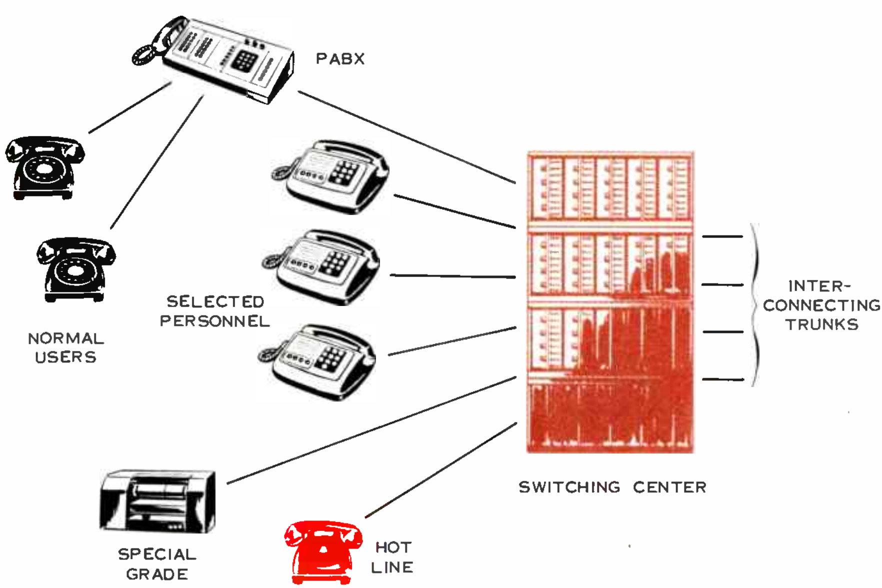


Figure 2. AUTOVON *switching centers will accept a variety of subscribers, providing direct connections to the network for priority and hot line telephones, normal phone use through* PABX, *and limited data and facsimile.*

lengthen this to a maximum of 10 seconds. Hot line service will be completed in less than two seconds. In addition, sophisticated memory and logic circuits will provide a four-level priority service, cued by push buttons on the touch tone telephone instrument. High priority calls will be routed through otherwise tied-up channels by seizing the circuits necessary to complete the call.

Priority 1 authorization will be held by the Secretary of Defense, Secretaries of Military Departments, Joint Chiefs of Staff, commanders of unified and special commands, and officers of four-star rank. Priority 2 will be shared by the operation deputies of the military services; the Director of the Joint Staff, Joint Chiefs of Staff; the Director of Operations, Joint Staff; and the Joint War Room operational communications.

Priority 3 is used for operational communications by other general or flag officers, by tactical commanders, and by the unified and specified command war rooms and service war rooms. Priority 4 applies to other urgent official communications of an operational nature. All other official communications are categorized as Priority 5.

Other services available to AUTOVON subscribers include conference call flexibility, allowing callers all over the world to be joined in conversation — as many as 30 or more at one time. Callers have at their disposal broadcast facilities, needed to disseminate information to large numbers of persons in remote areas, and recorded announcements which may be stored in the network for future release. Dial assistance switchboards will be available at various locations in each country to meet requirements for information, directory, intercepting, recording, or conference call connections. Abbreviated dialing will be provided for often-dialed numbers, and computer memory banks controlling switching and routing will also be open for reprogramming should the need arise.

AUTODIN

The AUTODIN network is the data equivalent of AUTOVON, and will process digital traffic for all commands of the Armed Forces. The initial increment of AUTODIN in the United States includes five AFDATACOM major relay centers currently operated by the Air Force. Added to this will be four more switching centers in the U.S. and 10 overseas: three in Europe, one each in Alaska and the Caribbean, and five in the Pacific (Figure 3).

AUTODIN is designed for use with punch cards, perforated or magnetic tape, teletype, digitalized graphic signals such as facsimile, and computer-to-computer language (Figure 4). Digital coding techniques also will provide for secure communication of classified information over the network. With full duplex, two-way connections between centers, sending stations will receive immediate confirmation of message delivery. Automatic equipment will hold a circuit until an acknowledge signal is received from the other end.

AUTODIN wideband switching and transmission circuits will allow processing large volumes of digital data. One common teletype coding technique requires 7½ binary digit pulses, or *bits,* to identify a single letter. This includes start and stop pulses. Assuming an average of 6 letters per word, a 100 word-per-minute teletype system must have a speed of 75 bits per second. In this reference, it is interesting to note that a man can talk at approximately 16 bits per second, or absorb about 60 bits per second when listening. Computers and other digital devices will transmit

AUTOVON

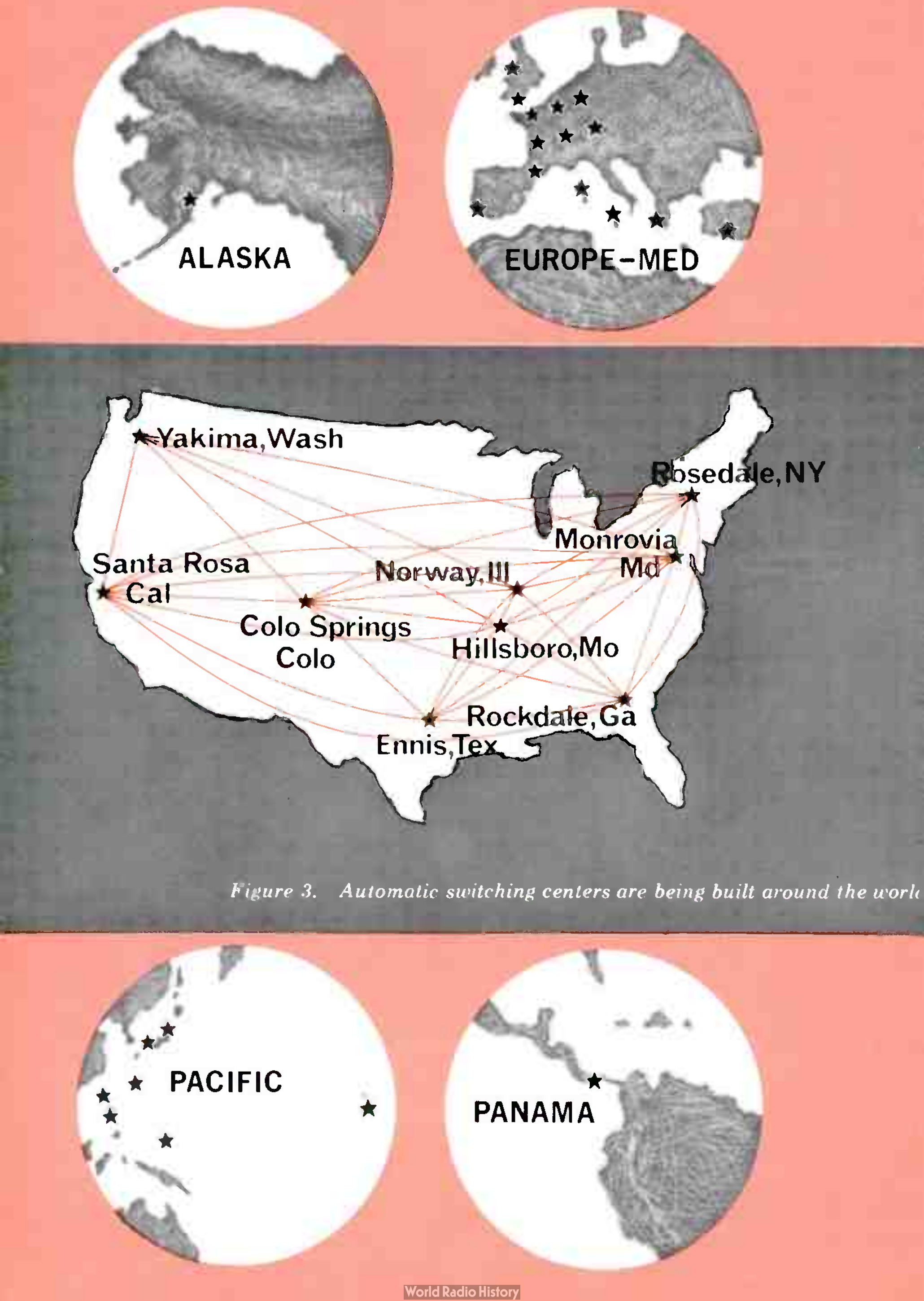
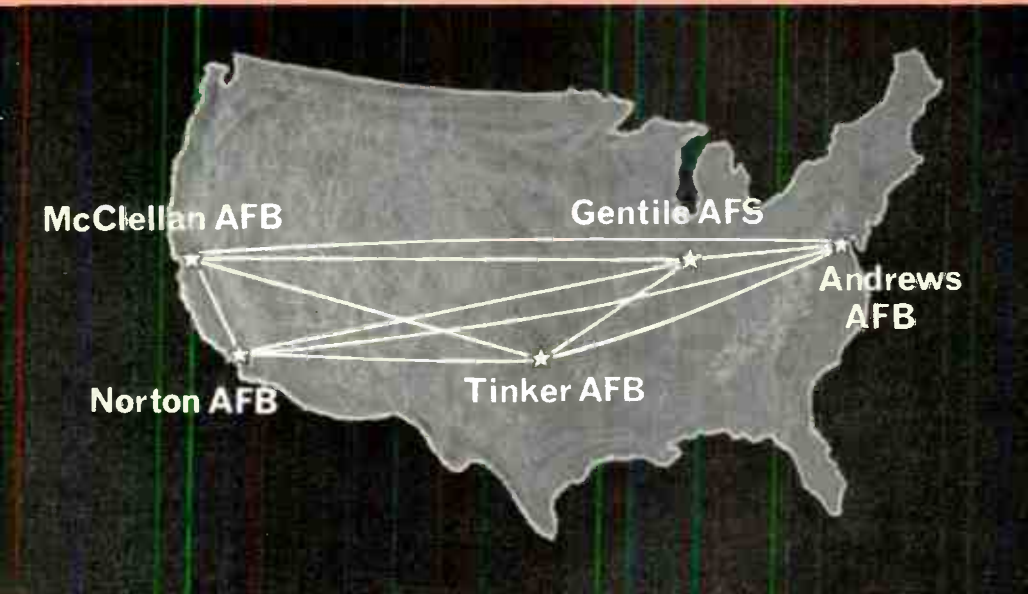


Figure 3. Automatic switching centers are being built around the worl

AUTODIN



Shown in the United States are the initial installations now in operation.

over AUTODIN at rates anywhere from 75 bits per second to as high as 4800 bits per second. The system will have the equivalent capacity of more than 160,000,000 words daily, and may be expanded above this.

AUTODIN Switching

To provide a smooth and efficient flow of information through the network, AUTODIN centers utilize both circuit and message switching. Circuit switching is the common practice of connecting the calling party directly to the person (or machine) being called. However, to improve handling of large volumes of digital traffic, message switching allows storage of incoming messages until they can be forwarded to the next switching center, or to their ultimate destination. In this way, as soon as a switching center has relayed a message to another center, it is free for a new incoming call. The technique avoids long delays under "busy conditions", and prevents worldwide circuits from being tied up while one message is being completed.

The AUTODIN switching center is similar to that of AUTOVON, although its needs and functions are specialized for handling data-type digital signals rather than voice. Overall control of the center is maintained by the communication data processor units. In addition, the circuit and message switching units are interconnected for versatility.

Four types of services can work into AUTODIN through the switching centers. They are identified as compound, magnetic tape, high speed teletype, and

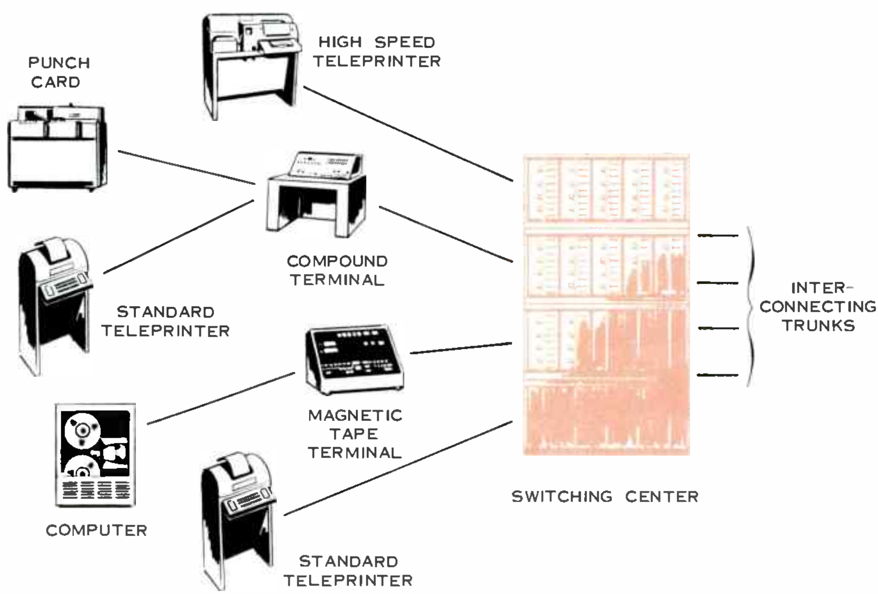


Figure 4. AUTODIN *switching centers work with four types of data terminals. Computer-controlled circuit and message switching give the network the equivalent capacity of over 160,000,000 words daily.*

standard teleprinter terminals. The compound terminal transmits and receives teletype and punch card messages. Magnetic tape terminals serve to connect computers to the system. High speed teletype terminals operate similarly to the compound terminal except that their rate of transmission is higher. These three terminal types require elaborate circuit control methods to insure acknowledgement of messages received. The fourth input to the network is the standard teleprinter terminal.

Leased Facilities

In the United States and wherever else possible, facilities for AUTOVON and AUTODIN are being leased by the Government from existing telephone and telegraph companies. In the total National Communications System, of which DCS is a part, over 30 million channel-miles are devoted to AUTOVON and AUTODIN. Facilities outside the Continental United States are in 96 different countries and are mostly owned by the U.S. Government.

A total of 182 major and minor relay stations will serve the networks, with about 2200 tributary stations connected to them. One of the prime reasons for this large number of stations is survivability through decentralization — a most important factor to the military. Switching centers will be built mostly underground in what are called "hardened" facilities. Trunks in and out of these centers also will be heavily protected. In addition, a multiplicity of transmission means and paths will increase the survivability. Key components along the transmission system will be provided with mobile replacements — in fact, many switching centers will be capable of being airlifted to more secure locations.

Wherever possible, the same trunks will be used by both networks. On a dial-up basis, AUTOVON callers may actually be using AUTODIN trunks to avoid unneeded duplication. However, this in no way affects the justifiable need for redundant circuits in all military networks. It merely eliminates building two facilities over parallel paths in virtually the same locations. Also, it is thought that the increased desirability of secure voice communications will force more and more AUTOVON conversations to the easily coded digital transmission methods of AUTODIN.

Both networks will rely on most every means of transmission available, including telephone cable, microwave, high frequency radio, troposcatter, and undersea cable. The Defense Communications Satellite Progam, established by DCA, is studying the feasibility of satellites in the systems, and has planned for a series of synchronous or near-synchronous satellites to be launched in 1966. In addition to adding more paths to increase survivability, satellites could provide communications to remote areas as quickly as ground stations could be set up.

Conclusion

AUTOVON and AUTODIN have grown from the widespread military and diplomatic forces' need for reliable contact with each other, and their responsibility to centralized command. Technological advances in recent years have made possible the sophisticated machinery to carry out such a task. Indeed, while it will be a number of years before the entire Defense Communications System is ultimately completed, it is surely the most advanced telecommunications network in the world today. The telephone industry, by playing a valuable role in the construction and operation of AUTOVON and AUTODIN, will undoubtedly profit greatly through association with such a project.

GLOSSARY

Listed are a number of terms often encountered in literature about* AUTOVON *and* AUTODIN *and are a supplement to those terms used and explained in the accompanying article.

ADPE—Automatic Data Processing Equipment. As a system, communications and data equipment linked together to process and transmit data.

ANALOG—Literally, resembling something else; the voltages on a telephone line are the *analog* of the original speech. In computers, the principle of performing calculations by measuring voltages, resistances, etc., as opposed to the counting processes in a digital system.

AUTOMATIC ELECTRONIC SWITCHING—High-speed electronics techniques used to control small mechanical switches in completing connections between users. Computer controlled, automatic switching in AUTOVON/AUTODIN permits priority routing, quick restoral in case of failure, dial-up conference calls, and many other services without an operator.

BAUD—The unit of speed in the transmission of binary information, corresponding to one bit-per-second. Standard 100 word-per-minute teletype transmission operates at 75 bauds, or 75 bits-per-second.

BINARY DIGIT—A unit of information in a two-element binary code. Commonly referred to in the contracted form, *bit*.

CIRCUIT SWITCHING—Common central office switching where two telephone lines are connected in order to complete a call.

CONUS—Continental United States.

DIGITAL DATA—Information expressed in numerical values based on some particular base numbering system. The binary system, for example, uses a base of two digits.

EDPE—Electronic Data Processing Equipment.

FULL DUPLEX—A communications system allowing simultaneous transmission in both directions. A telephone system is full duplex.

MESSAGE SWITCHING—A "store-and-forward" capability used in AUTODIN switching centers allowing messages to be recorded, then forwarded at a more appropriate time.

OFF-HOOK—A *hot line* service providing instantaneous and automatic ringing at a distant phone when the receiver is lifted at the local subscriber's set. Also known as *automatic ringdown*.

PABX—Private Automatic Branch Exchange. An automatic version of the PBX switchboard.

SECURE COMMUNICATIONS—Transmission methods employed to prohibit unauthorized persons from gaining access to classified messages. May include cryptographic coding. AUTODIN, using digital coding techniques, can easily be made a *secure* network.

SWITCHING CENTER—The centralized location of automatic switches, where calls are routed, or stored and forwarded as in AUTODIN message switching. Includes switching units and computer processing units.

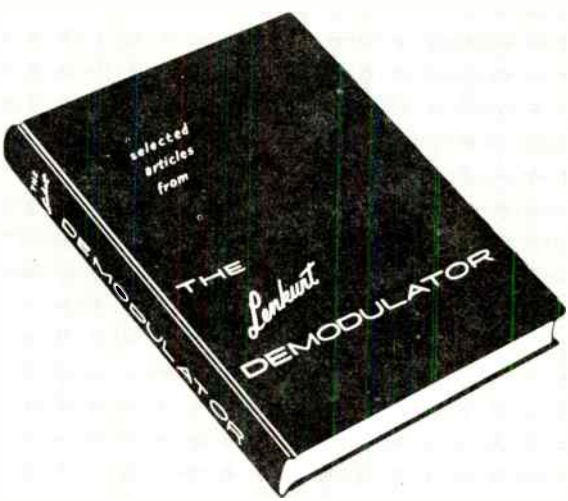
TOUCH TONE DIALING—Multiple-tone dialing technique replacing the common dial pulse system. The telephone instrument has push buttons instead of a dial, and sends various tones corresponding to address numbers.

NOTICE

The present *Demodulator Reprint Book,* published in 1959, is OUT OF PRINT and copies are no longer available.

A NEW *Demodulator Reprint Book* is now being prepared. This larger revised edition will contain selected *Demodulator* articles from issues published through December 1965. The new cloth-bound book will be a valuable source of reference for subjects relating to telecommunications. An announcement will be made in *The Demodulator* when the new book is available.

THE EDITOR
THE LENKURT DEMODULATOR



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THE LENKURT DEMODULATOR is a monthly periodical circulated to technicians, engineers, and managers employed by companies or government agencies who *use and operate* communications systems, and to educational institutions. Job title, company affiliation, and nature of business must be indicated on subscription requests. Each issue features an instructive article on a subject dealing with the science of telecommunications. Permission to reprint articles may be obtained by writing THE EDITOR.

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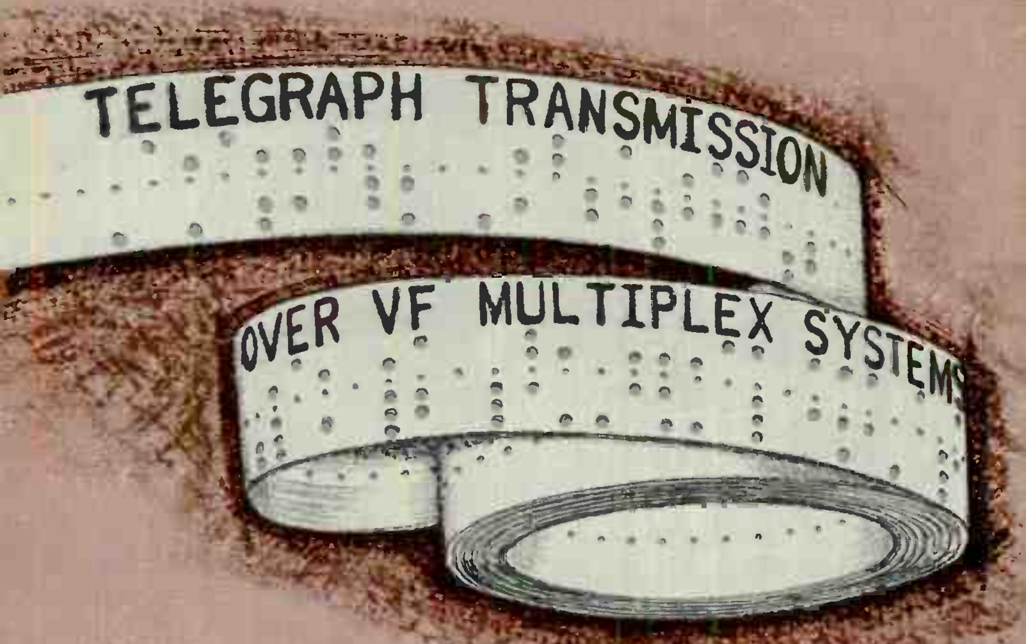


The techniques of modulation and frequency division multiplexing play an important role in the transmission of telegraph messages over voice-frequency channels. This is especially significant in view of the present need for more and more channels in which to handle the increasing amount of business information now being transmitted over telephone networks.

This article reviews briefly the techniques used to convert telegraph pulses to voice-frequency tones and describes several direct-current telegraph loops in common use. Also, the voice channel loading effect of multiplexed telegraph signals is discussed.

Telegraph systems provide a means of transmitting information using electrical pulses which conform to a preestablished code. In earlier days, telegraph messages were transmitted by hand-operated *keys* using the familiar Morse code. Modern telegraph systems, however, use electromechanical machines, called teleprinters, page printers, or tape printers, that employ some type of machine code.

Conventional telegraph machines use the standard 5-level Baudot code and normally operate at transmission speeds of 60, 75, and 100 words per minute, at pulse rates of 45, 57, and 75 bits per second, respectively. A new 7-bit code called ASCII (American Standard Code for Information Interchange) was recently developed and is expected to find wide application in data processing systems as well as for message processing. Telegraph systems currently using the new ASCII code have added an eighth bit to provide a parity check, thus making it an 8-level code. The new 4-row keyboard teleprinters designed to handle the new code operate at a transmission speed of 100 words per minute with a pulse rate of 110 bits per second. These various telegraph machines provide a printed copy of the message or a punched tape which is then used to operate a printer.

Telegraph machine signals consist of a sequence of current and no-current pulses of equal length, known as *mark* and *space*, respectively. Using the 5-level Baudot code, for example, the letter A is indicated by a signal of mark-mark-space-space-space. Before these dc telegraph signals can be transmitted over standard voice frequency communication channels, they must be converted to ac tones. There are two basic methods used to convert the dc loop pulses to tones suitable for transmission over a voice-frequency multiplex channel. These are amplitude modulation, and frequency modulation.

In both AM and FM telegraph multiplex systems, a tone oscillator, in each transmitting channel, is used to provide the necessary voice-frequency carrier. Frequency division is the type of multiplexing ordinarily used and so the carrier frequency in each channel is different.

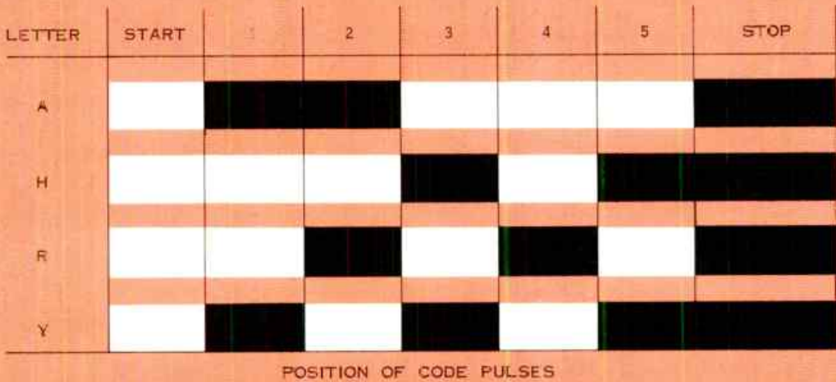
Amplitude Modulation

Amplitude modulation methods are historically related to direct-current telegraphy. In dc telegraph, a battery or other source of direct current is keyed on and off. At the receiving end, the signals are detected by some sort of magnetic device. In AM, the process is similar except that a tone oscillator is keyed on or off to indicate mark and space conditions and, for this reason, is sometimes referred to as on-off modulation.

This method has several disadvantages. It does not use bandwidth efficiently, since two sidebands of the carrier are produced and, unlike single-sideband voice communications methods, the carrier and one sideband cannot be completely eliminated and still do a satisfactory job.

Sidebands are produced when the modulating wave causes the carrier to change from one value or state to another. In voice communications, the modulating waveform is continuous, thus causing modulation products (sidebands) to be formed continuously. If the carrier and one sideband are eliminated, the other sideband remains to convey the modulating intelligence.

In telegraphy, where on-off pulses are the modulating signal, modulation products are formed only during the transition from "on" to "off," and from



60 SPEED	22	22	22	22	22	22	31
75 SPEED	18	18	18	18	18	18	25
100 SPEED	13.5	13.5	13.5	13.5	13.5	13.5	19

PULSE DURATION IN MILLISECONDS

Figure 1. Examples of the 5-level Baudot code for the letters A, H, R, and Y, including pulse lengths for the three standard telegraph speeds. Mark pulses are shown in color while space pulses are shown in white.

"off" to "on." These modulation products are transients whose bandwidth is a function of the keying or switching rate. Except when a pulse is started or ended, no modulation products can appear in the transmission path. Thus, it would be impossible to continuously transmit a steady mark or space.

The information-carrying characteristic of an AM signal is its amplitude. For this reason, AM is particularly vulnerable to impulse noise and changes in transmission level. Impulse noise is particularly disturbing. Noise pulses caused by electrical storms, switching transients, and similar disturbances, may equal or exceed the information pulses in amplitude and duration. Under severe conditions, impulse noise may completely obliterate an AM information pulse.

Frequency Modulation

In FM systems, the carrier frequency is shifted in one direction for a mark condition and the opposite direction for a space condition. A *diode keyer* in the tuned circuit of the tone oscillator changes the circuit resonance so as to shift the tone back and forth between the two frequencies. Such frequency shifting does not occur instantaneously, however. The inherent resonance of the tuned circuit causes the resulting waveform to change smoothly from one frequency to the other. The amount of shift is the same for both directions and varies from about ± 30 to 42.5 Hz depending upon the operating speed of the telegraph equipment. This type of modulation is also referred to as frequency-shift keying (FSK).

Since the mark and space signals are represented by different frequencies of equal strength, amplitude variations have no effect on the signal unless the signal has the same or less amplitude than the noise. This contrasts strongly with amplitude modulation where a mark is represented by the presence of the carrier and a space is represented by the absence of the carrier. Level changes due to fading, noise, and other interference have a strong effect on AM signals. FM systems can tolerate level changes of about 40 to 50 dB, and are about 12 dB less sensitive to impulse noise than AM systems.

Bandwidth

The bandwidth required for a voice frequency multiplex telegraph channel depends on such things as the code pulse rate, noise, filter attenuation to adjacent channels, and whether or not both sidebands are transmitted (AM systems). A bandwidth of 120 Hz is usually satisfactory for 5-level code telegraph signals at speeds up to 100 words per minute for both FM and double-sideband AM systems. The usual bandwidth for 8-level coded telegraph signals is 170 Hz.

Since the required bandwidth is much smaller than that required for speech signals, a normal 3-kHz voice band can be divided by frequency division multiplexing into sub-bands or channels each capable of transmitting a telegraph signal. Approximately 18 channels can be obtained with 170 Hz spacing, while up to 26 channels can be obtained with 120 Hz spacing. This means that up to 18 or 26 voice-frequency multiplexed telegraph signals can be transmitted simultaneously over a single voice channel.

Telegraph Loops

The circuit between the telegraph machine and the multiplex terminal is called a *loop* circuit. Each telegraph loop is made up of two legs which are the conductors (full metallic or ground return) between the terminal points of the loop. In half-duplex operation, the same loop is used for sending and receiving. However, full-duplex operation, which permits simultaneous transmission in both directions, requires both a sending and a receiving loop.

Because of differences in applications and because of the variations in lengths, any one of a number of circuit arrangements may be employed in telegraph loops.

Neutral Loops

One of the simplest and most direct circuit arrangements is the *neutral* or *open-and-close* loop, illustrated in Figure 4(A). The neutral loop requires a battery only at the central office, and the difference between mark and space is determined by whether or not current is flowing in the loop.

When the printer is sending, closing of the printer contacts closes the loop circuit and the current flowing in the loop applies a potential to the multiplex-channel keying circuit. In the receiving direction, the carrier frequencies are applied to a discriminator. In the discriminator, the two frequencies that

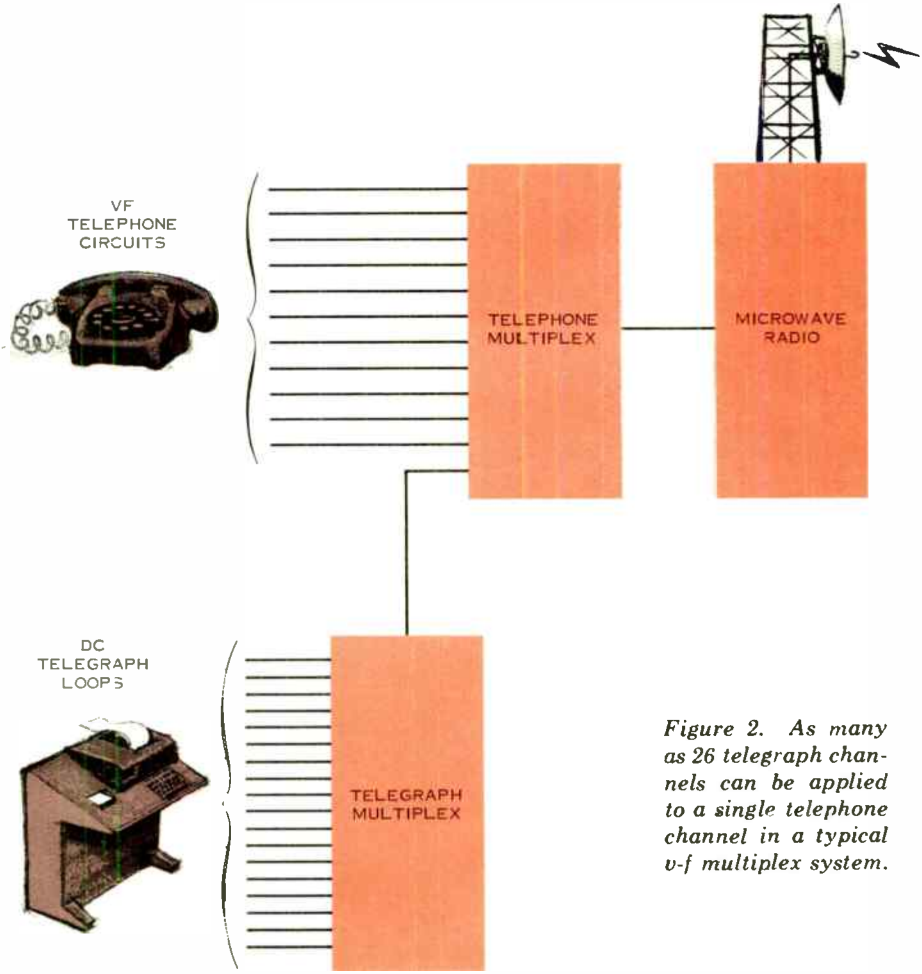


Figure 2. As many as 26 telegraph channels can be applied to a single telephone channel in a typical v-f multiplex system.

are used to transmit the marking and spacing conditions are separated and are rectified to obtain dc for operation of the polar receiving relay. The contacts of this relay open or close the receiving neutral loop to reproduce the transmitted character at the receiving printer.

Balanced Loop

While neutral loops offer the advantage of simplicity, they are restricted to the shorter loops in which either leakage or the distributed capacity of the path does not severely affect the signal. To reduce these problems a balanced loop (also called effective polar loop) may be used. An example is shown in Figure 4(B).

A balanced loop is similar to a neutral loop in that the difference between mark and space is determined by whether or not current flows in the loop. However, the balanced loop differs

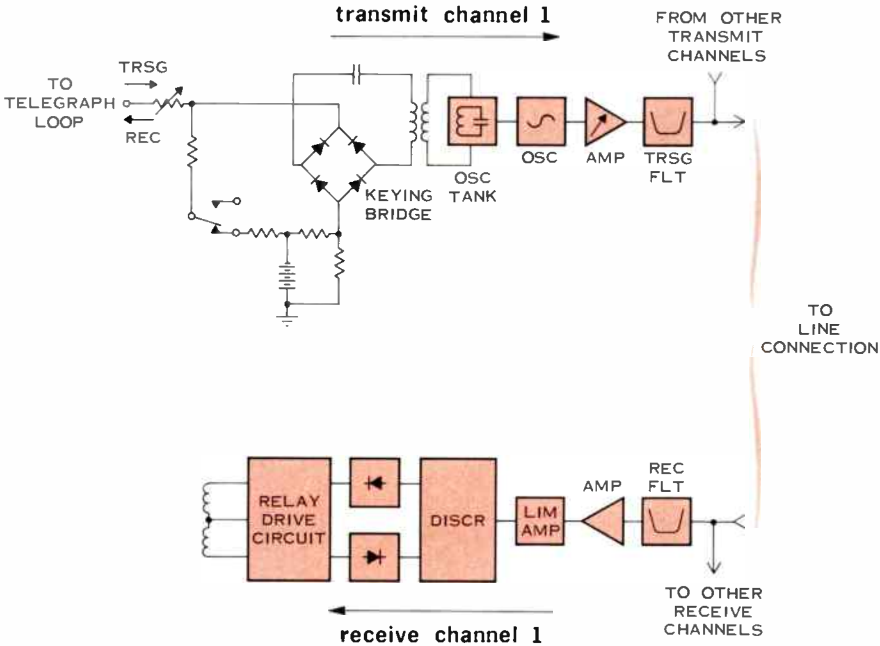


Figure 3. Simplified schematic diagram of telegraph multiplex terminal operating into a half-duplex neutral loop.

from the neutral loop in that a battery potential is applied at the printer location as well as at the central office. The printer battery, in conjunction with the battery potential applied to the marking contact, applies a higher potential to the loop. The increased potential improves the rise time of the marking pulse which tends to increase the length of the pulse. In addition, the increase in potential permits operation over longer loops.

When a spacing signal is received, application of equal potentials to both ends of the loop discharges the line more rapidly than simply opening the loop, resulting in an improvement of the pulse shape. Adjustments can be made in battery potentials to eliminate bias in the loop as required for changing conditions in the loop.

Polar Loop

The most effective transmission method commonly employed is called polar operation. In this case equal currents of opposite polarity are used for the marking and spacing conditions. In addition to the two voltages, this method requires the use of a polar relay in which the direction of current flow in a winding causes the relay to operate to either the marking or spacing position. Since printers normally oper-

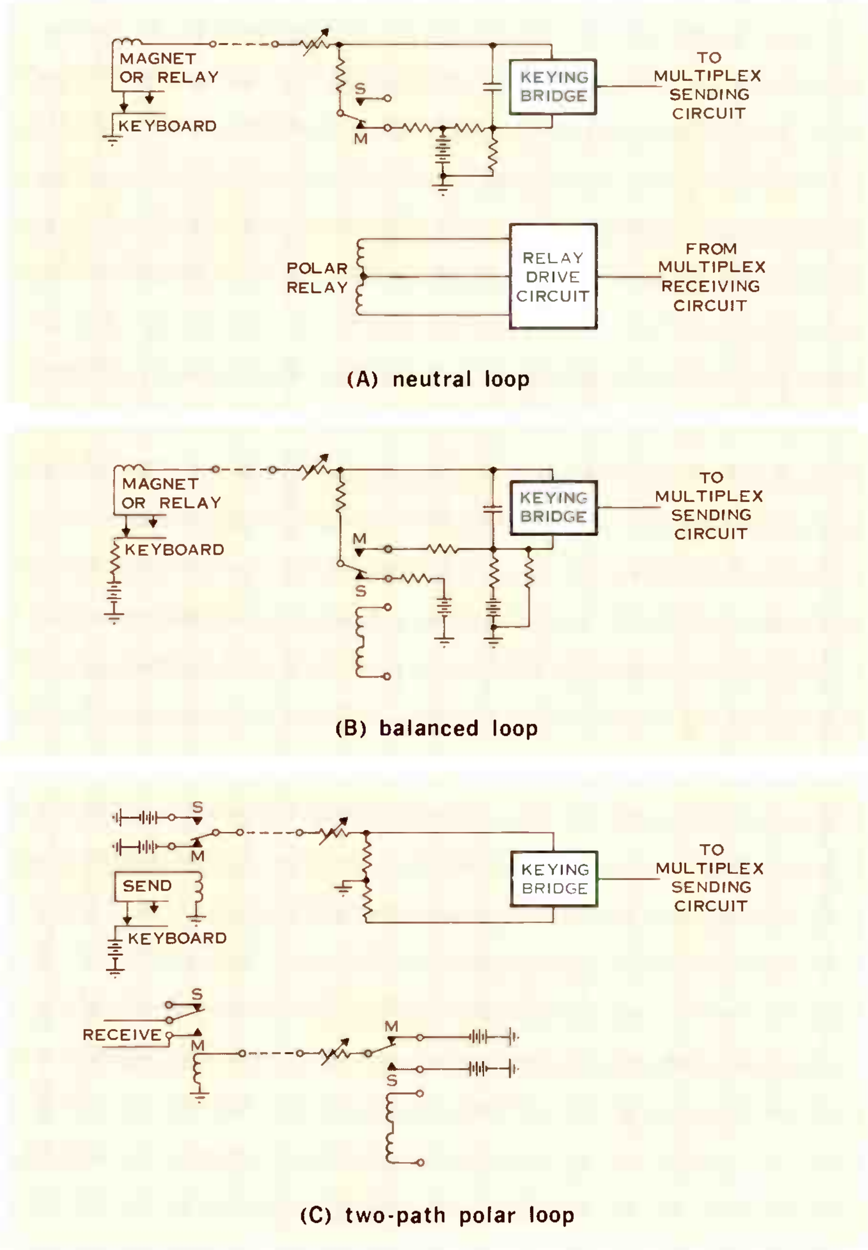


Figure 4. Three types of basic telegraph loop circuits. (A) neutral, (B) balanced, and (C) two-path polar.

ate only from on and off signals, a relay is usually required at the printer location. An example of a polar loop is shown in Figure 4(C). Where battery potentials are the same, the loop characteristics do not change between the sending of a mark or space signal, and if the relay is properly adjusted, the mark and space signals are equal and no bias is obtained.

However, because of the requirement for two batteries, the method is normally only used in transmission from the office to the subscriber, and either neutral or effective polar transmission is used in transmitting from the subscriber to the office.

Break Feature

In a half-duplex loop it is sometimes necessary for the operator at the receiving printer to interrupt the sending printer. This requirement led to the use of an additional relay in the telegraph loop, called the break relay, arranged to accomplish this purpose. The receiving operator may interrupt by opening his loop.

When the receiving loop is opened (effective spacing condition) signals received from the distant terminal are applied to the local-terminal keying circuit, but are inverted. The combination of the retransmitted signals with the original signal causes a continuous spacing signal condition at the sending terminal. When this occurs, the sending operator knows that the receiving operator wishes to interrupt.

Hub Operation

In some telegraph applications, it is occasionally desirable to connect a number of telegraph circuits together in such a way that telegraph signals originating in one circuit are transmitted to all other interconnecting circuits. A method of doing this is through a *hub* board. In this arrangement the dc sides of the multiplex channels are connected together on a high impedance basis. Thus, only a small amount of current is required.

Battery potentials of ±130 volts are required in the hub equipment unit. The hub is supplied with a +130 volt potential through the hub potentiometer. The hub circuitry is such that in the normal marking condition the hub voltage is +60 volts.

The changes in current that result from one circuit sending a space signal into the hub changes the hub potential from +60 volts for marking to −30 volts for spacing. When applied to the sending portion of the remaining channels, these potentials effect simultaneous transmission of the desired signal condition. Three telegraph circuits are interconnected in the simplified diagram of a hub shown in Figure 5. Each circuit is connected to a multiplex channel through a hub-equipment unit.

Hubs may be operated either half or full duplex as with normal telegraph loops. Like the normal telegraph loop, it is sometimes necessary on half-duplex hubs for a receiving operator to break in.

Interruption is accomplished as in the normal loop by a receiving operator sending a spacing signal into the hub. The circuit is arranged so that the hub potential drops to −60 volts when two or more machines are sending spacing signals into the hub. This low potential causes all machines to go to spacing, including the original sending machine, and the sending operator then knows that someone wants to interrupt.

Channel Loading

When transmitting several telegraph tones over a voice frequency channel of a multiplex system, great care must be exercised in establishing the levels at which the signals are applied. Multiplex telegraph signals have greater average power than voice signals. If the power handling capability of the multiplex system amplifiers is exceeded, intermodulation products from the telegraph tones have far greater interfering effect on other channels than do voice signals.

For this reason, a standard signal level is usually specified for voice frequency telegraph signals transmitted over multiplex voice channels. This level is conservative, and is based on the loading effect produced by the maximum number of telegraph channels that can be handled by the voice channel. A common standard per-channel level is −21 dBm at the zero transmission level point. For most applications,

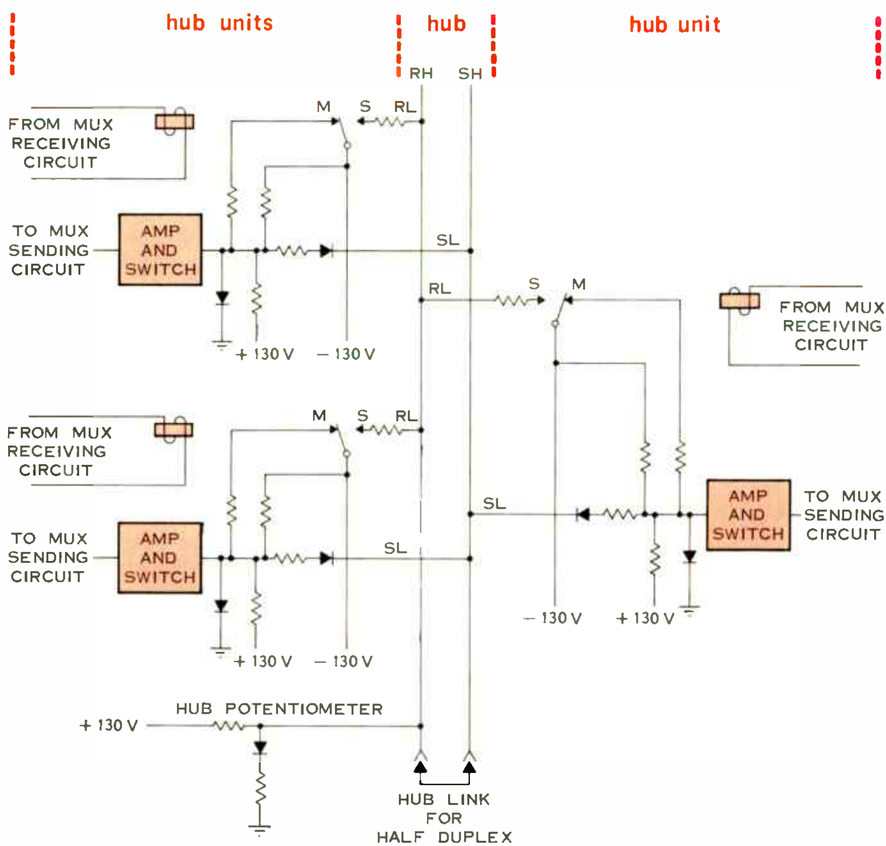
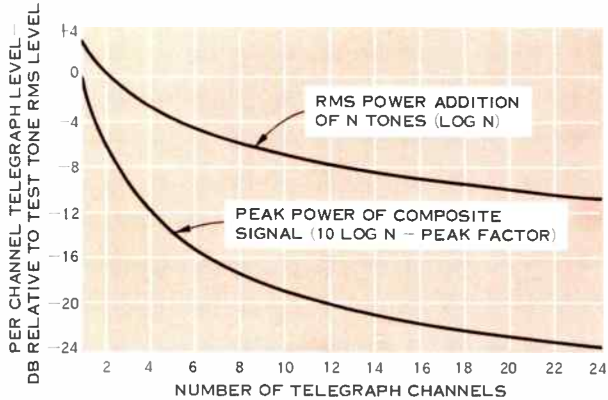


Figure 5. Hub operation, showing three multiplex telegraph channels interconnected on the d-c loop side.

Figure 6. *Theoretical maximum transmission levels for various numbers of telegraph tones transmitted over a multiplex voice channel.*



this level is high enough to provide good service over a voice frequency multiplex channel.

However, in applications where the maximum telegraph channel capacity is not used, it may be desirable to increase the level of each telegraph tone in order to improve the signal-to-noise ratio. The increased level is a function of the number of telegraph tones to be transmitted.

In calculating the loading effect, peak power must be used, since distortion will occur if the peak power exceeds the load handling capacity of the multiplex equipment. When telegraph signals are applied to a single voice frequency channel of a multiplex system, the permissible peak power is normally +3 dBm at the zero transmission level point. This value is assumed in the following discussion.

For a single telegraph channel, the calculation of peak power is straightforward. A sine wave is normally assumed. Peak voltage of a sine wave is 1.4 times the rms value of the wave, or 3 dB greater in power than the rms power value. When only one telegraph channel is involved, the level of the telegraph tone may be equal to the level of the normal test tone, since both signals are sine waves.

As the number of telegraph channels increases, the peak power that the composite waveform may reach also increases. Since there is a possibility that this value can become quite high for a large number of tones, a *peak factor* is used. This peak factor is based on the statistical probability that the peak power of a complex wave will almost never add up in such a way as to exceed the sum of the rms value of the wave and the peak factor. For a single tone, the peak factor is 3 dB. Peak factor increases as the number of channels is increased, reaching a maximum of 13 dB for approximately 20 channels.

As an example, assume that ten telegraph channels are to be applied to voice frequency multiplex channel normally adjusted to a −16 dBm test tone level. In this example, peak power should not exceed −13 dBm. Each telegraph channel transmitting level must be lower than −13 dBm by the sum of the combined power of the ten tones (rms power addition) and the peak factor.

First, the combined tone level is calculated by taking ten times the logarithm of the number of channels (10 log 10 = 10 dB). Adding a 12-dB peak factor to this 10-dB level gives a peak value 22 dB above a single channel peak. The per-channel transmitting power is then obtained by subtracting the 22-dB peak level from the maximum permissible level (—13 dBm minus 22 dB = —35 dBm). Similar calculations may be made for different numbers of telegraph channels. Figure 6 shows how the telegraph tone levels must be reduced as the number of channels increases. It is important to note that these calculations yield *theoretical* maximum levels for telegraph tones and pertain to the loading of a *single* voice channel in a multiplex system.

Conclusion

The transmission facilities provided by telephone communications systems constitute a vast network which is capable of interconnecting locations almost anywhere in the world. Although these facilities are made up in many forms and have different types of transmission media, they do have one very important thing in common — the standard voice frequency channel, which has a useful bandwidth of about 3 kHz.

While this vast network of multiplexed telephone channels was designed primarily to handle speech signals, the circuits can be used to transmit other forms of information such as telegraph. The techniques of modulation and multiplexing provide a practical means of converting the dc telegraph signals to ac tones suitable for transmission over telephone circuits.

Through the use of frequency division multiplexing, as many as 26 narrow-band voice frequency telegraph channels can be derived within a single 3 kHz telephone channel.

Such efficient use of a single telephone channel is a tremendous asset in view of the present growth of machine communication to process business information.

Lenkurt Electric Co., Inc.
San Carlos, California

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The Lenkurt 25A is a versatile telegraph and data carrier transmission system. It can handle a variety of multi-channel telegraph applications with speeds up to 100 wpm or data applications with speeds up to 200 bps. The 25A works over many types of transmission facilities such as open-wire, cable, or radio multiplex — or 2-wire or 4-wire physical circuits. More information about this high-performance system is contained in Form 25A-P4, available on request.

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MARCH, 1966
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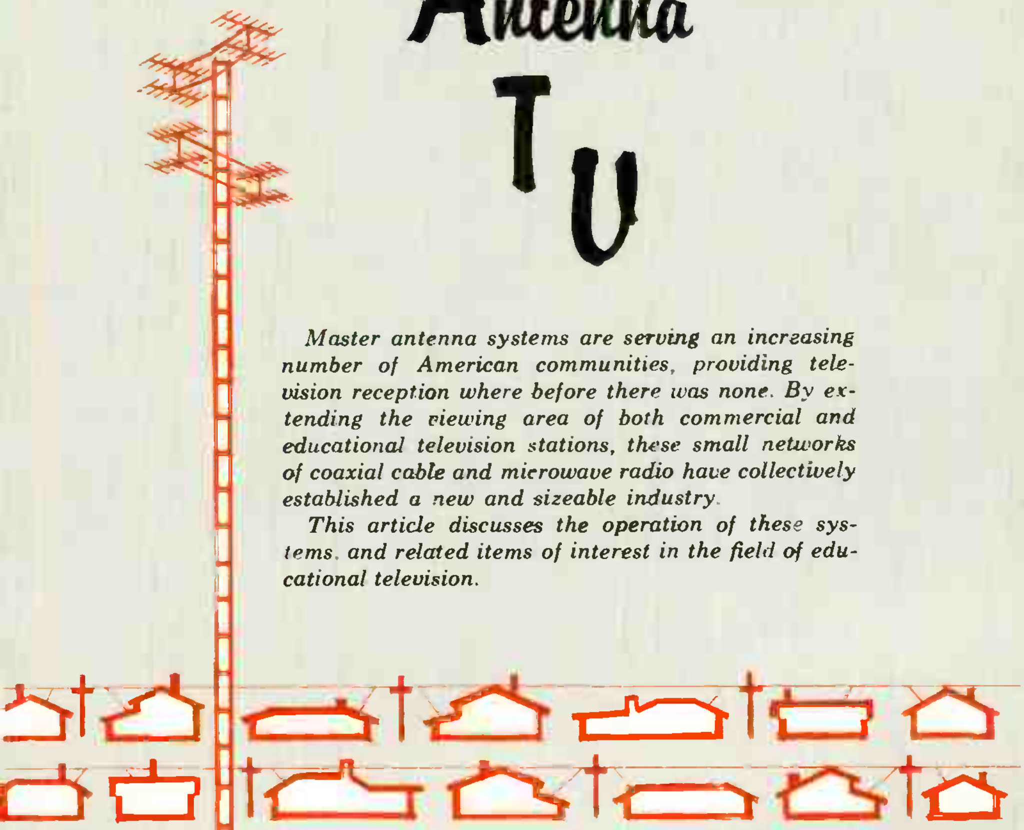
Community Antenna TV



Master antenna systems are serving an increasing number of American communities, providing television reception where before there was none. By extending the viewing area of both commercial and educational television stations, these small networks of coaxial cable and microwave radio have collectively established a new and sizeable industry.

This article discusses the operation of these systems, and related items of interest in the field of educational television.

The establishment of the television industry in the United States brought with it new concepts of entertainment, news, and education, adding immediacy and depth. At the moment there are over 700 television stations in this country (Figure 2), and over 100 construction permits have been granted by the Federal Communications Commission to establish new stations.

Simultaneously with the growth of television — and almost unnoticed during its early years — cable TV systems appeared, stretching TV coverage into otherwise poor signal areas. CATV, for Community Antenna Television, is now a sizeable industry of its own serving close to two million U.S. homes. Similar systems also are developing in other countries, including Canada, Mexico and Great Britain.

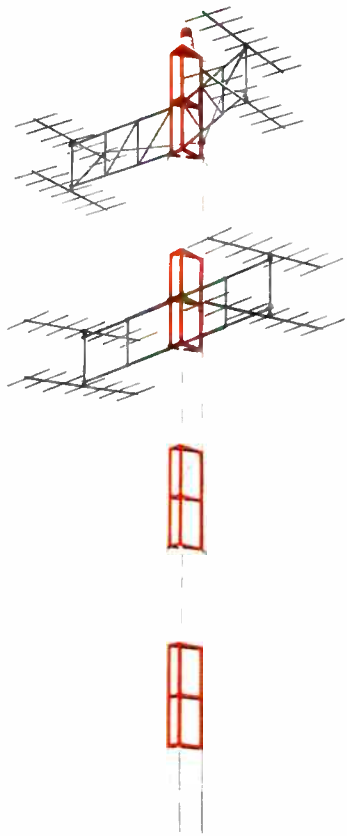


Figure 1. High gain Yagi antennas, one for each television station, receive off-the-air signals to be relayed through CATV system.

A Beginning

CATV is essentially a master antenna service for receiving television signals and distributing them to home receivers. When the first television stations went on the air in the late 1940's, it was found that signals in outlying areas were not always powerful enough for satisfactory reception. Potential TV viewers were either too far from the broadcasting station, or were in a shadow area behind a nearby mountain or other obstruction. Even the construction of costly roof-top antennas was not always successful.

The first meager steps toward the new CATV industry were made by local citizens joining forces to construct master antennas on nearby hilltops. The signal was carried down the hill by standard TV lead-in wire strung from tree top to fence post to pole, and interrupted regularly with unsophisticated booster amplifiers. The results were not always ideal. The first commercial installations, with better antennas and coaxial cable came in 1950. Soon equipment manufacturers were entering the field with specially designed CATV receivers, cable amplifiers and other components. When greater distances separated the TV station from the community, microwave radio replaced long coaxial cables as the most economical method of insuring good TV reception.

CATV Today

Today, approximately 1600 CATV systems are in operation, another thousand have the go-ahead from local franchising agencies, and over two thousand are pending approval. However, it should be understood that not all of these will be built immediately or even in the near future. Some predict about 100 new systems for 1966, increasing to a total of about 2600 by the end of 1970. Only one state does not have at least one operating system, but even there applications for service have been filed. Some larger states have hundreds of independent systems.

CATV systems vary greatly in size and capability, from small operations carrying as few as two channels, to advanced and more elaborately equipped facilities bringing as many as 12 TV channels and a number of FM radio signals to the subscriber. The average subscriber receives five stations, while less than one percent get 10 or more channels. Four percent receive only two channels. Soon equipment advances may make it possible for a CATV system to carry 20 or more channels. Small operations may have only a hundred or so subscribers, while the largest in the United States serves nearly 20,000.

	VHF	UHF	TOTAL
COMMERCIAL	486	101	587
EDUCATIONAL	66	49	115
TOTAL	552	150	702

Figure 2. Numerical breakdown of TV stations now on the air.

In its early years, CATV served the small population centers scattered some distance from television stations. More recently, however, CATV has been brought to the doorsteps and even into the parlors of major cities like New York, Los Angeles, and San Francisco. Tall buildings, natural obstructions, airplanes, and other such factors will degrade television signals from even nearby stations. The advent of color television also has increased the need for high-grade signals for satisfactory picture reproduction.

In addition to providing improved TV reception, CATV frequently includes bonus services placed on otherwise unused channels. An example is weather information from a camera continuously scanning temperature, wind, and other gauges. Another service allows home viewers to read the latest news as it is typed on news-wire machines.

At least one operator has gone further than that, setting up television-like studios to provide news, discussions, speeches, children's programs, and even live sports events. Equipment includes mobile units, video tape recorders, and professional studio consoles. Commercial background music also may be supplied by CATV, using already installed cables to carry recorded music to business concerns.

The CATV Signal

The first need of a CATV system — like the home receiver — is a good signal from the broadcast station. In some countries, broadcasters will provide a direct program feed from the station. But American CATV operators pick up television signals "off the air" with specialized receiving equipment. High gain antennas, typically of Yagi design, are situated on a mountain peak or other advantageous point in the terrain. These antennas are selective, narrow band devices, most efficient at only one frequency or channel. Therefore, a separate antenna is usually installed for each channel to be received. Ideally, TV signals at the antenna site should have a minimum strength of 50 microvolts per meter.

Special receivers detect the TV signals, convert UHF to VHF if necessary, and amplify them to suitable levels for transmission. This portion of the CATV system is known as the "head end" equipment. If the signals are to be fed directly into a cable trunk line, standard VHF television frequencies are used. However, if a local VHF station is carried on the cable, interference will usually result between the direct signal from the transmitter and the signal on the cable. In such cases it is necessary to translate this station's programs to a different channel prior to distribution.

When distant TV stations are to be carried, it is often more practical to use microwave radio links over sometimes as much as hundreds of miles to reach distribution trunk cables. For the head end equipment to feed a microwave system, incoming TV signals first must be demodulated to more usable frequencies. That is, the carrier frequency must be removed, leaving only pure video information in the range from 10 Hz to 4.2 MHz (Figure 3) with the audio on a subcarrier at 4.5 MHz. This is called a *composite signal*, and may be used to directly modulate the microwave radio. It is also possible to separate the video and audio signals at this point and place the sound signal on a higher frequency program channel.

The output of the microwave radio is transmitted by highly directive parabolic antennas to receiving stations 20 or 30 miles away. Then the signal may be retransmitted to another repeater, or fed into more head end equipment for cable distribution.

The Federal Communications Commission regulates all radio frequency allocations, and recently created a new Community Antenna Relay Service (CARS) for exclusive use by all CATV operations. The band is from 12700 to 12950 MHz.

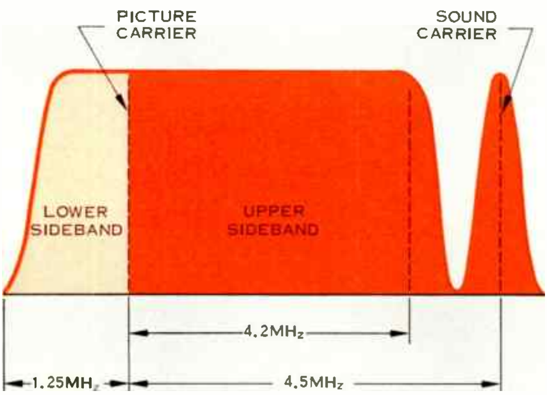


Figure 3. Television spectrum. Upper sideband of video, plus audio carrier make up composite signal.

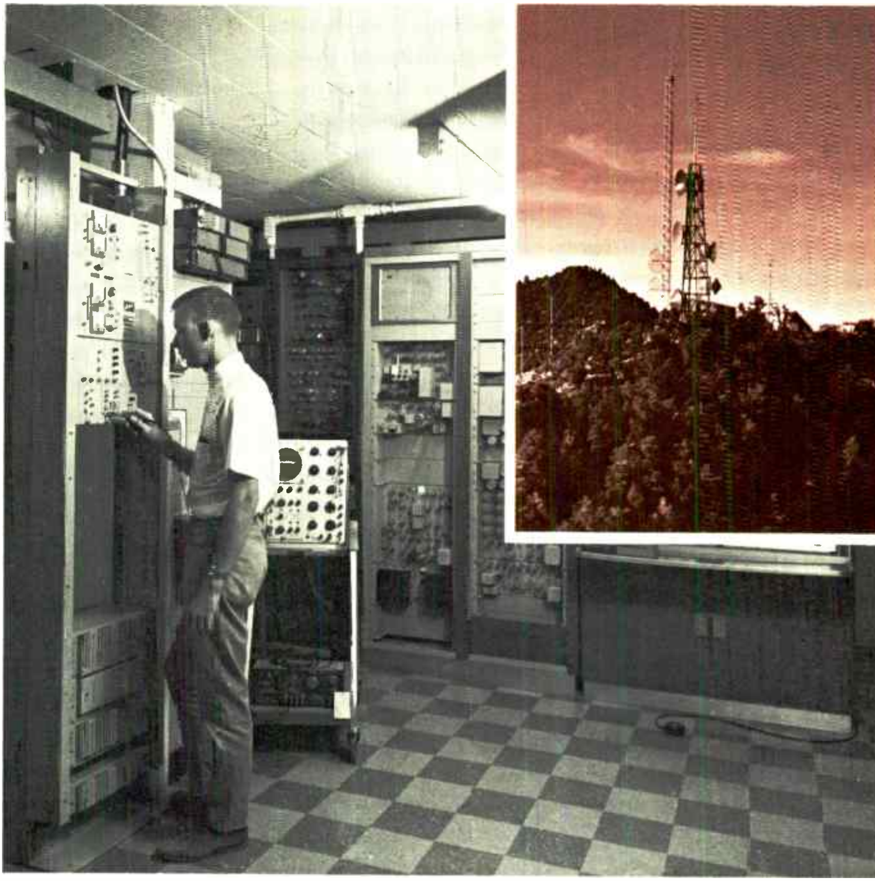


Figure 4. Lenkurt 76TV microwave radio in typical installation. Repeater stations (inset) receive and retransmit signals.

Approximately 25 percent of the CATV systems in the United States use microwave radio, the typical system requiring two or three hops to bring the signal to the cable distribution point.

Picture Distortion

The transmission of television, especially color television, by microwave includes a number of critical problems. Distortion, poor frequency response, and other transmission irregularities all tend to degrade the quality of the final picture image. Of particular importance is the extreme sensitivity of television signals to non-linear phase shift. Ideally, the entire system should be free of non-linear phase shift from almost zero frequency to at least 4.5 MHz (to cover the bandwidth of a video signal). In practice, this is difficult if not impossible to achieve. Components in the

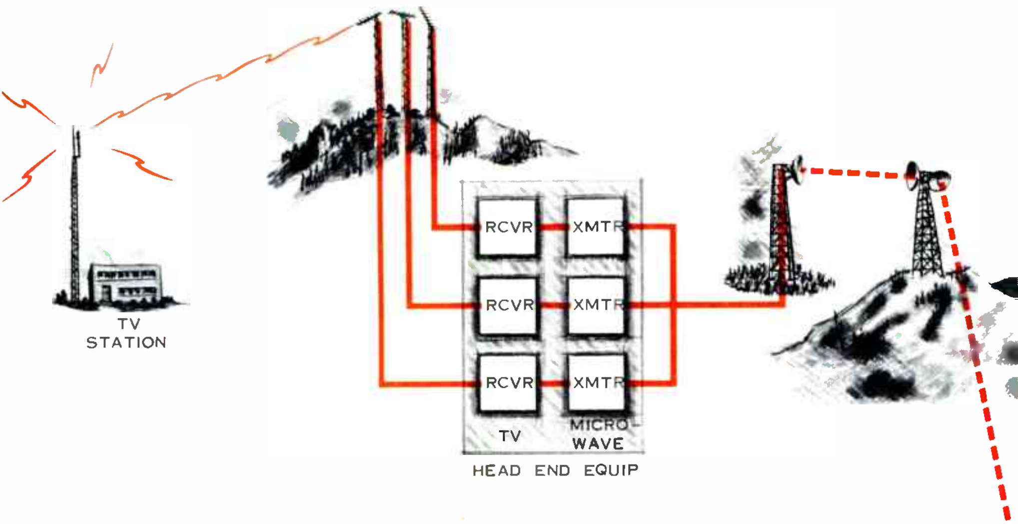


Figure 5. CATV system picks up signals outside the reach of home antennas; may use microwave radio to relay them to cable distribution point. Utility poles carry coaxial cables through residential areas.

system delay some frequencies more than others, distorting the waveform. Although delay distortion of speech or music is not readily detected by the ear, similar distortion of a television signal is very noticeable and grossly affects the quality of reproduction.

Color television is particularly vulnerable to *differential phase* and *differential gain.* The color appearing on the screen is determined by the exact phase relationship between two signals, the color burst and the color subcarrier. An unintentional shift in phase results in a different hue of color. Similarly, change in amplitude of the signal determines the saturation or *richness* of the color. (For additional discussions of these areas see the *Demodulator,* February, 1962; October, 1963; November, 1963; January, 1965).

Delay distortion is directly related to the bandpass characteristics of the entire transmission system, including head end equipment, microwave links, and cable facilities. System design must provide for a very wide bandwidth free from irregularities well beyond the actual frequency limits of the television signal itself. For example, the Lenkurt 76TV microwave system (Figure 4), designed specifically for television transmission, has a frequency response of ±0.5 dB from 20 Hz to 5.5 MHz.

Subjective testing has shown that phasing errors of 5° or more will be detected by the viewer as a change in hue. Likewise, he will find a 2 dB change in color saturation objectionable. In the 76TV, differential phase is less than 0.5° per terminal, while differential gain is held to 0.2 dB at up to 90 percent of applied picture loading.

Ultimately, the signals must be fed into the cable trunk line for distribution to home TV sets. In a system not using microwave, this occurs immediately after the signals are received by the master antenna. With microwave, more head end equipment is found at the final radio hop. Here, signals must be brought to proper levels, remodulated to VHF frequencies, combined, and fed into the main trunk lines (Figure 5).

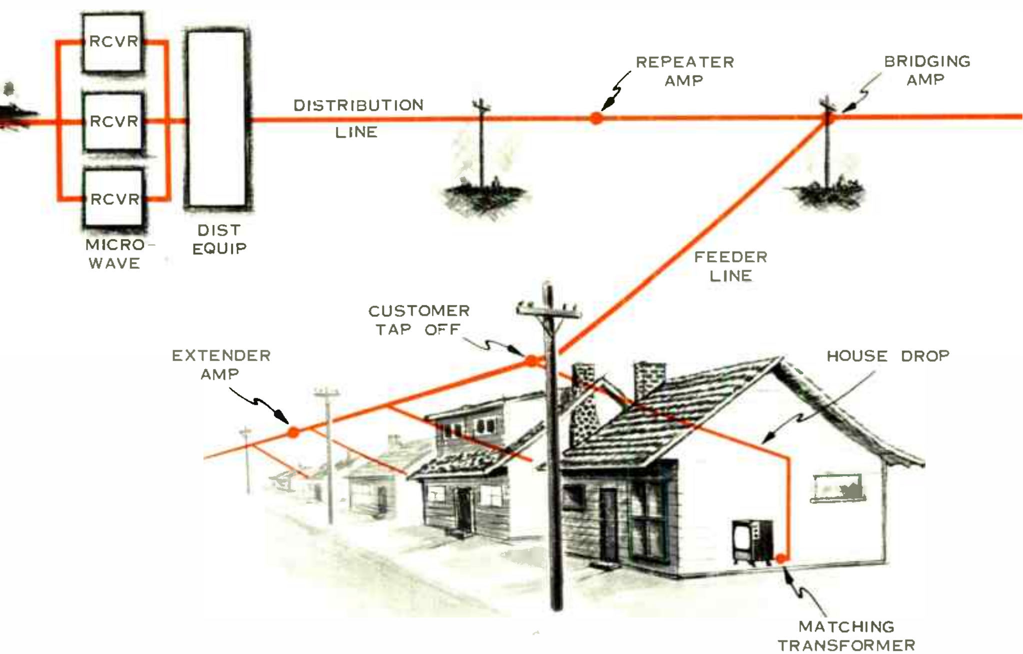
Cable System

The trunk line is the basic carrier of the CATV system and is never tapped to feed individual subscribers. At the intervals along the trunk line are a number of repeater amplifiers (Figure 6) to compensate for signal loss. These are usually less than a mile apart. Bridging amplifiers divert the signals onto feeder, or distribution lines.

Customer "tapoff" units (Figure 7) are placed along the feeder cables. These cause a slight disturbance on the line and therefore a limited number — usually 30 to 40 — are allowed on one line. Extender amplifiers, spaced about every 600 to 700 feet, are used to boost the signal along the feeder line. From the tapoffs come the house drops leading to the subscriber's TV set. However, before a connection can be made, the cable impedance of 75 ohms must be matched to the 300 ohm input impedance of the set through a matching transformer, placed on or near the back of the set.

Troposcatter

CATV operators in other countries have added their own variations to the methods of signal transmission. In Canada, for example, military-developed techniques of troposcatter are being used in some systems spanning rugged terrain. Dependent on the ability of the troposphere to diffuse or scatter a por-



tion of a high frequency signal well beyond the horizon, tropo systems send VHF television skipping over distances from 100 to 500 miles (Figure 8). Stationary tropo antennas may include a tower-supported wire mesh reflector 50 to 60 feet high, almost 300 feet wide, stretching in a parabolic curve around the antenna fixture mounted on the head end amplifiers. Economy is the prime justification for the technique — both the home receivers and the transmission wire are less expensive.

In the British system, head end equipment supplies approximately 40 watts of video power to the trunk lines, which may be up to 6000 yards long. Feeder lines may branch off the trunk lines for

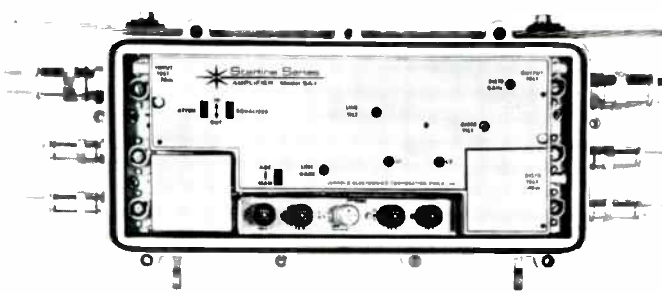


Figure 6. CATV signals pass through trunk-line and extender amplifiers before reaching the home receiver.

end building. These antennas are highly directional, and have good ability to reject co-channel and adjacent channel interference.

British CATV

In Great Britain CATV is called "wired broadcast" or "communal aerial system", and uses two different methods of signal transmission. One system is essentially the same as that used in the United States, relaying signals at standard VHF TV frequencies directly to the receiver. Another popular technique is an outgrowth of the older "wired radio" system. This radio relay system was basically a public address system supplying audio directly to speakers in the home. The television version transmits unmodulated video signals (3-10 MHz) over twisted wire pairs to TV receivers built without the customary r-f front-distances up to about 2000 yards. It has been found that four video signals with their accompanying audio, and four additional radio channels may be carried on two twisted pair (four wires) in a shielded cable.

Educational TV

Sharing some of the problems, and related in many ways to CATV are the three overlapping areas of educational television (ETV), instructional television (ITV), and closed circuit television (CCTV).

ETV is generally meant to include non-commercial broadcast stations, both VHF and UHF. ITV refers to program content rather than facilities, and relates directly to formal education. CCTV describes the transmission of television by cable or microwave to a predetermined audience, as opposed to public broadcast. Additionally, our reference here is primarily to the use of CCTV in education.

There are four general types of licensees operating educational television stations: (1) universities, (2) public school systems, (3) statewide ETV commissions, and (4) non-profit "community" corporations. More than half the ETV stations in the country fall into the first two categories, being directly responsible to educational institutions. Likewise, a statewide commission's prime interest is usually with the school systems of the state. And while the community stations may have no direct connection with schools, they usually carry a regular schedule of instructional programs.

The average ETV station broadcasts 5 or 6 days a week, 10 to 11 hours a day. Programs are divided almost equally between classroom instruction and more general programming planned for home viewing by all age groups. Instructional material more likely will be seen during the normal school hours, with more general programs in the early evening, and informative discussions or entertainment features for adult viewing in the late evening.

At this time there are 115 ETV stations on the air, with another 65 under construction or with applications pending. Currently more than half of the ETV stations are on VHF frequencies (channels 2-13), but most reserved allocations for the future are in the UHF band (channels 14-83).

In the School

Closed circuit television is utilized by many schools to make more advantageous use of teachers and instructional material. There are about 800 CCTV installations in this country, split almost equally between elementary and secondary schools, and colleges and universities. These may operate within one school, delivering lectures or demonstrations to other buildings, or between various schools in a district. Within a single school, coaxial cable easily connects the cameras and studio equipment

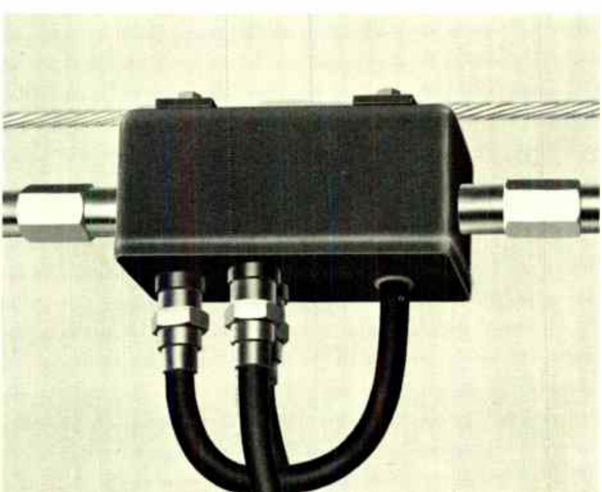


Figure 7. Tapoff units connect house drops to feeder lines, and are designed to prevent interfering signals from reentering cable.

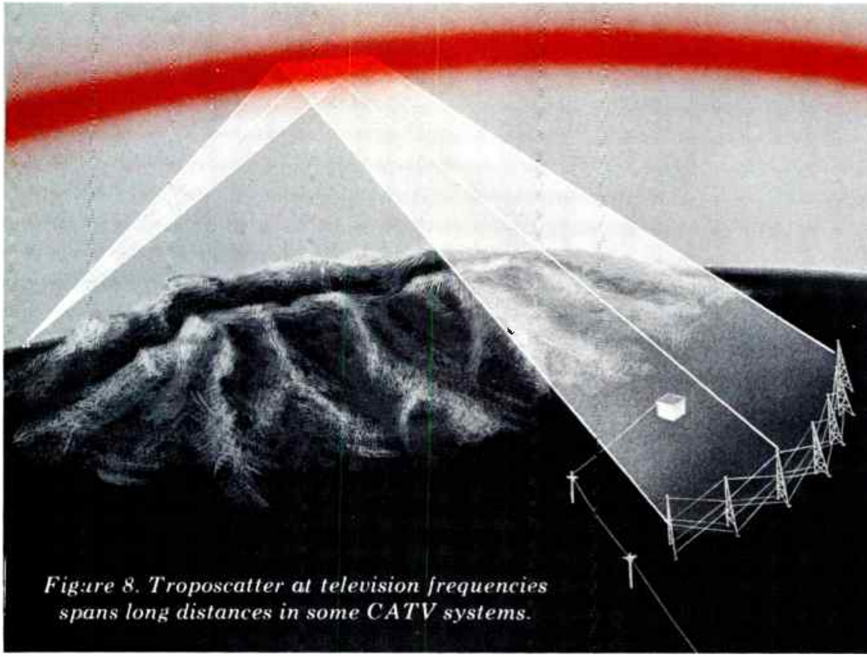


Figure 8. Troposcatter at television frequencies spans long distances in some CATV systems.

to other viewing locations, sometimes with three or four programs on a single cable. Longer runs between separate schools can also be practical, using telephone company cables on a leased basis.

An increasing trend in ITV is the use of microwave radio. The FCC has allocated 31 channels in the 2500 to 2690 MHz band for use by educational institutions. Some districts use these channels as direct links between two schools. Others operate a central transmitter beaming programs in several directions at once, much like a standard broadcast station, to be received off the air at various schools within the district.

ITV systems also operate point-to-point microwave relays on two higher frequency bands. The primary allocation is in the 12200 to 12700 MHz band. However, the FCC will consider applications on a case-by-case basis for the 6575 to 6875 MHz band when the operator can show that it is not technically feasible to use the higher frequency.

Frequently CATV systems will carry ETV programs, thereby greatly extending the range of the station. These may even be piped into the schools, hospitals, or other such facilities in distant towns for little or no charge. In many cases CATV operators also will allow two ETV stations to share programming over a spare microwave channel.

Networks

Many states have already installed widespread microwave networks connecting educational institutions hundreds of miles apart. Similarly, moves have been made to connect large numbers of ETV broadcast stations into educational networks. And many CATV systems are beginning to resemble small networks. It is possible that someday a combination of these efforts will bring to this country a "fourth" major television network joining the best educational and cultural programs in all parts of the nation. Moreover, a fifth network, with commercial UHF stations linked from one end of the country to the other, is being considered.

There are also other possibilities for bringing educational and cultural programs to larger audiences. One quite successful experiment has been undertaken by Purdue University, transmitting previously video taped programs from specially equipped airplanes circling 23,000 feet over Indiana. Daily, over a half-million students in six states (a total of 127,000 square miles) receive courses ranging from elementary to college-level subjects.

From the beginning, telephone companies have been involved with CATV systems, allowing cables to be strung on their utility poles. More recently, telephone companies have supplied cable transmission channels for CATV systems and instructional TV operations on a lease or tariff basis. And now many operating companies are expressing interest in becoming CATV operators themselves.

The Future

In the next few years both CATV and educational television undoubtedly will experience many changes. Advancing techniques will allow for greater numbers of channels to be carried over microwave and cable facilities, bringing even more programs into homes and schools across the nation. Satellite technology may add a new dimension with the possibility of broadcasting directly to schools — or even home receivers — anywhere in the nation.

In the United States close to 98 percent of the homes have at least one television set. Three percent of these homes are served by CATV. Educational programming is now available to an estimated 130 million people—another 10 million to be added this year with 14 new ETV stations. In addition, instructional television today reaches two out of three of the nation's 50 million students.

As new networks link one station or relay system to another and new operations spring up, a continually expanding measure of entertainment and education will be easily available to the television public.

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THE LENKURT DEMODULATOR is a monthly periodical circulated to technicians, engineers, and managers employed by companies or government agencies who *use and operate* communications systems, and to educational institutions. Job title, company affiliation, and nature of business must be indicated on subscription requests. Each issue features an instructive article on a subject dealing with the science of telecommunications. Permission to reprint articles may be obtained by writing THE EDITOR.

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the Lenkurt Demodulator

dB and other logarithmic units

Decibels and many other logarithmic units are used extensively in the telecommunications industry to define the qualities and functions of transmission circuits. In view of the significance of these terms, the Demodulator is treating the subject for the third time with an increased emphasis on understanding the usefulness of logarithmic terms.

This article explains the most commonly used dB terms, and includes a number of convenient conversion formulas.

Special Reprint Book notice on back page

Dealing with very large and very small numbers is often necessary in the telecommunications industry. For example, the frequency of a typical microwave radio is 12,000,000,000 cycles per second. Or, the power represented in conversational speech is measured at about 1/100,000,000,000 watts/cm^2. These are obviously unwieldy terms.

Powers of Ten

However, there are various methods of handling them conveniently. Expressing numbers as *powers of ten* is a first step to simplicity. We know that $10 \times 10 = 100$, and can be written 10^2. Likewise, $10 \times 10 \times 10 = 1000$, or 10^3. By definition, the exponent 3 means that the number 10 is used as a multiplier 3 times. 12,000,000,000 cycles per second then becomes 12×10^9 cycles per second (or 12 GHz).

Note that $10^1 = 10$; $10^0 = 1$. Numbers smaller than 1 also can be treated using powers of ten. By definition, 10^{-1} is the same as $1/10^1$, or simply 1/10. In this way, the power rating for conversational speech mentioned previously can be written 10^{-11} watts/cm^2.

When discussing two relative values, it is sometimes convenient to use the term *orders of magnitude*. This is only another way of expressing powers of ten. That is, one order of magnitude (10^1) is 10 times as much; two orders of magnitude (10^2) is 100 times as much. Simple division indicates that a plane flying 1000 miles per hour is 100 times faster than a horse traveling at 10 miles per hour. It could be said that the plane is two orders of magnitude faster than the horse. Notice that orders of magnitude are really concerned with the exponent of the number. If a number is 1000 times greater than another, $1000 = 10^3$, or *three* orders of magnitude greater.

Logarithms

All of the figures in these examples have had the same "base" number of 10. If we treat the exponent of the base number separately, another useful shorthand is achieved, called *logarithms*. In $100 = 10^2$, the logarithm of 100 is 2. That is, the common logarithm (abbreviated Log_{10}) of a number is the power to which the base 10 must be raised to produce the number. The written form is $\log_{10} 100 = 2$. In practice the subscript $_{10}$ is usually eliminated when referring to common logs. Another log system used in mathematics has a base number of 2.718, and is written $\log_e$ or ln.

The use of logarithms is advantageous in many forms of complicated calculations. Remember that to multiply like numbers, it is only necessary to *add* their exponents ($10^2 \times 10^3 = 10^5$); to divide, *subtract* exponents ($10^5 \div 10^3 = 10^2$). Logarithms are used in the same way. Multiplications and divisions involving large numbers may be carried out by adding or subtracting the corresponding logs and then converting back. In fact, any series of events involving multiplication or division, if expressed logarithmically, may be handled by simple addition and subtraction. This is particularly valuable in the telecommunications industry, where a variety of measurements are necessary to describe the properties of a signal as it passes through the system. Voltages, currents, and powers are measured, noise identified, and losses assessed. These are all made much easier by the use of the logarithmic system.

Decibels

The basic unit of measure in communications is the *decibel,* derived from the less practical unit, the bel, named in honor of Alexander Graham Bell. A *deci*bel is a tenth of a bel.

DECIBELS	POWER RATIO
1	1.259
2	1.585
3	1.995
4	2.512
5	3.162
6	3.981
7	5.012
8	6.310
9	7.943
10	10.0
20	100.0
30	1000.0
40	10,000.0

Figure 1. The Relationship Between Decibels and Power Ratios.

Early experimentation proved that a listener cannot give a reliable estimate of the absolute loudness of a sound. But he can distinguish between the loudness of two *different* sounds. However, the ear's sensitivity to a change in sound power follows a logarithmic rather than a linear scale, and the decibel has become the unit of measure of this change. A difference of 1 decibel, abbreviated dB, in the power supplied to a listening device produces approximately the smallest change in volume of sound which the normal ear can detect. The relationship between any two power values can be calculated in decibels as:

$$dB = 10 \log \frac{P_1}{P_2}$$

where

P_1 *is the larger power*

It should be emphasized that a given number of decibels is always the relationship between two powers, and not an absolute power value by itself (Figure 1). For example, the gain in an amplifier, or the attenuation of a pad, can be expressed in decibels without knowledge of the input or output power of the device.

dBm

Frequently, it is convenient to represent absolute power with a logarithmic unit. One milliwatt is generally accepted as the standard reference for such purposes in the telephone industry, and signal powers can be written as being so many dB above or below this reference power. When this is done, the unit becomes dBm, in the expression:

$$dBm = 10 \log \frac{P_1}{P_2}$$

where

$$P_2 = 1 \text{ milliwatt}$$

By adding a definite reference point, dBm becomes a measurement of absolute power, rather than just a ratio, and can readily be converted to watts. 10 dBm indicates a signal 10 times greater than 1 milliwatt, or 10 milliwatts; 20 dBm is 100 times greater than 1 milliwatt, or 100 milliwatts. A 30 dBm signal applied to an amplifier with 10 dB gain will result in a 40 dBm output. Or, a standard test tone (0 dBm) will be measured as −15 dBm after passing through an attenuator of 15 dB.

It is important to note at this point that most meters used in the telephone industry are calibrated for measurements of voltage appearing across a 600-ohm termination (standard transmission line impedance). If the circuit to be measured is of a different impedance than that for which the meter is calibrated, the indicated power level will be wrong, and a correction factor must be taken into account. The relationship is:

$$dB\ (corrected) = dB\ (indicated) + 10 \log \frac{600\ ohms}{circuit\ impedance}$$

For example, a +6 dB reading across a 500-ohm line is calculated:

$$\begin{aligned} dB &= 6 + 10 \log \frac{600}{500} \\ &= 6 + 10 \log 1.2 \\ &= 6 + 0.792 \\ &= 6.792\ dB \end{aligned}$$

Level Point

In most telephone systems the toll switchboard is defined as the zero transmission level point (0 TLP), and the levels of both signal and noise at other parts of the system are usually referred to that point. A point in the transmission system where a signal has experienced 16 dB attenuation relative to the toll switchboard is known as the −16 dB level point. Note that *level* used this way is purely relative and has nothing to

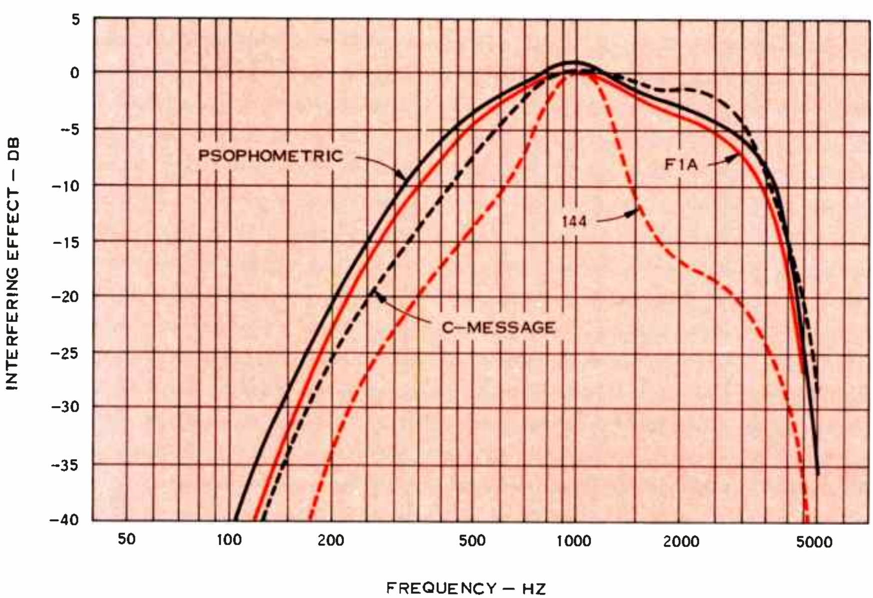

Figure 2. Weighting curves, based on listener response, show the relative interfering effect of noise on speech. All curves are referred to 1000 Hz except psophometric, which is based on measurements at 800 Hz.

do with actual power — a signal of any power will be down 16 dB at the −16 dB level point. When a standard test tone is transmitted over the circuit, its power in dBm at any point is numerically equal to the level in dB at that point.

dBm0

Another term, dBm0, is used to refer measured power back to the zero transmission level point, and has useful significance in system planning. Measurements adjusted to dBm0 indicate what the power would have been had it been measured at the zero transmission level point. For example, a tone measured at the −16 dB level point with a meter reading of +8 dBm, is equal to +24 dBm0.

In addition to dBm, there are a number of other logarithmic units used in the telephone industry which are expressed as dB above or below some reference power. One of the most common of these is dBrnc, used in the measurement of noise.

Noise Measurement

The Bell Telephone Laboratories and the Edison Electrical Institute did original research to determine the transmission impairment caused by noise interfering with speech. A large number of listening tests were made with different tones introduced as interference. The degree of interference was determined by comparing the power of each interfering tone with the power of a 1000-Hz tone that created the same degree of interference. A power of 10^{-12} watts, or −90 dBm, was selected as the reference power because it was found that a 1000-Hz tone at this power had a negligible interfering effect. Any noise power encountered that was greater than this could be given a positive value in *dB above reference noise,* or dBrn. These first measurements were made with the deskstand-type telephone popular in the 1920's, known as the Western Electric Type 144. From these measurements curves were plotted, called weighting curves.

dBa

Later, an improved handset (Western Electric Type F1A) came into general use, exhibiting a more uniform frequency response. Listener tests indicated that the new instrument gave approximately 5-dB improvement over the 144. Rather than change existing standards, a new reference noise power of −85 dBm (3.16×10^{-12} watts) was introduced. This also necessitated a change in the units, resulting in the adoption of dBa — decibels *adjusted.*

dBrnc

When the new 500-type handset was put into service in the 1950's, another line weighting was introduced, called C-message weighting. Since the new equipment improved on the old, an even higher reference power would have been required to express equal interfering effects with equal numbers. But this might have resulted in some unrealistic "negative" values of noise interference. So the reference power was returned to −90 dBm, and the units became dBrnc — decibels reference noise C-message weighted.

Weighting curves (Figure 2) for each handset compare interfering effects for various frequencies as referred to 1000-Hz interference. Noise measuring sets are frequency weighted in the same way so that meter readings obtained are meaningful in terms of what the ear detects. That is, the instrument does not measure noise intensity alone, but takes into account the frequency of the noise and how that particular frequency affects the ear.

Since there is no weighting effect on a 1000-Hz tone, straight forward conversion between dBa and dBrnc is possible by comparing reference power. A 1000-Hz signal having a power of 0 dBm yields 85 dBa and 90 dBrnc (Figure 3). But because weighting networks attenuate other frequencies differently, a uniform 3-kHz band of noise (flat or white noise) will not be measured the same as a 1000-Hz tone. White noise at 0 dBm will produce a noise reading of 82 dBa and 88 dBrnc. Approximate conversion is then accomplished by adding 6 dB to the dBa value:

$$dBrnc = dBa + 6.$$

For instance, if measuring with an instrument F1A weighted, a reading of 20 dBa would be equivalent to 26 dBrnc. The conversion factor is due to the 5 dB difference in noise reference power and an approximate 1 dB difference in weighting over the voice band.

Psophometric Weighting

In Europe and many other parts of the world, circuit noise is expressed in units established by the CCITT (International Telegraph and Telephone Consultative Committee). The unit, which is linear rather than logarithmic, is in terms of power measured in picowatts (10^{-12} watts), psophometrically weighted — written pWp. (Psophometric is from the Greek *psophos*, meaning noise.) The reference level, 1 pWp, is the equivalent of an 800-Hz tone with a power of −90 dBm, a 1000-Hz tone with a power of −91 dBm, or a 3-kHz band of white noise with a power of approximately −88 dBm. The shape of the psophometric curve is essentially identical to the F1A curve and similar to the C-message curve. Approximate conversions may be made as follows:

$$dBrnc = 10 \log pWp$$
$$dBa = -6 + 10 \log pWp.$$

Note that these terms all have absolute reference values of 10^{-12} watts, and are customarily written dBa0, dBrnc0, and pWp0 to relate the measurement to 0 TLP.

Noise Measuring Set Weighting	1000 Hz 0dBm	0-3 kHz 0dBm
F1A (dBa)	85	82
C-Message (dBrnc)	90	88

Figure 3. Relative readings received on F1A and C-message weighted noise measuring sets for single tone and white noise signals.

Signal/Noise

Occasionally the term signal-to-noise ratio (S/N) is encountered. The term, usually expressed in dB, indicates the number of dB the signal is above the noise. To obtain dBrnc from S/N, it is only necessary to calculate how many dB the signal is above reference noise power. For flat noise channels, the corrected reference (as mentioned previously for 3-kHz white noise) is −88 dBm. Conversions are, therefore:

$$dBrnc0 = 88 - S/N$$
$$S/N = 88 - dBrnc0$$
$$S/N = 88 - 10 \log pWp0$$

When it is necessary to measure speech or program volume in a transmission system, the simple dB meter or voltmeter is not adequate. The complexity of the program signal, as compared to pure sine waves, will cause the meter needle to move very erratically, trying to follow every fluctuation in power. This would obviously be difficult to read, and has no worthwhile meaning.

Volume Units

To provide a standardized system of indicating volume, a special instrument was created. Called a VU meter, it measures *volume units,* abbreviated vu. The VU meter is calibrated to read 0 vu across a 600-ohm line with a signal of 1 milliwatt (0 dBm) at 1000 Hz. The scale is logarithmic and reads vu above or below this zero reference. The instrument is not frequency weighted in any way, and while not designed for the purpose, it will read single frequencies directly in dBm. Its prime function, however, is to indicate the volume of complex signals in a way corresponding to the response of the ear. The reading is not an instantaneous value, but a value somewhere between the average and the peak value of the complex wave.

Other Units

Various other logarithmic units are used in the telephone and communications industries to conveniently compare like values. Crosstalk coupling in telephone circuits is indicated in dBx, or dB above reference coupling, and may be measured with a noise measuring set such as used to obtain dBrnc. Reference coupling is defined as the difference between 90 dB loss and the amount of actual coupling. Two circuits having a coupling of -40 dB could be said to have a coupling of 50 dbx.

Decibels may take on many other absolute values depending on their reference. Whereas dBm is a unit of power referred to one milliwatt, dBw is power referred to 1 watt. 0 dBw = 1 watt = 30 dBm. Similarly, dBk are decibels referred to 1 kilowatt.

Likewise, dBv is defined referencing 1 volt. However, in writing the equation for such a measurement, it is necessary to observe the following relationship:

$$dBv = 20 \log \frac{E_1}{E_2}$$

where

$$E_2 = 1 \text{ volt}$$

Note that the log of the voltage ratios is multiplied by 20, rather than 10 as in power ratios, expressing the square relationship between voltage and power ($P = E^2/R$). It is assumed that all measurements are across the same impedances.

Speech energy is commonly rated in terms of the intensity level of a speaker's voice measured one meter from his mouth. The standard reference acoustical power, 0 dBrap, is defined as 10^{-16} watts/cm^2.

Two other terms come into use in broadcasting: dBu, with 1 microvolt as the reference, and dBj, referred to 1000 microvolts. Both are measurements of signal intensity or receiver sensitivity. Any number of logarithmic units could be devised to suit special purposes, using decibels referred to some standard unit of power, voltage, or current.

Lenkurt Electric Co., Inc.
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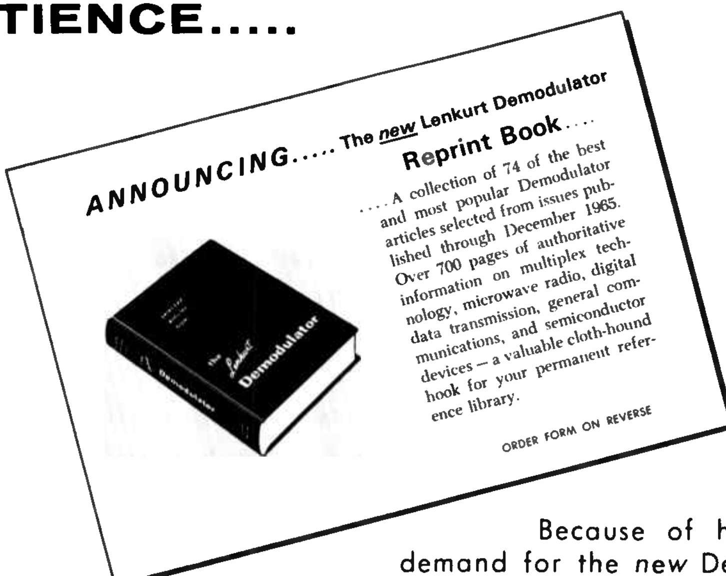

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MAY, 1966
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the Lenkurt Demodulator

Operating Standards

For Frequency Division Multiplex Systems

Operating standards in the communications industry help to provide an efficient means of linking together hundreds of separate and independent networks throughout the world. They are necessary to achieve uniform performance and to insure high quality transmission. This article mentions the prominent communications agencies that have contributed to standardization, and discusses some of the important characteristics needed to interconnect different frequency division multiplex systems.

One of the most valuable assets of any society is freedom of communication. The unrestricted transfer of information and ideas is vital to promote education, commerce, business, and government operations, and to protect the welfare and security of a free nation.

The vast telecommunications networks that have been developed in the United States and the rest of the free world have indeed become great national resources. These networks carry voice and telegraph messages and a variety of other forms of communications such as data, facsimile, and television, to almost any place in the world. The services provided by these networks must be reliable, economical, and increasingly useful in order to advance user satisfaction.

The enormous success of the communications industry certainly can be attributed to continual improvements made in the quality of service and to the increasing efficiency of equipment and facilities. This, of course, has resulted in lower costs and has permitted almost everyone to fulfill his essential communications needs.

Perhaps one of the most significant factors that has contributed to the progress of communication systems has been the development of universal *operating standards.* To achieve total worldwide communications, thousands of separate networks have to be linked together. It is extremely desirable, therefore, that each of these networks be able to handle the same types of electrical signals. If it is relatively easy to transfer signals from one system to another, the communications services are apt to be more economical and efficient. In a growing and dynamic world, it would certainly be impractical to develop communications networks that, because of technical differences, could not transfer messages to adjacent systems without complicated and expensive conversion equipment. This would be tantamount to railroad systems having tracks of different gauges!

It is also very important that each network preserve the quality of transmission. This means that the performance characteristics of these systems must conform to a *set of rules* which specify standards of operation. In answer to this, many written standards and practices have been developed to cover not only operating problems, but almost every aspect of electrical communications. These standards provide the basis of comparing and evaluating the performance of communications systems. Although the use of such standards often is not obligatory, they are essential and are generally recognized and accepted by the communications industry. The particular standards adopted depend, of course, on the type of system, its intended use, and the performance requirements necessary to interconnect it with other systems.

Who Issues Standards?

In the United States, the most widely used standards or performance objectives are those of the Bell System and the Department of Defense. The Bell System has developed most of the standard practices that are used by the telephone industry to interconnect long-haul multiplex and carrier systems in North America. These standards are contained in publications known as *Bell System Practices (BSP's).*

For the huge worldwide Defense Communications System (DCS), a separate set of standards has been established. Operation of the DCS is controlled by the Defense Communications Agency (DCA) which issues DCS Engineering-Installation Standards to assure uniform high-quality performance of each segment of the system. Where appropriate, these military standards agree with those developed for use by the telephone industry.

There are other agencies and organizations that play a very active role in developing operating standards for carrier and multiplex systems. Prominent among these are the Communication and Signal Section of the American Association of Railroads (AAR) and the Rural Electrification Administration (REA) of the Department of Agriculture. Also, the Electronics Industries Association (EIA) has been very active in helping to standardize the characteristics of digital data signals that are to be transmitted over communications systems.

Another important set of standards used in the development of carrier telephone systems is produced by an organization known as the International Telegraph and Telephone Consultative Committee (CCITT). This body is a branch of the International Telecommunications Union (ITU), located in Geneva, Switzerland. The ITU is an agency of the United Nations.

The CCITT issues recommendations for standardizing international telephone and telegraph circuits. The need for such recommendations developed originally in Europe where many dif-

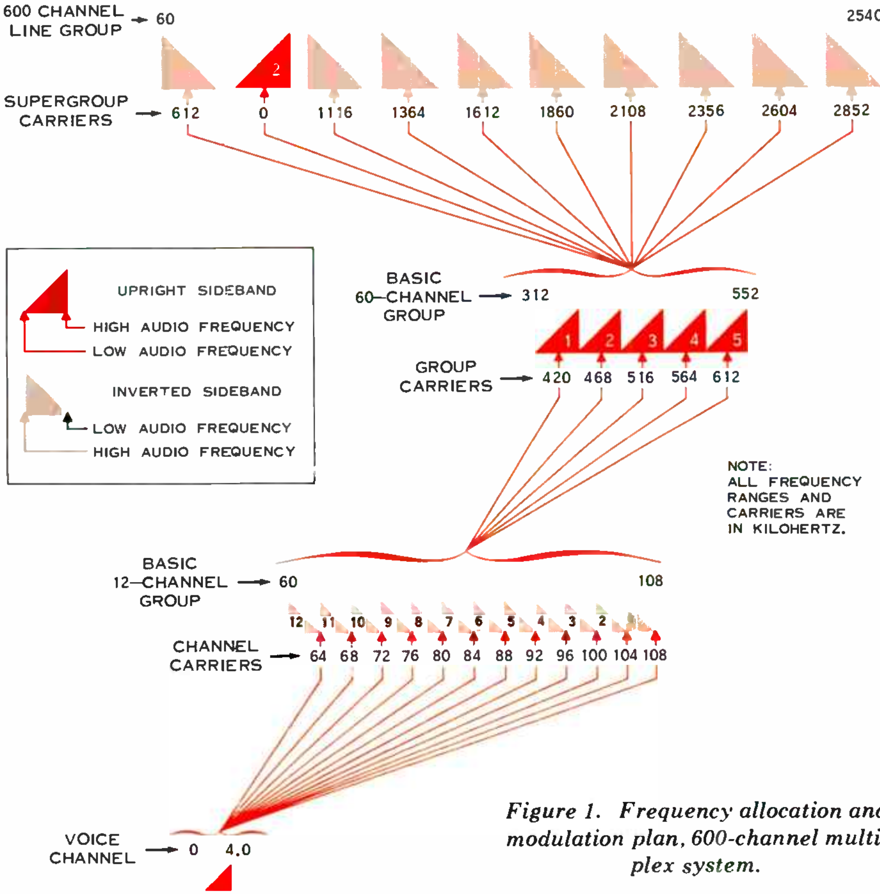

Figure 1. Frequency allocation and modulation plan, 600-channel multiplex system.

ferent telephone administrations had to interconnect at international borders. Unlike other parts of the world, Europe has many dense population centers concentrated in small political divisions. Because of the relatively short distances separating these populated areas, there is a great amount of telephone traffic between them. Therefore, it was necessary to establish an international cooperative organization where the nations involved could get together and agree on universal standards. Such agreements have been very effective in assuring that international circuits of various national telephone administrations and common carriers are compatible. Today, countries all over the world who are interested in promoting and developing international telecommunications networks, are represented in the CCITT.

Frequency Allocation and Modulation Plans

One of the most important aspects of interconnecting frequency-division multiplex and carrier systems is the assignment of frequencies. Each type of carrier and multiplex system employs some type of modulation scheme to shift the voice-frequency signals received from user equipment to some suitable line or baseband frequency range. These schemes are referred to as *frequency allocation and modulation plans.*

Whenever two carrier systems are connected in tandem, signals at the interface point must conform to the technical requirements of the receiving system. Of course, signals at line or baseband frequencies can be simply demodulated to the voice-frequency range and then transferred to the next system. This method, although acceptable, has proven to be rather inefficient in many cases. Extra equipment is needed to demodulate the signals and each additional modulation and demodulation step adds distortion to the signal. What was needed was a standard modulation plan which would allow different carrier and multiplex systems to be interconnected directly at line or baseband frequencies or at some intermediate stage of modulation. This would allow groups of channels to be transferred between systems without the need for extra equipment and unnecessary modulation steps.

When the Bell System began developing its wideband coaxial cable carrier system in the 1930's, considerable thought was given to standardizing single-sideband suppressed-carrier multiplex terminal equipment. One of the results of this effort was the establishment of a standard modulation plan for groups of channels. To accomplish this it was first necessary to standardize the spacing of channel carriers; the Bell System decided on a uniform spacing of 4 kHz. This would permit all channel carriers to be harmonically related to 4 kHz and would allow room to improve the quality of speech transmission with advances in filter design. The next step was to formulate a basic modulation plan that could be used in open-wire and multipair cable systems as well as wideband coaxial cable systems. With 4-kHz channel spacing, single-sideband suppressed-carrier open-wire systems operating with line frequencies above 30 kHz (this would place such systems above the Bell System's 3-channel type C carrier system) could only handle about 12 channels. So it was decided to establish a basic modulation plan for 12 channels. Coaxial cable systems (and later microwave radio systems) were, of course, not limited to 12 channels, but standard 12-channel groups could be used as building blocks to form systems with hundreds of channels by simply using additional stages of modulation. Since the practical lower frequency limit for coaxial cable was about 60 kHz, the standard 12-channel group was established with a frequency range of 60 to 108 kHz.

This standard 60 to 108 kHz 12-channel group has received wide acceptance as the basic building block for long-haul carrier and multiplex systems, and has been adopted by CCITT for use in international circuits and by the DCA for use in the Defense Communications System. Additionally, a standard 60-channel *supergroup,* formed from five 60 to 108 kHz channel groups, has been adopted for use in wideband systems. This supergroup has a frequency range of 312 to 552 kHz.

An example of a frequency allocation and modulation plan for a 600-channel multiplex system is shown in Figure 1. In the first modulation stage for this plan, each voice-frequency input signal modulates one of 12 *channel carriers* spaced 4 kHz apart. The lower sideband signals are selected to provide the standard 60 to 108 kHz 12-channel group. In the second modulation stage, five 12-channel groups each modulate

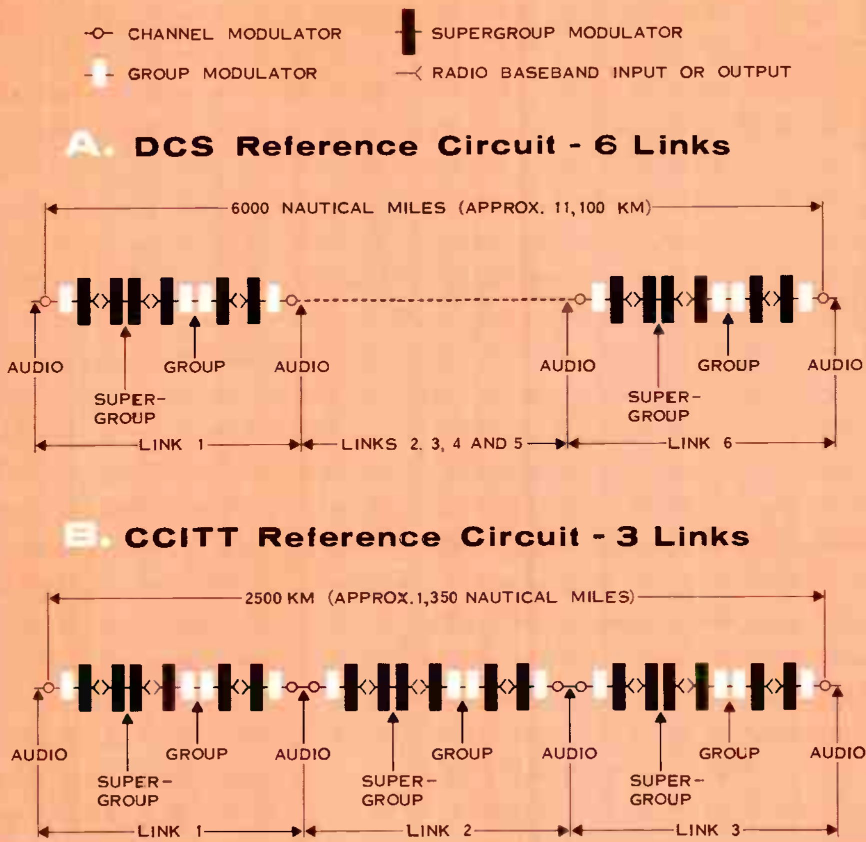

Figure 2. DCS and CCITT reference circuits.

a separate *group carrier* to produce a standard 60-channel supergroup with a frequency range of 312 to 552 kHz. Ten of these supergroups are needed to form the 600-channel system.

In the final stage of modulation, nine of the ten supergroups each modulate a separate *supergroup carrier*, resulting in line frequencies ranging from 60 to 2540 kHz. One of the supergroups (supergroup number 2) is applied directly to the line at the 312 to 552 kHz frequency level.

This particular 600-channel modulation plan is recommended by the CCITT and is standard for use in the Defense Communications System. The Bell System uses a slightly different line frequency range for their type L wideband carrier and multiplex systems. In the type L system, the modulation plan is the same as the one described, through supergroup 8. Supergroups 9 and 10, however, employ carriers of 1860 and 3100 kHz, respectively, resulting in an upper line frequency of 2788 kHz rather than 2540 kHz.

Modulation plans can be expanded to meet the future needs of higher density wideband multiplex systems requiring 2700 or more channels. These expanded plans are formed by additional modulation steps using higher order *master* and *supermaster* channel groups. By adhering to these standard frequency allocation and modulation plans, it is possible to directly interconnect 12-channel, 60-channel, and higher order channel groups of various carrier and multiplex systems, without having to first demodulate the signals down to the voice-frequency range.

For single-sideband suppressed-carrier open-wire carrier systems, the standard frequency allocation and modulation plan provides up to 12 channels. Since open-wire systems are typically 2-wire systems, the frequencies transmitted to the line must be different for each direction of transmission. This establishes what is known as an *equivalent 4-wire* system. The two directions are conveniently referred to as the *east-west* direction and the *west-east* direction.

After the 12 voice-frequency channels have been translated to the 60 to 108 kHz group level, they are then shifted to one of four staggered line frequency allocations. Staggered line frequency allocations are necessary to overcome unacceptable crosstalk where different systems share the same open-wire lead. The four staggered line groups are shown in Table 1.

This open-wire modulation plan is used in the Bell System type J carrier system and is specified standard by the DCA and CCITT.

The standard 60 to 108 kHz group modulation plan is also used in multipair cable carrier systems to provide 12 or 24 channels. The line frequencies for cable systems are also different for each direction of transmission. The DCA prescribes a 12-channel system with line frequencies of 6 to 54 kHz for one direction and 60 to 108 kHz for the other direction.

The standard 24-channel plan used by the telephone industry requires two basic 60 to 108 kHz 12-channel groups. Again the line frequencies are different for each direction of transmission. Typically, the channels in one direction

TABLE 1.

Staggered line frequency allocations for 12-channel open-wire carrier system.

SYSTEM		WEST-EAST	EAST-WES
DCA	CCITT		
A	SOJ-A-12	36 to 84 kc	92 to 140
B	SOJ-B-12	36 to 84 kc	95 to 143
C	SOJ-C-12	36 to 84 kc	93 to 141
D	SOJ-D-12	36 to 84 kc	94 to 142

are referred to as the *low line group*, and have a frequency range of 36 kHz to 132 kHz. The channels in the other direction, called the *high line group*, have a frequency range of 172 to 268 kHz.

Performance Objectives

In order to define standard performance objectives of communications systems, the CCITT, the United States telephone industry, and the DCA have established hypothetical *reference circuits*. These circuits are of a specified length and are composed of a certain number of links. The amount of equipment in each link varies depending on whether the transmission path consists of open wire, cable, or radio. The reference circuits are complete transmission systems interconnecting two audio-frequency terminals. Each link consists of a number of 4-wire, nominally 4-kHz voice-frequency circuits derived from single-sideband suppressed-carrier frequency-division multiplex equipment using standard modulation plans. Such hypothetical reference circuits are very useful in establishing guidelines for the performance characteristics of a communications system.

The CCITT reference circuit, Figure 2A, is 2500 kilometers (1550 statute miles) long and consists of three tandem links with interconnections made at group and supergroup frequency levels. In the United States, the telephone industry uses a reference circuit of 4000 miles, made up of a maximum of seven links. However, from a performance standpoint the two circuits provide essentially the same results.

The DCS reference circuit for wideband systems is shown in Figure 2B. This hypothetical circuit is 6000 nautical miles long and consists of six tandem links each approximately 1000 miles long and interconnected on a 4-wire basis at the audio-frequency level. Each link is divided into three sections of equal length and consists of wire or radio facilities plus necessary repeaters and frequency division multiplex equipment.

Through the use of these hypothetical reference circuits, it is possible for various manufacturers to develop multiplex equipment with uniform performance capabilities. In addition to the type of multiplexing and associated frequency allocations and modulation plans, the reference circuits are used to define and standardize other circuit characteristics such as noise objectives, power levels, impedances, pilots, and signaling in order to interconnect groups of channels at carrier frequencies.

Power level is a very important factor which must be considered when establishing guidelines for interconnecting multiplex systems. The amount of power required at the voice-frequency input and output circuits of multiplex systems is determined by the needs of the subscriber or user equipment, or the switching center in the communications system. In the United States, the standard used to set the levels for speech transmission is a 1000-Hz test tone at a level of 0 dBm0. The CCITT specifies an 800-Hz tone for the same purpose.

Both the Bell System and the DCA have standardized the input level of speech signals at -16 dBm and the output level at $+7$ dBm with a balanced circuit impedance of 600 ohms. These levels result in a net gain of 23 dB from the input of the multiplex transmit channel to the output of the distant multiplex receive channel. The CCITT also recommends a voice-frequency circuit impedance of 600 ohms, but has not specified any standard voice-frequency power levels.

Conclusion

The development of universal operating standards for carrier and multiplex systems has certainly been a tremendous help in advancing worldwide international communications. Such standardization has made it possible to transfer groups of channels at carrier frequencies directly from one communications system to another. This has resulted in communication services with greater efficiency, better quality, and lower costs.

Direct Distance Dialing in the United States is an excellent example of what can be achieved through the use of operating standards for communications transmission equipment. With the advent of worldwide multiple-access communications satellites the need for universal standards for interconnecting carrier and multiplex systems will certainly become more and more significant.

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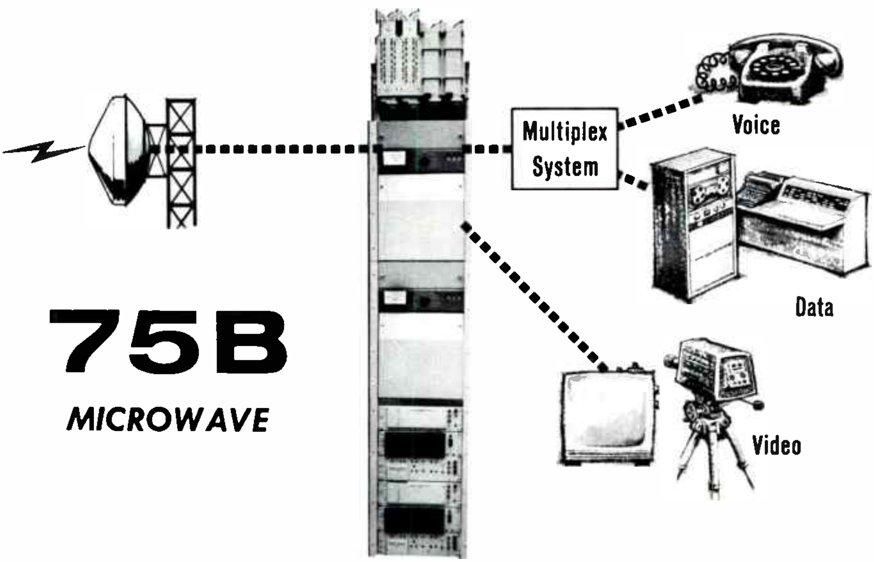

75B
MICROWAVE

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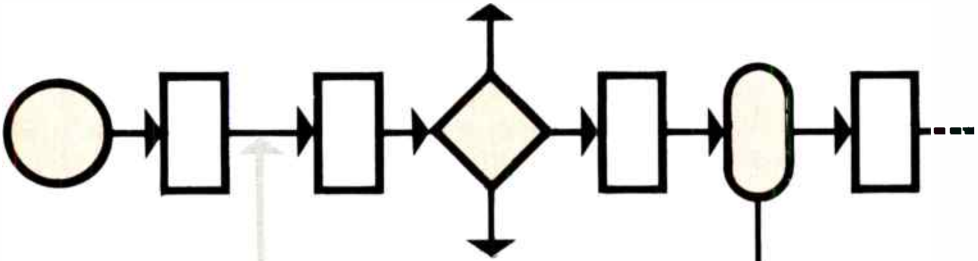
the Lenkurt Demodulator



the COMPUTER *and* *the* ENGINEER

The computer has become a most valuable and necessary piece of equipment in the design laboratory of today's electronics firm. Through accuracy and speed in complicated mathematical calculations, the computer has made possible design techniques never before feasible.

This article concerns the scientific digital computer and its uses by the electronics engineer.

In a relatively short life, computers have grown out of desk-top calculating machines, relying on a complicated assortment of gears and mechanical linkage to perform their task, to electronic giants capable of thousands of mathematical calculations a second. Computers are called on for mammoth data processing duties, for controlling automated machines and, of prime interest to the electronics design engineer, to carry out complicated mathematical computations with a speed and accuracy never before possible.

Computer Types

Computers are of two basic types, analog and digital. There is very little similarity between the two, and they serve markedly different functions. The analog computer is used primarily as an analysis device, where the physical properties of a system—mechanical, electrical, or whatever—can be represented by analogous circuits in the computer. With amplifiers, potentiometers, and circuits performing such mathematical functions as differentiation and integration, the computer imitates the real item. Once an electrical model is devised, analyzing the system is particularly suited to the analog computer.

In operation, the system is tested under varying conditions by changing voltages in the computer, with all calculations carried out simultaneously. Results are displayed immediately on one of a number of readout devices: strip-chart recorders, X-Y plotters, or even oscilloscopes. The complexity of the model has little to do with the computer's problem-solving speed.

Digital computation, a more time consuming step-by-step process, is much more accurate, and is limited basically only by the preciseness of input data and the ability of the machine to store and use significant figures. Digital computers, with which this article is primarily concerned, are basically manipulators of stored information.

The computer (Figure 1) consists of a central processing unit, where the actual computation is done, a storage unit providing the machine with an electronic memory, and input and output devices with which the user can communicate with the machine.

The facts of a problem are held by the machine's memory — commonly a large grouping of magnetic cores which can retain thousands of single electrical pulses for later use. Instructions to the computer, called a *program,* are written by the operator to guide the machine in processing and changing the stored pieces of information in a prescribed way. In data processing operations, huge quantities of information are handled, facts sorted, relatively easy computa-

Figure 1. Typical scientific computer is the IBM 1620 II, used at Lenkurt. Behind the central processing unit is magnetic disc storage drive.

tions done, and the information reassembled in meaningful form — a printed page, a stack of punched cards, payroll checks, or perhaps utility statements.

Scientific Use

Scientific computers serve an entirely different function. The computer first sees the scientific or engineering problem as a small set of numerical facts; the answer will probably be no more than a short series of digits. But inside the computer an enormous amount of work has gone on. Mathematical computations which might have taken weeks — if possible at all — had they been done manually by the engineer, are now obtained from the computer in minutes.

The digital computer does all its work by breaking down complex mathematical functions into many simplified arithmetic operations, much as the mechanical desk calculator does multiplication by a series of additions. With the computer, difficult mathematical problems can be reduced to processes of addition, subtraction, and logical decisions, performed thousands of times per second.

The theoretical knowledge necessary to construct a computer had been available for many years, but it was not until reliable electronic components freed the computer from the limitations of the mechanical calculating machine that the "brain" began to grow.

The first computers were merely arithmetical manipulators able to do a sequence of calculations without further human control. They were complex, could handle huge quantities of data, and did it with speed never before imagined possible. But they were tied to limited specific processes, much like a platoon of girls operating desk calculators with instructions to carry out a single set of arithmetic operations and list them on a form.

Logic Added

An historic hurdle was overcome when logic functions were added — the computer now had the ability to make decisions. This is synonymous to having the desk calculator operators stop at some point in their work, analyze the figures, and proceed on one of a number of alternate paths. In its simplest form, logical response is to answer "yes" or "no" to a certain stimulus; that is, to take one of two possible paths. Judging the presence or absence of some variable condition, and making a decision based on this, is the function of the elementary logic circuit of the computer. In this way the modern computer makes decisions which will determine some later course of action.

Beyond the ability to make decisions comes the faculty of *learning* from experience. Some computers can analyze their own method of approaching a problem and improve on it the next time the same problem is presented.

The availability of electronic computers in the early 1950's for use by design engineers greatly relieved the tedious routines followed until that time. One of the common — and laborious — problems faced by telecommunications equipment designers is the creation of new filter networks. (About 90 percent of *all* networks are filters, defined as circuits to separate wanted signals from groups of signals.) Well-developed methods are available for making accurate mathematical models of network circuits, involving extensive use of complicated calculations with

32	16	8	4	2	1	NUMBER
0	0	0	0	0	1	1
0	0	0	0	1	0	2
0	0	0	0	1	1	3
0	0	0	1	0	1	5
0	0	1	0	1	0	10
1	0	1	0	1	1	43

Figure 2. Sample of binary number equivalents used in computers. Values increase in multiples of two, right to left.

Figure 3. Electric typewriter allows engineer to supply data to computer as design problem progresses. Results of computation also appear on the typewriter.

many repetitive functions. This suits the digital computer perfectly.

Binary System

The digital computer, as opposed to the analog machine, deals with discrete numbers. Just as the mechanical calculator relies on the exact positioning of gears for computation, the computer positions binary pulses in a precise method, storing them in its electronic memory until they are to be used.

The binary number system in computer machine language differs from conventional decimal numbers only in concept. In customary numerical expression using the decimal (or more correctly the "denary") system, a number such as 837 is merely a convenient method of listing the number of units, tens, hundreds, etc. In each position there is a possibility of 10 different integers, 0 through 9. The number 43 really means $(4 \times 10^1) + (3 \times 10^0)$.

The binary system is based on the possibility of only two choices in each position, written as either 1 or 0. Examples are given in Figure 2. The number 43 is written 101011, or $(1 \times 2^5) + (0 \times 2^4) + (1 \times 2^3) + (0 \times 2^2) + (1 \times 2^1) + (1 \times 2^0)$. While this may seem a cumbersome notation, it is very appropriate for digital computers because it corresponds with the natural two-state ability of many electronic components—relays, magnetic core memory units, and flip-flop circuits.

Computer Languages

To communicate with the computer, the user must either *talk* in the basic machine language or, if he prefers, provide a means of translation in some other language. In modern computers, several programming languages are available to the user, depending on the specific purpose at hand. Special "compiler" programs already in the computer perform the translation. In this way the programming languages are *user ori-*

ented, rather than machine oriented, and allow the programmer freedom of expression and thought in terms more familiar to him.

A language developed primarily for scientific work is FORTRAN (FORmula TRANslation) which bears a close resemblance to the native tongue of the engineer and scientist — mathematics. A similar language more popular in Europe is ALGOL (ALGebraically Oriented Language). Add to these COBOL (COmmon Business Oriented Language), PL/1 (Programming Language/1), and a growing number of languages for specific disciplines.

The facts of the scientific problem, along with the program to be used to reach a solution, are fed to the computer from one of a number of input devices — magnetic tape, magnetic discs, punched cards, perforated tape, and the typewriter. The computer stores this information in its electronic memory for future use. Certain facts may be supplied directly by the engineer, operating an electric typewriter at the computer console (Figure 3). Answers also will appear on the typewriter when the problem is completed.

When a program is first written for a problem, a flow diagram is constructed expressing the logic of the approach. A non-mathematical example in Figure 4 illustrates decision making, computer style.

Iteration

The example also points to the repetitive ability of the digital computer, one of the best used qualities of the computer in scientific calculation. By the mathematical process of *iteration* (or

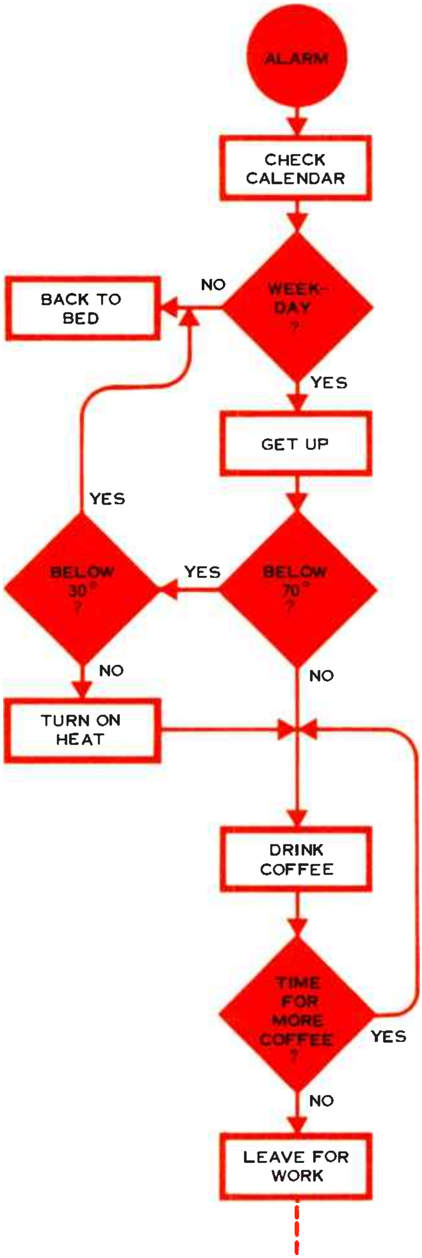


Figure 4. Simulated flow diagram illustrates logic used in planning computer program.

repetition) the computer can begin with a guess at the answer and then check it. If necessary, the computer will modify and repeat the original calculation. With each iteration, the machine answer approaches the correct value sought.

One very common example of an iteration, known as the Newton method, is illustrated graphically in Figure 5. The problem is to find the special value of a, the point where the curve crosses the x axis. The point also corresponds to $y = 0$. The computer makes a first guess x_1 for the value of a and then computes y_1. Since y_1 is not zero, the computer seeks a better approximation of a. It computes the slope (or tangent) of the curve at (x_1, y_1), then calculates where this straight line intersects the x axis. This value, x_2, is taken as a better approximation of a. The sequence is repeated until the value of a is known as accurately as the capabilities of the machine will allow. The iteration is then stopped.

Another example of iteration is the solution of the quadratic equation

$$x^2 + ax - b = 0$$

This can be rearranged to give

$$x(x + a) - b = 0$$

$$x = \frac{b}{x + a}, \text{ and finally}$$

$$x_{n+1} = \frac{b}{x_n + a}.$$

With $n = 0$, a guess for the value of x is entered and the calculation is carried out for the right side of the equation. If the answer is equal to x, our guess has been accurate. If not, the first answer is substituted for x_1, and the calculation

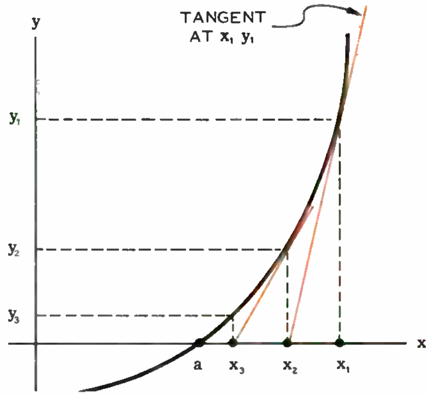


Figure 5. Computer uses iteration to find the point where curve crosses x axis.

is carried out again — approaching the true value of x at each iteration.

Given the equation

$$x^2 + 6.3x - 14.8 = 0,$$

the iteration approach would proceed in this way:

$$\text{Let } x_0 = 0$$

$$x_1 = \frac{14.8}{0 + 6.3} = 2.349$$

$$x_2 = \frac{14.8}{2.349 + 6.3} = 1.711$$

$$x_3 = \frac{14.8}{1.711 + 6.3} = 1.847$$

$$x_4 = \frac{14.8}{1.847 + 6.3} = 1.817$$

$$x_5 = \frac{14.8}{1.817 + 6.3} = 1.823$$

$$x_6 = \frac{14.8}{1.823 + 6.3} = 1.822$$

$$x_7 = \frac{14.8}{1.822 + 6.3} = 1.822$$

The iteration, in this case, was stopped when four significant figures in the answer stabilized. This process, by its nature, will converge on the smaller of the two roots of the equation. The second root is found immediately by dividing the final term in the equation by the first root:

$$\frac{-14.8}{1.822} = -8.122$$

Monte Carlo Method

Another important mathematical practice appropriate to the iterative ability of the computer deals with problems of probability, and is called the Monte Carlo method. In the manufacture of electronics equipment, component values vary randomly within the specified tolerance range assigned. With the help of the computer it is possible to calculate what percentage of finished products will be within a nominal performance range. Using the Monte Carlo method, the computer generates values for the circuit components in a random manner, much the way a factory worker would select actual parts during assembly. Each time different component values are selected, the circuit performance is checked by the computer. From this, a distribution curve can be plotted, indicating the probable number of acceptable units. If the percentage is too low, tolerances must be tightened for components most likely to affect the circuit, and the process is repeated. In this way the design engineer can use his mathematical model of the circuit to advantage, predicting with great accuracy the performance of the finished units leaving the assembly line.

Filter Design

A typical filter network design problem will illustrate the procedure used by the engineer. A standard network and its loss-frequency curve are shown in Figure 6. The designer will work with a number of computer programs, each intended to help him solve particular problems and approach a practical end product. The first program used will aid him in selecting the optimum parameters defining the performance of the filter. After feeding the program into the computer's memory, the engineer enters tentative values for certain design characteristics, such as passband edge frequencies (f_1, f_2), the frequencies at the infinite loss points ($f_{1\infty}$, $f_{2\infty}$, $f_{3\infty}$, $f_{4\infty}$) and the passband ripple. Then, one by one, the designer supplies the computer with sample frequencies in the stopbands and passbands. The computer calculates the loss at each of these points from which an accurate loss curve can be plotted.

When satisfied with his basic design, the engineer must determine the components to be used. A second program is placed in the computer's memory. Using what is called *insertion-loss theory*, the engineer and the computer can now generate the component values needed to satisfy the original design characteristics.

The results of this second operation are used in a general analysis program to prove the design under practical conditions (till now the components were considered to have pure capacitance and inductance). The computer then produces values for total loss, phase shift, envelope delay, reflection coefficient and input impedance.

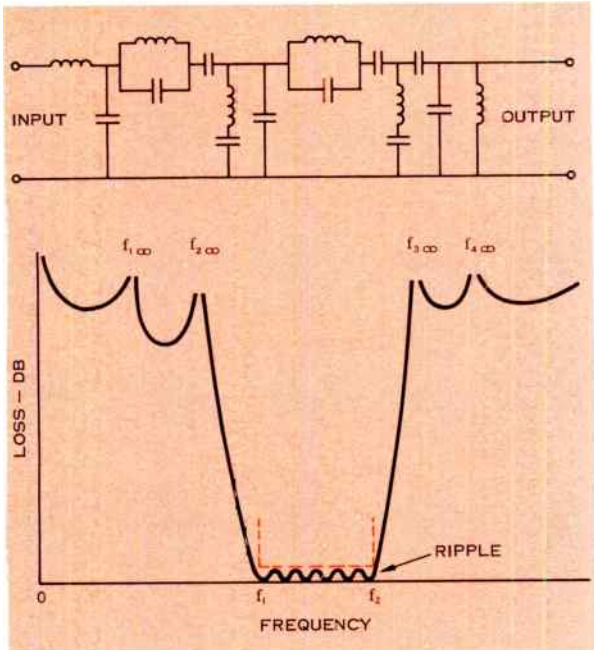


Figure 6. Filter circuit (top) will produce this type of loss-frequency curve. Engineer and computer determine optimum component values to provide proper passband frequencies, acceptable ripple, and infinite loss points.

A fourth computer step is used to examine tolerances, using the Monte Carlo method previously described. When all the design programs have been completed, a prototype model can be built, and the filter circuit is well on its way to becoming part of a new product.

Prior to the time when engineers had the work-horse capabilities of the computer at their disposal, a method known as *image-parameter* theory was used in filter design. Insertion-loss theory was just not practicable due to the vast amount of calculation necessary. But with the computer a slightly superior product can be produced using the insertion-loss technique, with the added advantage of the method's simplicity and straightforward approach.

Time Saving

Approximate times for performing the various steps may be listed, considering computers now found in most engineering facilities, like the IBM 1620II used at Lenkurt, but understanding that more modern and likewise faster machines are being produced constantly. The first program, used to select the basic performance parameters for a fairly complicated filter, takes the engineer and the computer together about 30 minutes. Component values are produced from the second program in about 10 minutes. The general analysis program takes an additional 10 to 15 minutes. Longest because of its many repetitive steps is the Monte Carlo tolerance analysis, running about 60 minutes. Total time: less than two hours.

The image-parameter method is not only slower by two to ten times depending on the complexity of the filter, but lacks the complete information obtained by the insertion-loss technique. In image-parameter design it is still necessary to construct a "breadboard" model to check dissipation characteristics and tolerances — all done on the computer with insertion-loss methods.

Other Uses

In addition to the design of networks, a great deal of time on the computer is spent analyzing one-time mathematical problems that do not lend themselves to generalized programming. Again, time-saving benefits of the computer are invaluable to the design engineer. As long as mathematics is a part of electronics design, computers will be a tool for the design engineer. And even today's methods, superior by many orders of magnitude to the labors of only a few years ago, will be supplanted as more sophisticated techniques are created, and man and machine learn better how to work together.

Probably one of the most promising developments in the last year is graphical on-line simulation of electronic circuits. Using a "light pen" the designer can draw his circuit on the face of a computer-coupled cathode ray tube. With proper commands to the machine's memory, he can analyze circuit functions under specific parameters. Using the light pen, almost instantaneous modifications are possible, allowing the engineer to redesign the circuit and see the results while still at the computer.

The same device is used in other disciplines to analyze the structure of bridges, buildings, and even three-dimensional mathematical representations — viewed from side, top, or bottom.

Time Sharing

There is a trend to centralize computer facilities for use by a number of subscribers on a time-sharing basis. Several large computer centers have already been established around the country, with users connected remotely through telephone communications facilities. Time-sharing computers can allow two-dozen or more users on the line at one time, efficiently sandwiching their programs together with such speed that the user seldom, if ever, notices a delay. The user gains the advantages of large computer capability, while only paying for the computing time he needs.

The scientific computer, modern-day design tool of the electronics engineer, will improve in its usefulness as quickly as man's inventiveness will allow. Along the way, the computer is not only absorbing a tremendous work burden, but is making possible increased quality and better performance in electronics equipment.

HERTZ?

In an effort to reach worldwide understanding in published scientific and technical work, standardization of terms is essential. The Institute of Electrical and Electronics Engineers (IEEE) standards committee recently accepted a number of new standards of electrical units and symbols, established in close cooperation with many international organizations. At the beginning of the year the *Demodulator* adopted the IEEE recommendations.

One of the most noticeable changes is the adoption of the name *hertz* as a unit of frequency. While *cycles per second* is still widely used, and technically correct, *hertz* is now preferred in that it is more understandable in all languages.

In general, symbols for units are written in lowercase letters, except when the name of the unit is derived from a proper name. Thus, it is *hertz,* but *Hz* when abbreviated; *decibel,* but *dB.*

Compound prefixes are to be avoided: $\mu\mu$F (micromicrofarad) becomes pF (picofarad); mμs is now ns (nanosecond); and the familiar kmc (kilomegacycle) should be GHz (gigahertz).

Prefixes for Metric Units

10^{12}	tera	T
10^{9}	giga	G
10^{6}	mega	M
10^{3}	kilo	k
10^{2}	hecto	h
10	deka	da
10^{-1}	deci	d
10^{-2}	centi	c
10^{-3}	milli	m
10^{-6}	micro	μ
10^{-9}	nano	n
10^{-12}	pico	p
10^{-15}	femto	f
10^{-18}	atto	a

Examples of Usage

decibel	dB
gigahertz	GHz
hertz (cycles/second)	Hz
kilohertz	kHz
megahertz	MHz
picofarad	pF
siemens (mho)	S ($1\ S = 1/\Omega$)

Readers may consult the IEEE Standard Symbols for Units (No. 260, January 1965) and other IEEE documents for more complete details.

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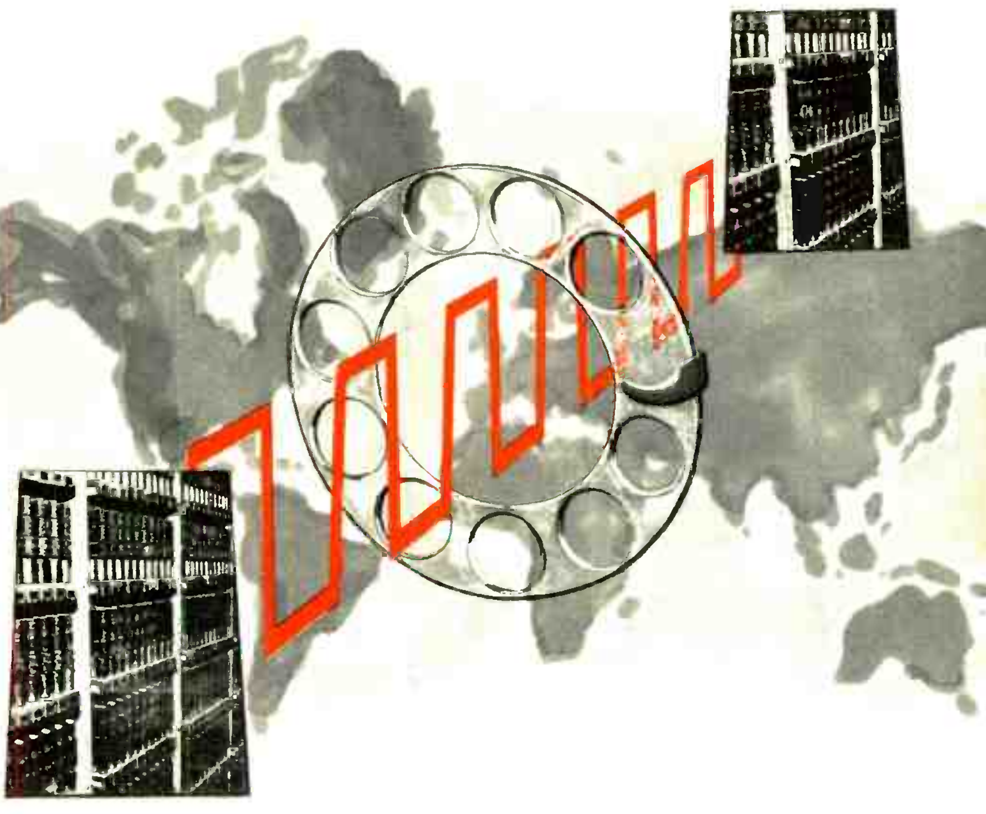
the Lenkurt Demodulator

SIGNALING
over telephone trunks



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Signaling provides the means for operating and supervising a telephone communications system; it establishes connections, announces incoming calls, reports the fact that a line is busy. The functions of signaling are indeed most vital to the basic operation of the telephone plant.

Trunk signaling involves many considerations that are quite different from the basic techniques employed in signaling over a subscriber loop. This article reviews some of these considerations and describes the major techniques used to transmit signaling information over physical and carrier-derived trunk circuits.

Without signaling, a telephone system cannot operate. Even the simplest system, such as two local battery telephones connected by field wire, requires some means for the users to attract one another's attention when they want to talk. In early telephone systems, users simply cranked a hand magneto which caused a bell to ring at the subscriber station or a flag to drop at a switchboard. Over the years, signaling systems have had to keep pace with the advances made in telephone switching and transmission systems. The increasing complexity of the worldwide telephone plant has had a tremendous influence on the evolution of signaling techniques, from the simple hand cranked magneto to the many techniques employed today.

Many different signaling methods have evolved during the transition from one type of switching office or transmission system to another. Today's telephone plant includes various types of local exchange and toll switching offices, such as manual, step-by-step, panel, crossbar, and the modern electronic switching offices. In addition, there are many types of open-wire, cable and microwave radio transmission systems interconnecting the various switching offices.

Signaling Functions

There are a multitude of signaling functions that must be transmitted between the various manual and dial switching offices. These include functions whereby people must communicate with machines, machines must communicate with other machines, and machines must communicate with people.

The major functions can be somewhat arbitrarily classified as *ringing, supervisory,* and *address* (or dialing). Ringing signals are used to operate a visible or audible alarm to alert someone of an incoming call. Supervisory signals are used to convey information regarding switchhook conditions (on-hook or off-hook) at either end of a telephone circuit. Address signals convey dialing or digital information which is necessary to establish the desired connection.

In subscriber loops, supervisory and address signals are accomplished by means of direct current, while alternating current is used for ringing. Direct current signaling is also used on short-haul trunks between switching offices. However, such methods are not adequate for signaling on longer trunks, such as inter-toll, or on trunks derived from carrier or multiplex systems. As a result, various alternating current signaling systems have been developed for use over long-haul v-f and carrier-derived trunks.

Ringdown Trunks

In certain trunks, especially those interconnecting manual offices, it is necessary to transmit a ringing current to signal the switchboard operators. This type of signaling is known as *ringdown.* The ringing alternating current used in subscriber loops is at a frequency of 20 Hz. This same frequency is also used in certain short-haul trunks. On trunks equipped with composite telegraph, 20-Hz ringing cannot be used because of interference. In these circuits, a signaling frequency of 135 Hz is used. Neither of these frequencies, however, is suitable for long trunks because voice-frequency repeaters cannot pass them. Consequently, a 1000-Hz signaling tone, well within the v-f amplifier passband, has been adopted for use on longer circuits. To prevent voice signals from falsely operating the signaling equipment, the 1000-Hz tone is interrupted (modulated) at a 20-Hz rate.

Address Signals

Probably the most important and the most complicated signaling function is address or dialing. This function directs the operation of the switching equipment in the automatic offices. Consequently, the evolution of the various switching systems has brought about changes in address signaling techniques. Address signals originate at the telephone dial and consist of a train of dc pulses corresponding to the number dialed. Modern "touch calling" systems, which use keys or pushbuttons instead of a dial, employ tones at different frequencies rather than dc pulses.

In the step-by-step systems, the switching equipment responds directly to the dc pulses. However, in panel and crossbar systems, the switches cannot be controlled directly by the dial pulses. Consequently, these systems require a device known as a *sender* which stores the dial pulses and then controls the movement of the switches.

There are four basic methods commonly used to transmit address or dialing signals for use by the various switching offices. These are known as *dial pulsing, revertive pulsing, panel call indicator (PCI) pulsing,* and *multifrequency (MF) pulsing.*

Dial pulsing is the earliest and most commonly used method of transmitting address information — the numerical value of each digit is represented by the number of pulses in a train (ten pulses

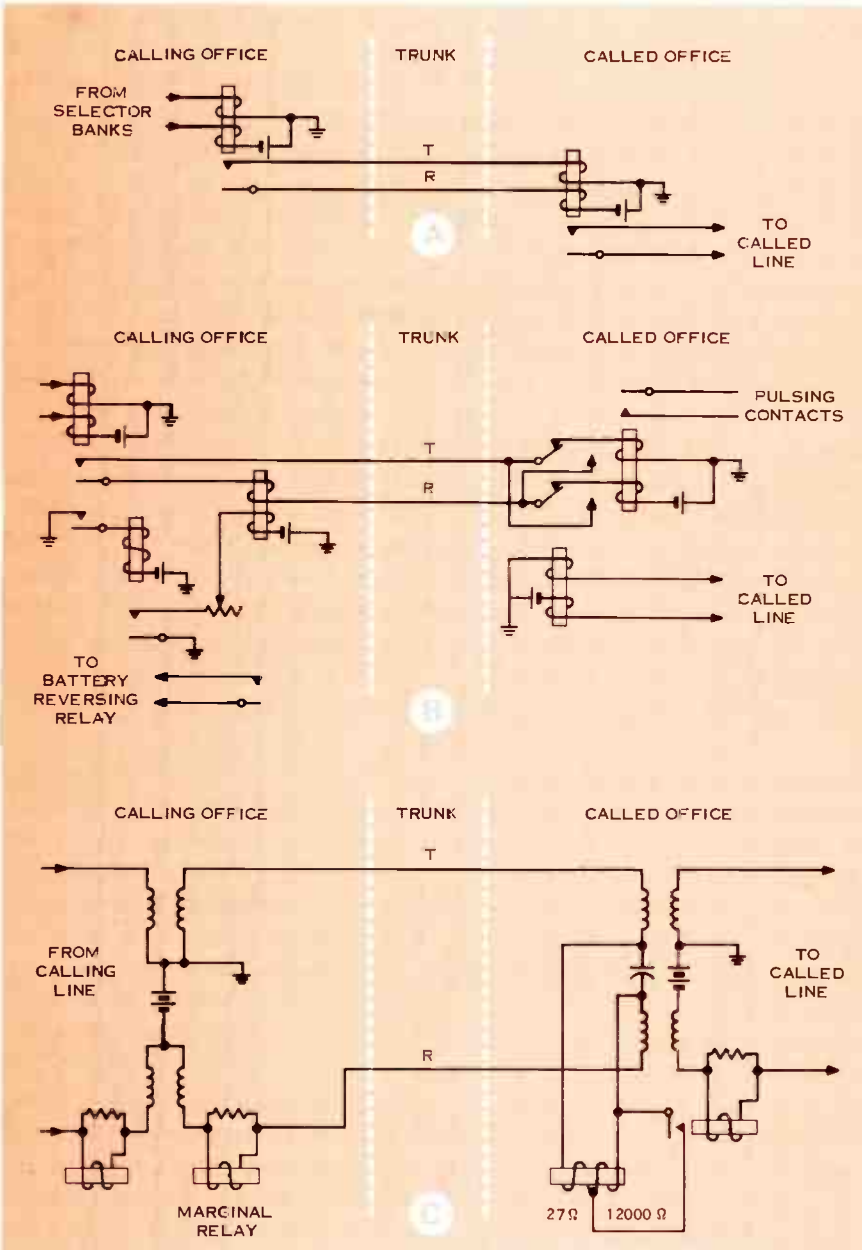

Figure 1. Loop signaling is accomplished by altering the flow of current in the trunk conductors. Three methods used to accomplish loop signaling are: (A) wet-dry, *(B)* reverse battery, *and (C)* high-low.

represents 0). Dial pulsing is used in all types of switching offices.

Revertive pulsing was originally developed for use in panel switching offices. In this type of pulsing, the address pulses are not transmitted by the originating office. When a call is made, a loop to the distant office is closed. This starts the movement of a panel selector switch at the distant office. As the selecting wipers pass each terminal, a commutator transmits pulses back to the sender at the originating office. When the proper number of these *revertive* pulses, corresponding to the called number, are received by the sender, a signal is sent back to the distant end to stop the movement of the selector. Revertive pulsing is used in certain crossbar offices as well as panel offices.

Panel call indicator (PCI) pulsing is a method of transmitting address signals between a dial office and a manual office. This technique converts pulses received from a dial office to lamp indications which appear on a switchboard. The switchboard operator then connects the incoming call to the called number and rings the subscriber.

Multifrequency (MF) pulsing is the newest method of transmitting address pulses between switching offices. Digital information is transmitted in the form of short tone bursts. Six signaling frequencies are used, each digit being represented by a combination of two of the six frequencies. The signaling frequencies fall within the speech band and are simply processed through the trunk in the same manner as speech signals. (A different form of multifrequency pulsing has recently been introduced to subscriber loop circuits through the use of telephones with pushbuttons instead of the conventional dial.)

Historically, signaling systems designed to transmit supervisory and address information have evolved from simple dc systems operating over 2-wire short-haul interoffice trunks, to complicated ac systems operating over multichannel carrier and microwave transmission systems. Today, there are essentially two fundamental techniques used to derive signaling paths on trunk circuits. The first of these is known as *loop signaling*. This technique requires a dc loop, and is the method used in all subscriber loops and in most short-haul 2-wire trunks. The second signaling technique, known as *E & M*, is used with both ac and dc signaling systems on 2-wire or 4-wire physical trunk circuits, and on carrier-derived trunk circuits. This type of signaling is standard for use in all intertoll trunks.

Loop Signaling

Loop signaling is the simplest of the two, and is used in certain exchange trunks, short-haul toll-connecting trunks, and one-way dialing toll trunks, where 2-wire voice-frequency circuits are employed. The dc signaling current flows over the same conductors used for voice transmission.

This type of signaling is accomplished by simply interrupting the condition of a dc voltage applied to the line to transmit both supervisory signals and dialing information. The range of loop signaling is usually limited to about 25 miles because of the dc resistance of the conductors.

There are three methods currently used to apply loop signaling to a 2-wire voice-frequency trunk: *wet-dry, reverse battery*, and *high-low*. (See Figure 1.)

In the wet-dry method, signaling information is indicated by the presence (wet) or absence (dry) of a battery and ground condition on the line at the called end of the trunk. Normally in the wet condition, the battery is placed on the ring conductor and ground on the tip conductor.

As its name implies, reverse-battery loop signaling is accomplished by reversing the polarity of the battery on the line to indicate supervisory conditions. For one condition, battery is on the ring conductor and ground on the tip conductor. The opposite supervisory condition is indicated by reversing the polarity of the battery, thus causing a polar relay to operate or release at the distant end of the trunk. This is the most prominent type of loop signaling used between exchange offices. To increase the operating range of reverse-battery loop signaling, batteries are sometimes placed at both ends of the circuit, in series. This variation of reverse-battery operation is called *battery and ground* signaling.

The third method, high-low, is used principally for supervisory signaling within a central office or from an automatic to a manual office. This type of signaling employs a marginal relay. During on-hook condition, a high resistance is placed in the loop. For off-hook, the resistance in the loop is reduced to a low value allowing more current to flow, and thereby causing the marginal relay to operate.

E & M Signaling

As mentioned previously, loop signaling is limited to trunks of about 25 miles in length. Also, such systems do not provide simultaneous signaling in both directions. In order to overcome these limitations, and especially to extend the dialing range of telephones, another type of signaling was developed.

This method of signaling employs two leads to connect the signaling equipment to the trunk circuit. These two leads are designated E and M, respectively. The name for the two leads was probably acquired from designations appearing in early drawings for this type of signaling circuit. The M lead transmits battery or ground signals to the distant end of the circuits, while incoming signals are received on the E lead as either a ground or open condition. Thus, the M lead reflects conditions at the near end of the circuit while the E lead reflects conditions at the far end.

There are several methods of deriving an E and M circuit to permit signaling between offices on a dc basis. These arrangements are known as simplex (SX), composite (CX), and duplex (DX). A simplex signaling circuit is obtained by means of a center-tap coil placed at both ends of the voice-frequency trunk circuit, as shown in Figure 2A. Signaling currents flow in both directions through the coils and, therefore, do not induce any interfering voltages into the voice channel. Conversely, voice currents do not flow through the simplex conductors (or legs) extending from the center tap of the coils. Since the two trunk conductors provide a parallel path for the signaling current, the dc resistance is approximately one-fourth of that presented to a loop-signaling arrangement over the same trunk. Thus, the dc signaling range is extended considerably. However, simplexing has certain disadvantages and has been largely superseded by the duplex arrangement.

In the composite method, a filter is used at each end of the trunk to separate the signaling current from the speech signals. The filter is called a *composite set*. Two composite signaling paths can be obtained from the two conductors of a v-f trunk and four can be obtained from a phantom circuit arrangement. This type of signaling, shown in Figure 2B, is used typically

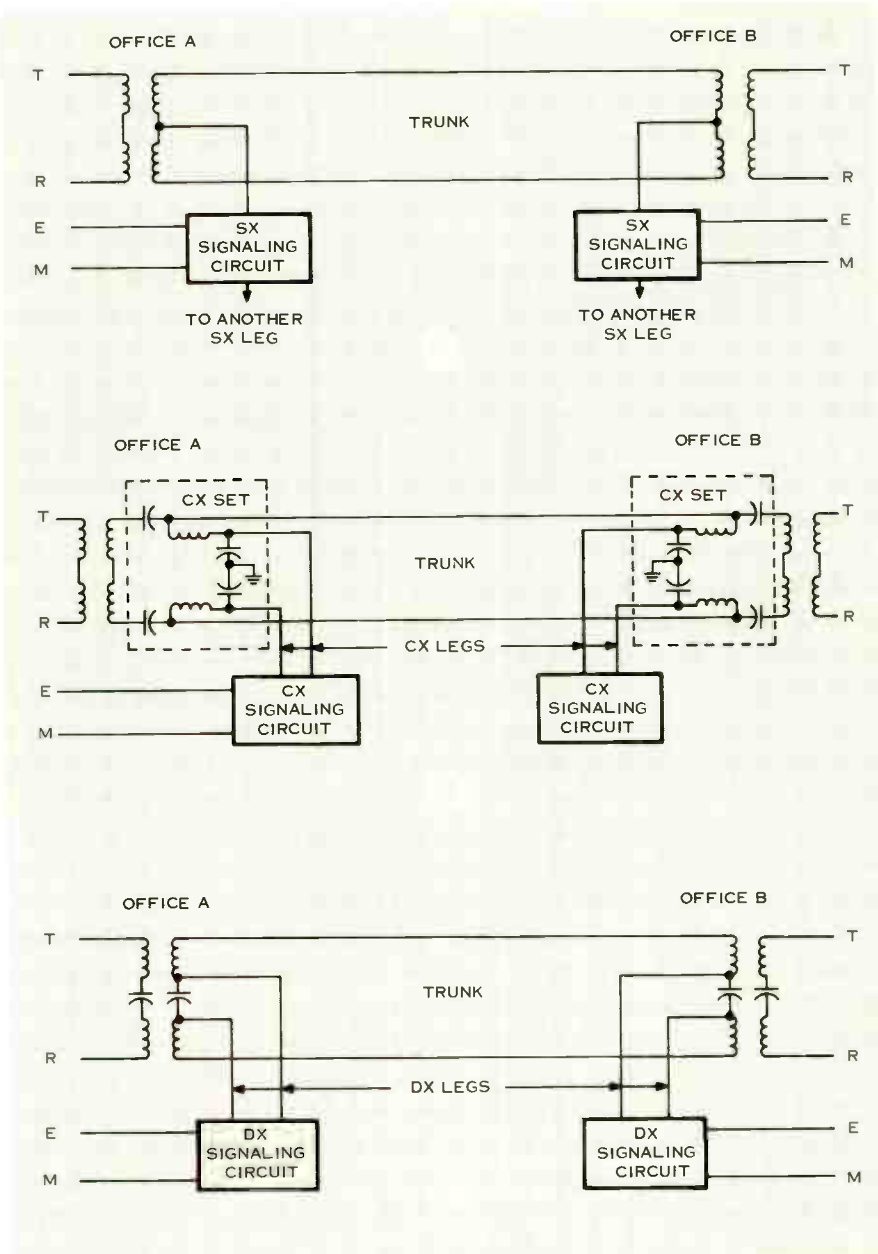

Figure 2. There are a number of different circuit arrangements designed for E & M signaling over telephone trunks. The three most prominent d-c arrangements are: (A) simplex, *(B)* composite, *and (C)* duplex.

on trunks derived from quadded cable where the conductors are arranged in phantom groups.

The duplex signaling arrangement, like the composite method, uses one conductor of the v-f circuit for signaling on a ground return basis, and the other conductor for ground potential compensation. Ground potential compensation is required because of the inherent instability of ground-return circuits. The composite set or filter, however, is not used with the duplex circuit. Instead, the signaling circuit is connected to the trunk pairs by means of a center-tap transformer and a capacitor, as shown in Figure 2C. This signaling arrangement is used primarily in paired cable trunks.

AC Signaling

The dc signaling systems described thus far are limited to relatively short v-f trunks containing a dc path. These systems are not suitable for use on long v-f trunks employing repeaters, or for carrier or multiplex trunk circuits because a dc path is not available. As a result, ac signaling systems had to be developed for use over the more modern exchange trunks and on the longer toll-connecting and intertoll trunks, especially where carrier is used.

The ac signaling systems use frequencies within the voice-frequency range so that the signals can be transmitted directly over the same path used for voice transmission. These ac systems usually employ E and M leads to connect the signaling circuit to the trunk. If the signaling frequency falls within the band used for speech transmission (typically 300 to 3400 Hz) the system is referred to as an *in-band* system. If the signaling frequency falls outside the speech band, the system is called an *out-of-band* system.

The ac systems must process the dc supervisory and address signals received from the switching office and convert them into ac signals for transmission over the trunk circuit. At the other end of the trunk, the ac signals must be converted back to dc signals before being applied to the switching equipment. Only one signaling frequency is required on 4-wire trunks. However, on 2-wire trunks two frequencies are required, one for each direction of transmission.

Early ac signaling systems used a frequency of 1600 hertz. On 2-wire trunks, 1600 hertz was used for one direction and 2000 hertz for the opposite direction. Later in-band systems used a frequency of 2600 hertz, with a second frequency of 2400 hertz for use with 2-wire trunks. The ac signaling frequencies easily pass through the same path used for voice transmission, and are amplified in repeaters in the same manner as speech signals. These so-called single-frequency (SF) signaling systems are used to transmit both supervisory signals and address or dial pulses. Multifrequency (MF) address pulsing, described previously, uses tones that are already in the voice band, so they do not require additional processing before being transmitted over a long-haul or carrier-derived trunk.

Signaling Over Carrier Channels

All trunk circuits equipped with carrier or multiplex equipment require some type of ac system for signaling. There are many different carrier signaling systems in use today employing either an in-band or out-of-band signaling frequency.

The most prevalent type of carrier signaling is accomplished with in-band frequencies. In-band signaling systems

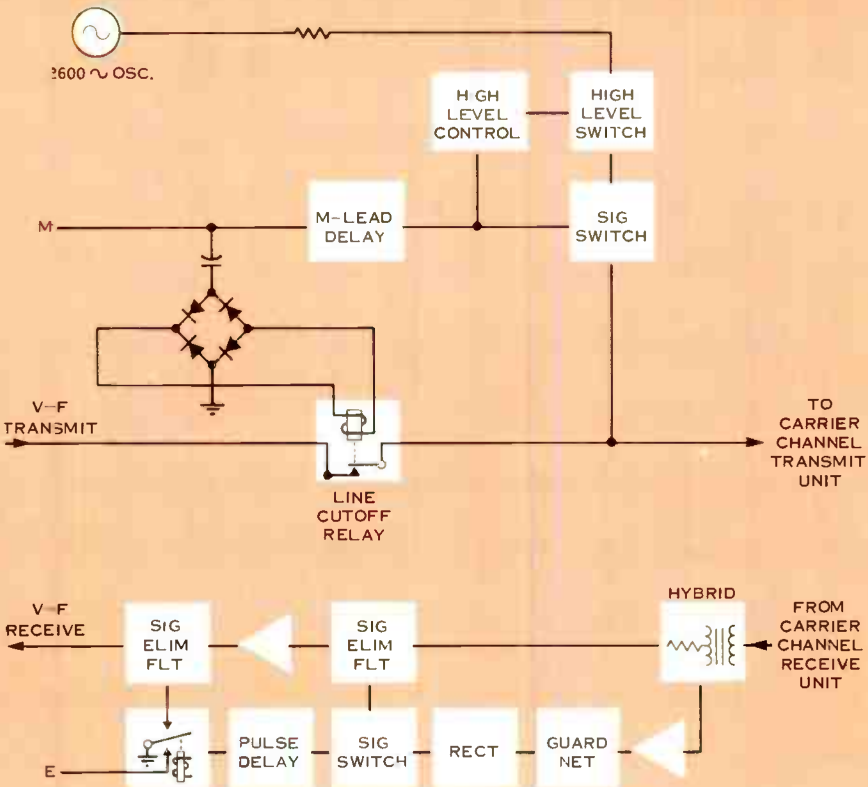

Figure 3. Simplified block diagram of a 2600-hz in-band signaling circuit applied to transmit channel and receive channel of a carrier system.

have an advantage over out-of-band systems in that they do not require extra bandwidth — the signals are passed directly through the voice channel. Another advantage is that signaling equipment is required only at the terminal stations of a trunk made up of several tandem links. Also, the in-band signaling system can be made a part of the office switching equipment rather than the particular carrier system, thus making it easier to patch trunk circuits to different carrier transmission systems.

The main disadvantage to in-band systems is that the signaling tones lie within the speech band. This leads to the possibility of speech energy at the signaling frequency "talking down" the signaling; that is, falsely operating the signaling equipment with speech energy. Protection against "talkdown" can be accomplished by using a time delay or guard circuit in the signaling system. By introducing a delay, the signaling circuit can be made insensitive to most voice energy or transient noise at the signaling frequency.

Additional protection is obtained by properly selecting the in-band signaling frequencies. Generally, it is desirable to use the highest possible frequency that will pass through the voice channel. Speech energy declines rapidly at the higher frequencies, thereby reducing the chances of "talkdown".

Most of the older voice-frequency telephone circuits use filters with an upper frequency cutoff of about 2800 hertz. For this reason, the most commonly used frequency for SF in-band supervisory and address signaling is 2600 hertz. In-band carrier signaling systems can be adapted for use with either loop signaling or E & M signaling arrangements.

Following the development of economical short-haul carrier systems, the need arose for inexpensive methods of signaling. This need resulted in the development of various out-of-band signaling systems. Out-of-band signaling equipment is generally less expensive than in-band equipment and also permits signaling during speech transmission, thus permitting extra functions such as regulation to be performed. Since the signaling frequency is outside of the speech band, there is no need for

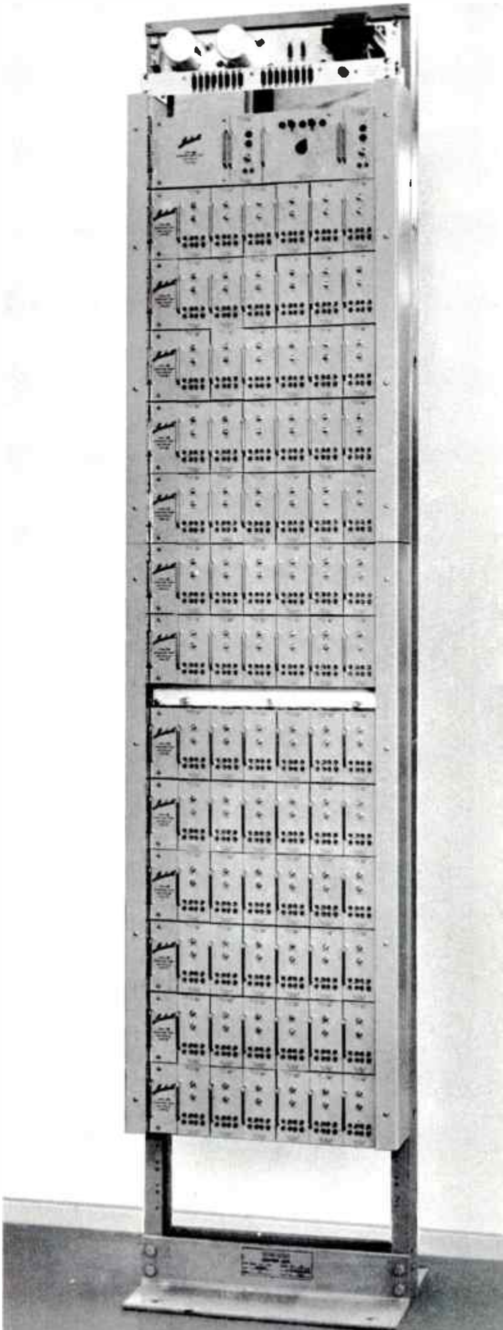

Figure 4. Photograph of typical in-band signaling units used with carrier systems. One signaling unit is required for each carrier channel. Many types of signaling units are required to accommodate the many different methods of signaling.

complicated guard circuits to prevent talkdown.

With out-of-band signaling, voice channel filters are designed with an upper cutoff frequency well below the top edge of the channel. This leaves the top portion of the channel passband available for transmitting out-of-band signaling tones. The most prevalent frequencies used for out-of-band signaling are 3700 hertz, which is standard throughout the Bell System, and 3825 hertz, which is recommended by the International Telegraph and Telephone Consultative Committee (CCITT) for use in international circuits.

Unfortunately, out-of-band signaling has certain disadvantages which tend to limit its use. Out-of-band signaling equipment has to be built-in to the carrier channel equipment and cannot be separated as in the case of in-band signaling equipment. This condition prevents randomly patching the circuit to other trunks.

Also, out-of-band signaling requires some sort of dc repeater at the end of each link of a multi-link trunk. As the signal passes from one link to another, the signal pulses must be detected and then made to operate a relay. The relay, in turn, keys the signaling equipment in the succeeding link. Thus, signaling equipment is required at both ends of each link in the trunk.

Another economical type of signaling, using time division multiplexing techniques, is used in Lenkurt's 81A exchange trunk carrier system. This unique method provides signaling for all 24 voice channels of the system using one common signaling channel. Each voice channel is assigned a specific time slot for signaling, and all 24 slots are scanned 500 times per second. The resulting signaling frequency modulates a pilot in the carrier system that is also used for slope regulation.

Although out-of-band and time division multiplex signaling techniques may be more economical for certain short-haul trunks, they lack the flexibility and other advantages offered by in-band signaling systems, especially when applied to long-haul trunks. As a result, single-frequency (SF) in-band signaling for supervisory functions and multifrequency (MF) pulsing for address functions have become the standard methods of signaling in modern interoffice, toll-connecting, and intertoll trunks.

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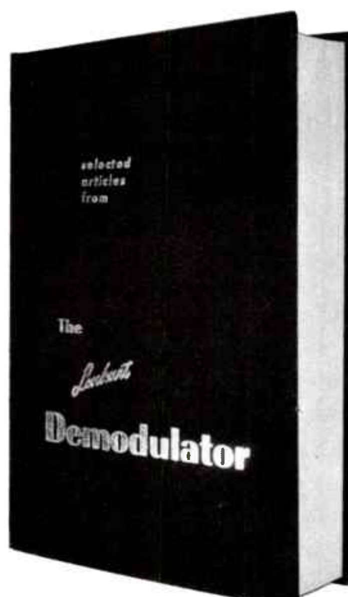

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AUGUST, 1966
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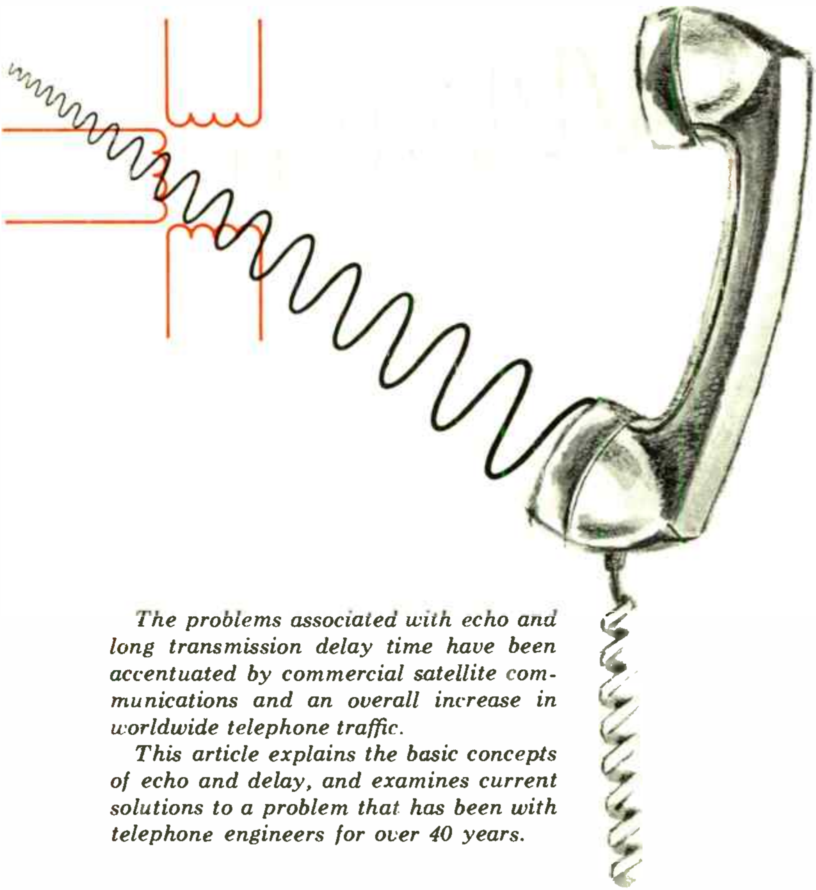
the Lenkurt Demodulator

Echo Suppression



The problems associated with echo and long transmission delay time have been accentuated by commercial satellite communications and an overall increase in worldwide telephone traffic.

This article explains the basic concepts of echo and delay, and examines current solutions to a problem that has been with telephone engineers for over 40 years.

Like the acoustic echo heard in a cave, or bounced from the side of a mountain, the echo in a long telephone circuit represents sound energy reflected back from some distant point. Unlike that found in nature, telephone echo is neither a pleasant nor desirable occurrence.

Telephone echo is created primarily at the far end of 4-wire transmission circuits where a junction is made with 2-wire subscriber loops. Because of unavoidable impedance mismatch at this point, energy transfer is not complete and some of the sound is reflected back to the talker. Thus, in the telephone receiver a talker hears his own voice, delayed proportionately to the length of the circuit.

Delay Time

Delay time, the basic factor causing echo to be objectionable, is a function of propagation rate and distance. The faster the propagation, the longer the distance that can be covered without serious degradation of the circuit. The upper limit, however, is the rate at which electromagnetic radiation travels in free space — 186,000 miles per second. And now communications satellites are proving that even this is not fast enough.

Delay is commonly measured in milliseconds (ms), or thousandths of a second. Long one-way delays of, say, 500 ms might be sensed by talkers only as a hesitancy in the response of the other party. But echo returned at a fraction of this time seriously degrades the circuit, and may cause the speaker to stammer, slur his words, or stop talking altogether. In fact, a round-trip echo delay of 45 ms is considered the maximum before some sort of echo suppression must be used. Of course, the speaker's tolerance to echo depends on both echo delay time and loudness. Echo with long delay and of sufficient magnitude is very noticeable.

Undoubtedly, the commercial satellite system of the future will be built around synchronous satellites whose orbital speed matches exactly the rotational speed of the earth. The satellite appears to "hang" in a stationary location over the equator at an altitude of 22,300 miles. A one-way telephone path through such a satellite is in the order of 50,000 miles, taking into account the geographical distance between ground stations. Delay between New York and London, for instance, is about 265 ms, or over a half-second for round-trip echo. This amount of delay has again accented the problem of echo suppressors.

Signal Path

The telephone path begins with a 2-wire loop from the subscriber's instrument to the local exchange office, and either 2- or 4-wire exchange trunks extending to the toll switching center. Long-haul 4-wire toll trunks eventually are returned to 2-wire subscriber loops at the receiver end of the circuit to complete the path to the called party's telephone. (See Figure 1.) In addition, signals are commonly returned to 2-wire circuits for more economical switching purposes.

Wherever it is necessary to match 2-wire to 4-wire circuits, hybrid transformers are used (Figure 2). Because of impedance irregularities at the hybrid, a certain amount of reflection occurs and echo is produced. Of primary concern is the echo resulting from the mismatch at the far end of the circuit.

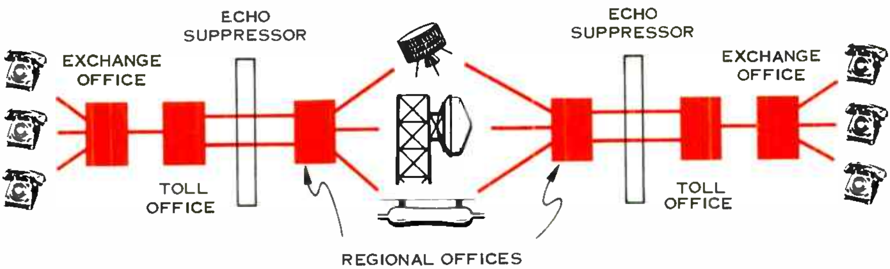


Figure 1. Telephone path through exchange, toll and regional offices. Echo suppressor is located at the beginning of 4-wire toll circuits.

Echo Return Loss

The terms transhybrid loss, return loss, and echo return loss are often used in discussing the operation of hybrid 4-wire terminations. Transhybrid loss is defined as the loss directly across the hybrid on the 4-wire side; i.e., the isolation between the transmit and receive branches of the 4-wire line. The return loss is the transhybrid loss *minus* the losses in the 4-wire to 2-wire paths of the hybrid and serves as a measure of energy returned to the talker. Echo return loss is measured with a known input of voice-weighted noise, between 500 and 2500 Hz. Echo return loss must be greater than 27 dB to meet toll switching requirements. The mean echo return loss at end offices is about 10 dB. (For further discussion, see the *Demodulator,* January 1964.)

It is possible to limit echo in short delay circuits by adding path attenuation. This loss will, of course, reduce the talker's signal as well, but echo returned through the same path will be attenuated twice as much. Until a few years ago circuits with up to 45 ms delay could be accommodated by the insertion of up to 14 dB total one way loss, known as terminal net loss (TNL). Beyond that point, further attenuation interferes seriously with the transmission of speech. Recent upgrading of nationwide service has reduced the amount of attenuation to be placed in high usage trunks. It is now policy to use echo suppressors in circuits having even 20 ms delay, or less.

First Suppressors

Original echo suppressor work began in the 1920's when the first transcontinental telephone systems were being planned. Carrier equipment was not commonly available and most transmission was at voice frequency, having the relatively low propagation rate of about 20,000 miles per second on loaded cable. At that speed a signal would travel a 900-mile circuit from New York to Chicago in 45 ms, or have an echo delay of 90 ms. As trunk lines were expanded across the country, the need for echo suppression became more and more obvious. But before echo equipment approached any degree of sophistication, carrier systems were developed, with their higher frequency increasing the propagation rate to over 110,000 miles per second. At this speed New York to Chicago is only 8 ms, or 16 ms echo delay.

The innovation of carrier transmission and its characteristically faster propagation speed greatly reduced the pressure on further echo suppressor development. Only years later as coast-to-coast trunks and longer undersea cables were established, did interest in echo suppression again take on new zeal. Even at 110,000 miles per second the round-trip echo delay from New York to London was about 70 ms—far too much to be handled by terminal net loss techniques.

Since it was impossible to reduce delay, impractical to eliminate echo at terminal points, and not feasible to introduce sufficient path loss to suppress echoes without reducing talker volumes to imperceivable levels, only one solution became immediately obvious. It was necessary to block the return path an echo must take. The first echo suppressors did just that, using amplified voice energy to activate a relay shorting the opposite path. More refined versions, such as the Bell 1A echo suppressor, detected relative speech energy between the two paths, and picked the stronger of the two to activate suppressor controls. In this case, rather than shorting the line, the suppressor introduced 40 to 50 dB loss in the return path. Because it was almost impossible for a second speaker to break in on a conversation, the circuit took on the qualities of a "push-to-talk" or simplex operation.

Split Suppressors

Echo suppressors were originally designed to be placed at the midway point of the voice-frequency circuit, but with carrier transmission and ocean cables, this was not feasible. First, suppressors were moved from the midpoint to one end of the circuit. Later, the "split" suppressor allowed one-half of the unit to be located at each end, providing some advantage. In all suppressors, attenuation is maintained after the party stops talking for the period of time necessary for the signal to make its round trip and return to the suppressor. This is called *hangover* time. By using split suppressors, hangover time can be held to a minimum, since one unit is always near the reflection point at the far end. Today, echo suppressors are commonly placed in toll or regional switching offices, and may also be found in future communications satellite ground stations.

One problem inherent in previous suppressors was *lockout*. If circuit switching resulted in two suppressors working in tandem—not at all unlikely, especially with Direct Distance Dialing—it was possible for each talker to take command of the suppressor nearest him, block the opposite

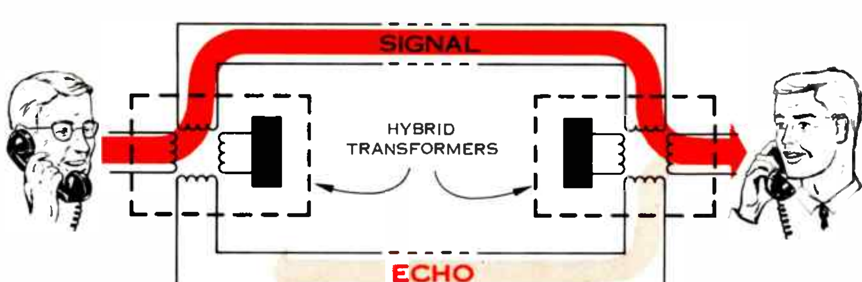


Figure 2. Hybrid transformers are needed at each end of a 4-wire circuit. Reflected energy at this point becomes echo.

path, and thereby prevent any further conversation until one of the parties stopped talking. Split suppressors have also reduced this possibility, but an extension of the problem still exists. A series of echo suppressors introducing individual path losses of about 12 dB could quickly add enough attenuation to reduce speech levels below acceptable values.

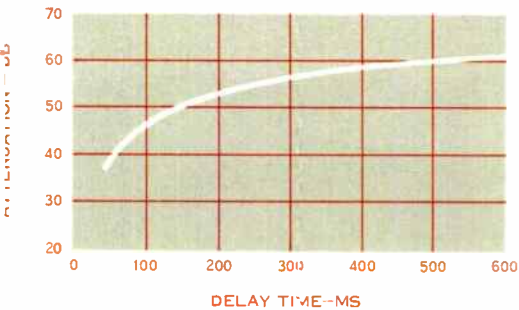


Figure 3. The attenuation needed to suppress echoes to tolerable levels is indicated for the average telephone talker.

Lenkurt 931B

In developing the 931B Echo Suppressor, Lenkurt design engineers were able to include the desirable feature of bi-directional operation — both parties talking at the same time—and minimize speech clipping at the beginning, and speech chopping in the middle of conversations. All modern echo suppressors have since adopted this bi-directional mode of operation.

The 931B suppressor has two modes of operation, allowing it to discriminate between single-party talking and two-party talking. With only one subscriber talking the suppressor is in Mode 1, and blocks echo by inserting 60 dB loss in the return path. Mode 2 provides for two-party talking by only partially suppressing echoes in both paths, on the assumption that echoes of short duration will be masked by the speech signals of the other party.

Mode 1

Block diagrams of the Lenkurt echo suppressor in both modes are shown in Figure 4. Identical units are placed at each end of the communications link, and contain two variable-gain amplifiers, an echo control switch, and associated control circuits. To analyze the set's operation, assume Station A is talking. A's voice operates the transmit control circuit closing both loss switches, producing a no loss, or unity gain condition in the transmit path. The echo control switch is normally closed.

At Station B, the signal operates the receive control circuit, opening the echo control switch. The loss switch remains closed and the signal arrives at the second talker unattenuated by the suppressor. The echo signal, reflected at the hybrid, is sensed by the transmit control circuit at Station B, but because of loss in the hybrid, lacks the energy to overcome the variable reference bias supplied from the receive path and is blocked by the echo control switch.

Mode 2

Mode 2 occurs when B attempts to interrupt A's conversation. B's speech energy is high enough to operate the transmit control switch on his end of the circuit. This results in closing of the echo switch, and the opening of loss switches in both paths at Station B. Similar action occurs as B's signal arrives at Station A. In Mode 2, with both parties talking, each speech signal is attenuated by two losses, and each echo signal reduced by four losses. These losses may be strapped in 1-dB

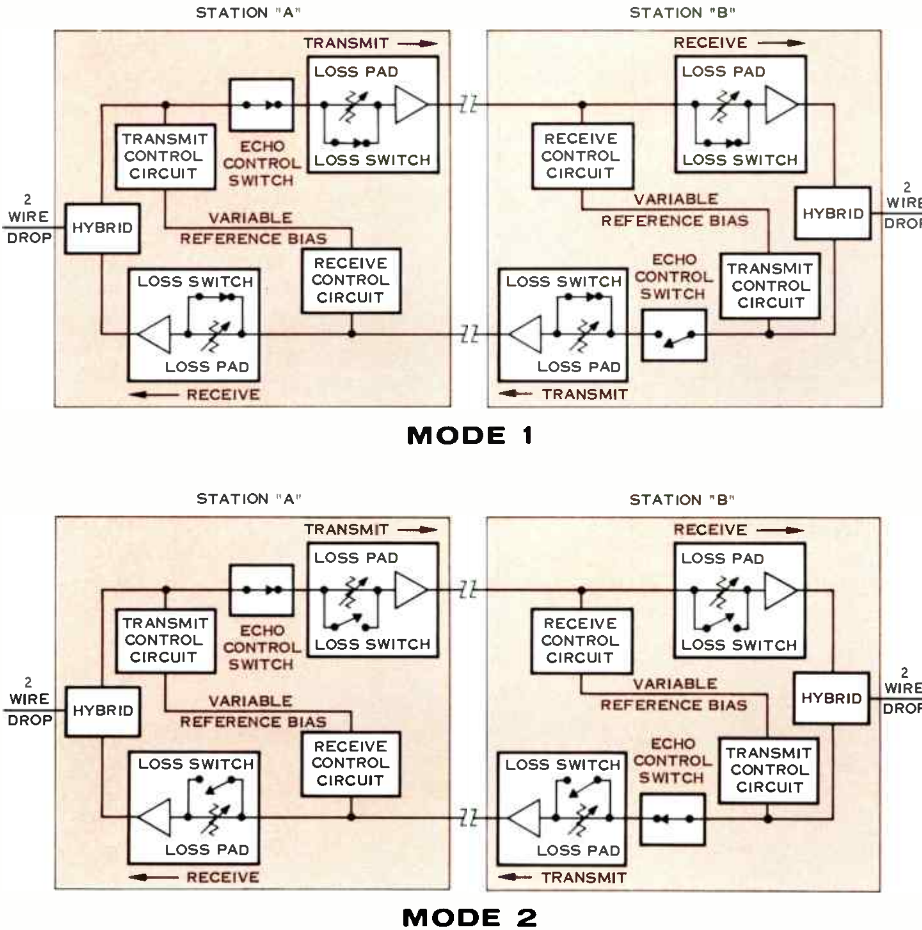


Figure 4. Simplified diagram of Mode 1 (uni-directional) and Mode 2 (bi-directional) operation of the Lenkurt 931B Echo Suppressor.

steps up to 6 dB, depending on the length of the circuit. If set for 6 dB, the talker's signal is reduced a total of 12 dB, while echo is suppressed 24 dB — this in addition to loss in the hybrid and other circuit losses.

The hangover time in the operation of the echo control switch is a compromise between echo and speech chopping. In Mode 1, the echo control switch at Station B is open as long as A is talking. When A stops talking the switch must be closed to allow B to talk. The time period before this happens is known as receive hangover and is set at 40 ms in the 931B.

In Mode 2, A is talking and B interrupts. The echo control switch at Station B closes to allow B's conversation to be transmitted. If B stops talking but A continues, the switch must again open to block any echo signal to A. If this time, known as transmit hangover, is too short chopping will result; if it is too long, A will hear a few moments of "trailing" echo. Transmit hangover is set at 85 ms in the Lenkurt echo suppressor as an adequate compromise.

In addition, it should be noted that since all operations consist of a *change* in attenuation rather than the actual opening and closing of the circuit, the recycling time between modes is not critical.

The Bell 3A echo suppressor operates similarly, using rectified speech energy through a differential detector to activate mechanical relays. During Mode 2 operation echo attenuation is introduced by a speech compressor, inserting loss in the receive path proportional to the level of the incoming signal.

Data Disabler

With the increasing use of voice circuits for data transmission, it is necessary to provide some means of removing the suppressor from the circuit. This is accomplished with a disabling circuit activated by the transmission of a continuous tone between 2000 and 2250 Hz for approximately 400 ms. Both paths are then held open for data until there is no signal for at least 100 ms. The suppressor then reverts to normal action. Note that echo delay presents no problem in one-way transmission, such as data or television.

Long delay and echo have been the subject of a number of recent tests performed by General Telephone and Electronics Laboratories, Bell Telephone Laboratories, Stanford Research Institute in Palo Alto, California, and others. Interestingly, talkers participating in the experiments became "sensitized" to the problems associated with exceptionally long delays, say of 1200 ms, and thereafter tended to be less tolerant of circuits with shorter delay. However, typical delays of 600 ms found in operating communications satellites apparently have not produced sensitizing to any noticeable degree and most customers find such service fully acceptable.

The Future

Authorities expect that round-trip delays longer than 600 ms must be avoided in future worldwide telephone links by limiting satellite systems to one hop. The practice in coming years may very well be to use satellites for only a portion of an around-the-world conversation, relying on conventional land circuits for the remainder of the path.

Generally, results of experimental testing and of actual performance in operating systems demonstrate that modern echo suppressors have satisfactorily eliminated undesirable qualities found in earlier models. Equipment such as this will be a necessary standard in all long telephone systems of the future, especially when distances are amplified by commercial and military communications satellites.

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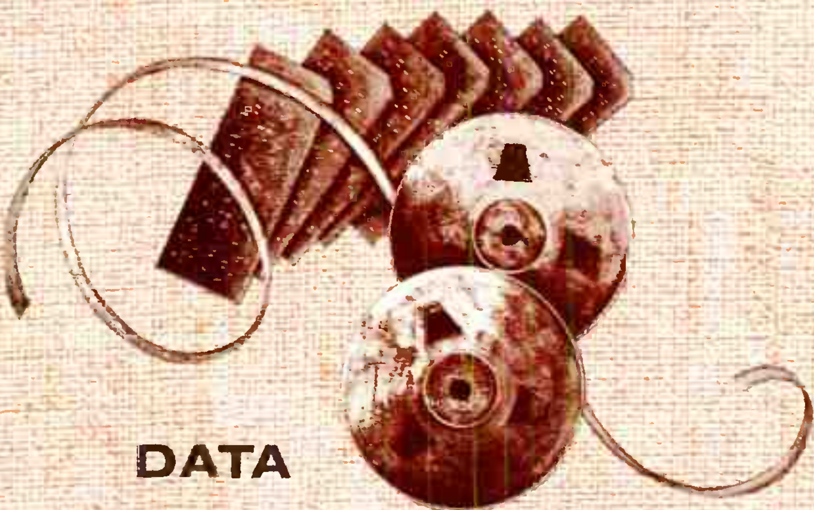


DATA COMMUNICATIONS

Part 1

Data communications is today a dynamic and rapidly expanding field stimulated by the increasing need to link electronic computers and other business machines across great distances. The resultant union of the data processor and the communicator has provided a vital service to the everyday operation of business, industry, and government.

This is the first of a two-part article which presents an introduction to data communications technology, and offers some insight into the future of this progressive field.

Less than one percent of the computers in service today are interconnected through a communications network. However, it is estimated that ten years from now at least half of the computers in operation will be working together on a real-time basis. It is also anticipated that the volume of digital data transmitted over communications facilities will eventually equal and perhaps exceed the volume of voice traffic.

A little over a century ago the first data message was transmitted on wire lines by Morse Code. But this was not the beginning of the realization that the spoken word could be represented by some analogous language. Ancient records confirm that semaphore-type data or information transmission systems using the visual sense for perception existed even before the Greek and Roman empires.

But what is data? It might be described as factual information required as the basis for making decisions. Thus, statistical reports, engineering documents, and historical records all contain data. Data covers a broad range of information and plays an important part in the decision-making processes of our everyday lives. In this discussion, however, the meaning of data is limited to digital forms of information used in machine-to-machine communication.

The economics of computers and other types of business machines are based on moving information to achieve optimum use of what are usually expensive facilities. Some of the large-scale computers now in service are capable of input rates as high as 10,000,000 bits per second (b/s). Bit, a contraction of *bi*nary digi*t*, expresses a unit of information in a two-element binary code. The elements are called "mark" and "space", and indicate the choice between two equally possible events.

The requirement for data communications arises because modern business machines and computers can record and store information more efficiently, more accurately, and significantly faster than can humans. At present, thousands of data messages are transmitted over telephone networks at speeds many times faster than could be achieved by human speech.

Communicating with Data

Data signals are transmitted over various types of telephone circuits. They travel on wire from telephone pole to telephone pole, through underground cables, from mountain top to mountain top over microwave facilities, on the ocean floor in submarine cables, and via communications satellites from continent to continent. Some type of data conversion equipment is required to change the digital machine signals to a form suitable for transmission over these facilities.

The data machine which provides an input to the transmit section of the conversion equipment, or *modulator*, can be a keyboard, printer, card reader, paper tape terminal, computer, or magnetic tape terminal. The output from the receive section of the converter, or *demodulator*, can be applied to a tape punch, printer, card punch, magnetic tape unit, computer, or visual display terminal. Typically, both the modulator and demodulator sections of the converter are combined into a two-way data transmitter-receiver, commonly called a *data modem* or *data set*.

Figure 1 illustrates a typical full-duplex data transmission system including the originating data processing equipment and the interface assembly which consists of buffer and control units. The interface assembly at the transmitter accepts data at a rate determined by the operating speed of the data processor, stores the data temporarily, and regenerates it at a rate compatible with that of the data modem. At the receiving terminal the interface assembly accepts the received data, stores it, then feeds it to the data processor at the appropriate rate.

Timing signals from the interface assembly at the transmitter are applied to the data modem to synchronize the computer and the data set. At the receiver, synchronization pulses are derived from the data stream to synchronize the computer.

When more than one data set feeds into a computer, the capacity of the interface equipment is of major concern since it must determine the time slot allocation for each line. Various types of interface assemblies are employed, such as magnetic core memories, shift registers, and delay lines. Not all data communications terminals employ an interface between the data processor and the data modem. Without an interface the input, data transmission, and output functions proceed simultaneously and at the same rate of speed.

Since data signals are rarely in suitable form for transmission over the various types of transmission facilities, a signal coding process is normally performed. Ideally, the transmission medium should have linear attenuation and delay characteristics, but this is never so in practice, and transmission impairments are always present to disturb the data signals. As a comparison, in voice communications a high degree of transmission irregularities can be tolerated. If a voice circuit has a heavy loss or is noisy, the speakers compensate automatically by increasing the intensity of their voices. If words are missed because of transmission difficulties, they are often understood anyway because of the redundant nature of speech. In contrast, there is no inherent redundancy in data signals unless purposely inserted and, therefore, trans-

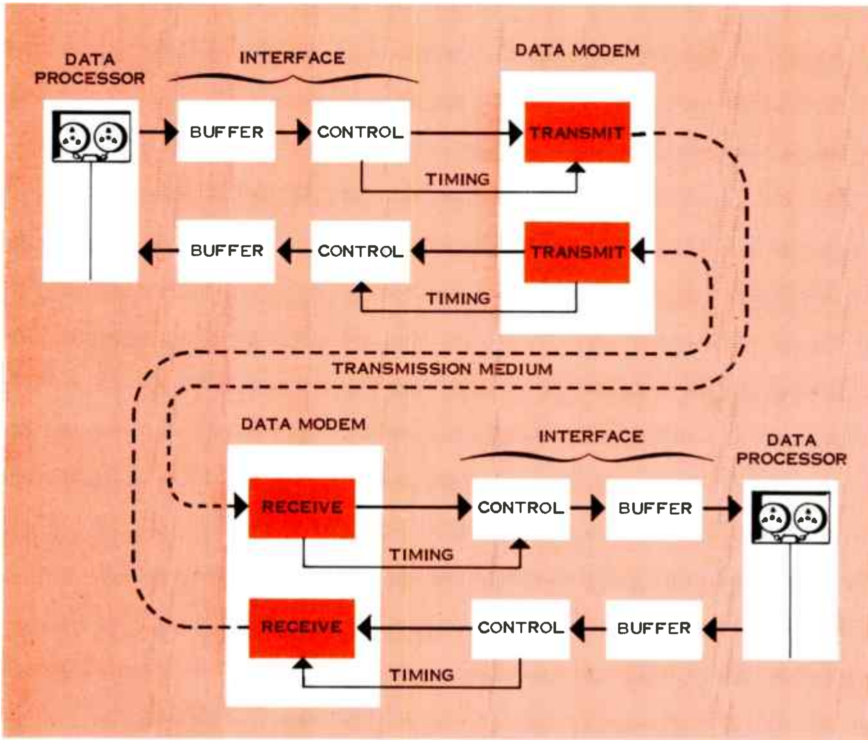


Figure 1. Typical full-duplex data transmission system arrangement. Timing signals from control unit synchronize data transmitter with data processor. At receiver, the element of time is important in reconstructing the original digital representation of the data. This is accomplished by deriving sync pulses from the data stream.

mission variations can only be compensated for over a very small range. In addition, data signals are sensitive to other transmission impairments which have little effect on speech.

Coding is undertaken to alleviate transmission irregularities, to increase the information capacity of the system, to enable error detection, and to provide message security. The coding process in the data transmitter (usually called *encoding*) simply rearranges the applied data machine signals into some other format. At the receiving end the reverse process *(decoding)* is performed to recover the original machine signals.

The diagrams in Figure 2 show the two types of information signals that are applied in digital form to a data modem. Shown in *A* is a binary *non-return to zero* (NRZ) signal. In *B* the same signal is shown in the *return to zero* (RZ) format. The difference between *A* and *B* is that in *A* successive marks or spaces follow one another, whereas in *B* there must be a return to the space level between successive marks. The voltage values of marks and spaces are arbitrary and may be positive, negative, or both.

Telephone Facilities

When data communications developed, a long established voice transmission facility already existed, and logically included service to those locations that would be the terminal ends of a data communications network. To provide economical data communications service, consideration must be given to using existing transmission facilities. It would be financially impractical to establish a completely new data communications network where existing voice facilities could satisfy the need.

An example of the use of voice facilities is the "time sharing" of a computer by several users for different purposes. Although the computer serves each user in sequence, it appears that all users are handled simultaneously because of the high-speed of the computer. A typical time sharing system uses a keyboard printer to connect to a remote computer via a data set. Eventually, data transmission over voice facilities might allow the automatic payment of bills, ordering of groceries, and a variety of other household tasks.

There are times when the nature of the data to be transmitted may prevent using normal voice facilities because of such factors as speed, quality, and compatibility. In this case, the use of a microwave wideband communications facility or a narrow band telegraph channel — but not a voice channel — might be required. At present, however, telegraph and public telephone line facilities are most commonly used for data transmission because of their wide availability and economy.

Data generated at such speeds that transmission requires part or all of a 3-kHz voice channel is normally referred to as *voice-band data.* Within this classification, data rates of 200 bits per second or less are called *low-speed* data. Data rates from 2000 to 2400 bits per second are referred to as *high-speed.* Between the two, data is called

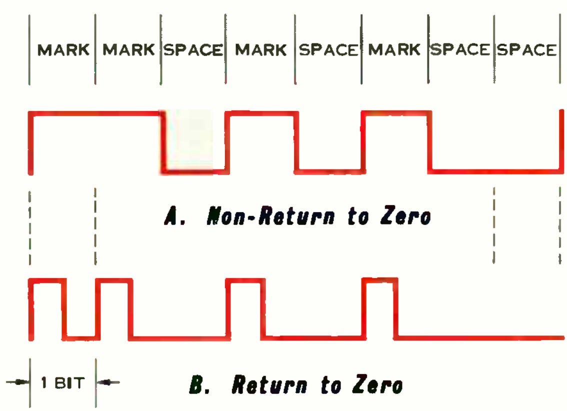
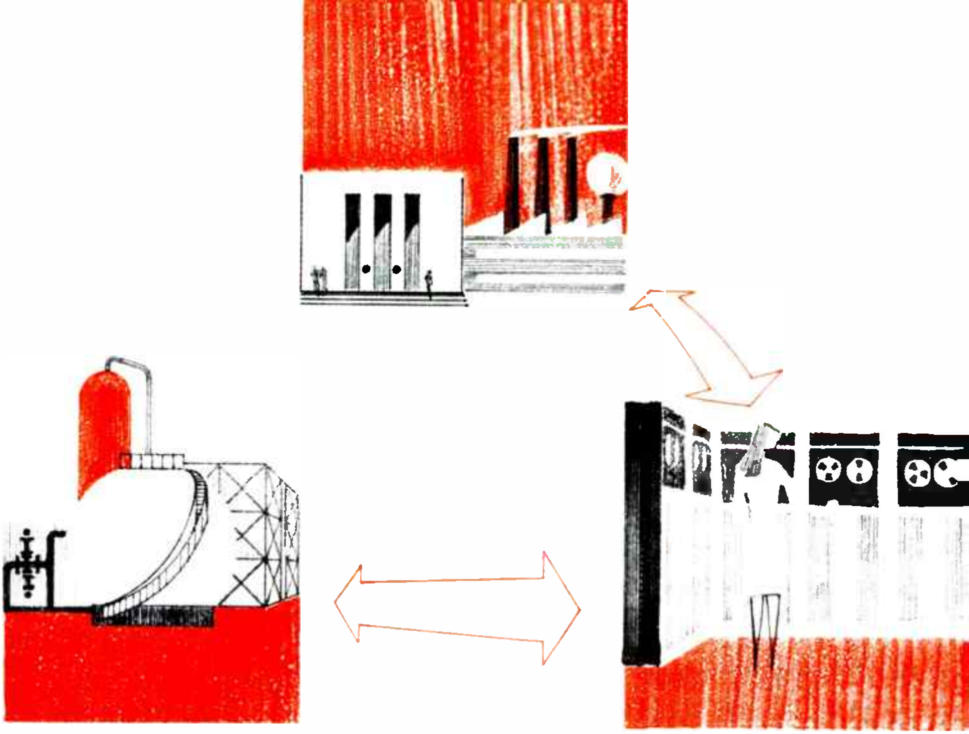


Figure 2. Digital representations of basic information signals.



medium-speed. For typical speeds and uses of voice-band data, refer to Table A.

Data signals at speeds requiring more bandwidth than a single voice channel are called *wideband*. The most popular use of wideband data terminals is for remote access in real time to high-speed digital computers. Many of the present wideband data communications systems

TABLE A. Typical Speeds and Uses of Voice-Band Data

Speed	Classification	Use	Number of Circuits Per Voice Channel
75 b/s (120 Hz channel spacing)	Low-speed	5 level 100 wpm teletype 5 level 60 wpm teletype Variable frequency telemetering Pulse duration telemetering Alarm and control	25
110 b/s (170 Hz channel spacing)	Low-speed	8 level 100 wpm ASCII coded teletype All applications of 75 b/s speed	18
200 b/s (340 Hz channel spacing)	Low-speed	Data collection networks (remote to computer)	7
1200 b/s 2400 b/s	Medium-speed High-speed	Computer to computer Secure voice vocoders Pipeline telemetry and control	1

With the increased decentralization of business and industry, and with the need for a worldwide government digital network, data communications extends the services of data processing equipment far beyond the confines of a single office.

operate with 48- and 240-kHz bandwidths, which are the group and supergroup allocations of common multiplex systems. These multi-voice channel allocations may be used for regular voice traffic during busy periods, and during normally slack times used as a single wideband channel for data transmission.

Because cost and not speed ordinarily determines what type of system can be used efficiently, data rates within the voice-band and wideband classifications can vary to a broad extent. Why produce a highly complex and expensive wideband data set when an economical lower speed system will serve equally well? There are over 160 different types of data sets now being manufactured. These operate with approximately 16 different transmission codes, at least 12 different transmission speeds, and numerous methods of error detection and correction.

Transmitting Information

During the past fifty years several investigations have been made concerning the theoretical digital signal capacities of communications channels. In the late 1920's, H. Nyquist, a mathematician at Bell Telephone Laboratories, established a relationship between the bandwidth of an ideal *rectangular* distortionless communications channel and the speed of digital transmission. (Rectangular refers to the bandpass characteristic of a channel — linear throughout the band, with sharp attenuation at the ends.) Nyquist showed that the signaling rate in bits per second is equal to twice the bandwidth in hertz of a lowpass ideal rectangular channel. For example, using Nyquist's criterion, the normal 3000-Hz bandwidth telephone transmission channel could handle a maximum of 6000 bits per second. However, it was realized that the distortionless conditions laid down by Nyquist were ideal and could not be achieved in practice.

Later, C. E. Shannon, then at Bell Telephone Laboratories, examined how much information a channel of given bandwidth would pass in the presence of noise. Shannon's analysis yields a rate of nearly 30,000 b/s for an average telephone channel with a good signal-to-noise ratio. Shannon did not provide a practical means of achieving such transmission capacity, and Nyquist's rate has not been attained in modern data communications. In contrast to the idealized rectangular model of Nyquist, the actual physical channels are not rectangular but have gradual cutoff characteristics and, therefore, require about twice the Nyquist bandwidth, or approximately 1 cycle per bit for optimum binary transmission. (For more detailed information concerning Nyquist's and Shannon's formulas, refer to the April and May 1965 issues of *The Lenkurt Demodulator.*)

Bits and Bauds

The speed of signaling, measured in terms of the amount of information transmitted per unit time, depends on the transmission path and its associated apparatus. Bits per second expresses the total number of *information* pulses in one second and includes redundant bits used for checking errors. If the pulses are of varying length, or if start and

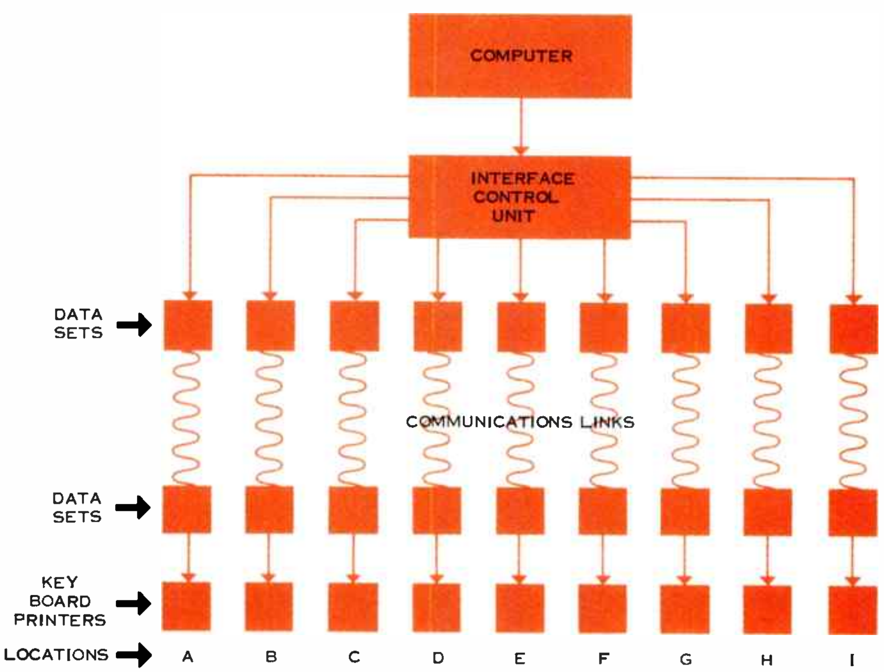


Figure 3. A typical time sharing computer employs an interface control unit to apportion its time among many users at different locations.

Figure 4. Data communications terminal (arrow) permits fast and reliable transmission at 2400 bits per second from a central computer.

stop pulses between each character are added that are not part of the message, the bit rate tells nothing of their number or duration. On the other hand, baud — from Jean Maurice Emile Baudot, an officer in the French Telegraph Service who contributed to early telegraph principles — is defined as the reciprocal of the time of the shortest signal element in a character. The term baud is often misinterpreted as a synonym for bits per second. However, the number of bauds equals the number of bits per second only when all time intervals are constant, and all signal pulses are information pulses, such as in binary transmission.

An example of the relationship between bits and bauds is in ordinary teletypewriter transmission, which makes use of a five-bit code, each bit being 13.5 milliseconds in length. The baud rate is therefore the reciprocal of 13.5 milliseconds, or approximately 74.2 bauds. A single character consists of a start pulse and the five information pulses or bits, each of 13.5 milliseconds duration, for a total of 81 milliseconds. A stop pulse of 19 milliseconds ends the character. The total time for a character is then 100 milliseconds.

Since the bit speed depends on the number of *information* pulses transmitted per unit of time, the equivalent rate for this type of transmission is 5/100 ms, or 50 bits per second.

Now, if a lapse period of 20 milliseconds is arbitrarily inserted between the stop pulse of this character and the start pulse of the next, the bit rate would be reduced to $\frac{5}{100 \text{ ms} + 20 \text{ ms}}$

or 41.7 bits per second. However, the teletype speed would remain at 74.2 bauds, because the baud rate depends only on the time length of the shortest pulse (13.5 milliseconds) in the character.

The number of words per minute can be determined using the ordinary telegraph definition of a word, which is 6 characters. The speed in bits per second is converted into bits per minute by multiplying by 60; hence, 50 bits per second equals 3000 bits per minute. Since there are 5 bits per character and 6 characters per word, there is a total of 30 bits per word. Dividing 3000 bits per minute by 30 bits per word equals 100 words per minute. Here again, the transmission rate in words per minute could be reduced by a slow teletypewriter operator, but the signal speed remains at 74.2 bauds.

It can be concluded that the baud rate is very important to the telephone engineer, since this rate establishes the type of telecommunications channel to be used. To a lesser degree the computer engineer is concerned with baud rate, but economics and speed of information flow are uppermost in this technology. Hence, to him the bit rate is the major concern, and is the more common expression used in dealing with binary data transmission.

Serial and Parallel Data

The terms *serial* and *parallel* are often used in descriptions of data transmission techniques. Both refer to the method by which information is processed. Serial indicates that the information is handled sequentially, similar to a group of soldiers marching in single file. In parallel transmission the information is divided into characters, words, or blocks which are transmitted simultaneously. This could be compared to a platoon of soldiers marching in ranks.

The output of a common type of business machine is on eight-level punched paper tape, or eight bits of data at a time on eight separate outputs. Each parallel set of eight bits comprises a character, and the output is referred to as *parallel by bit, serial by character.* The choice of either serial or parallel data transmission depends, of course, on the customer's data processing equipment and the transmission speed requirements.

Business machines with parallel outputs, however, can use either direct parallel data transmission or serial

TABLE B. Standards Organizations for Data Communications

Organization	Data Subdivision	Scope
International Telephone and Telegraph Consultative Committee (CCITT)	Study Group A	Data Communications including standards
Electronic Industries Association (EIA)	Committee TR30	Data transmission electrical standards
American Standards Association (ASA)	Subcommittee X3.3	Data transmission electrical standards
Institute of Electrical and Electronics Engineers (IEEE)	Data communications and telegraph systems committee	Electrical Information exchange

transmission, with the addition of a parallel-to-serial converter at the interface point of the business machine and the serial data transmitter. Similarly, another converter at the receiving terminal must change the serial data back to the parallel format.

Both serial and parallel data transmission systems have inherent advantages which are somewhat different. Parallel transmission requires that parts of the available bandwidth be used as guard bands for separating each of the parallel channels, whereas serial transmission systems can use the entire linear portion of the available band to transmit data. On the other hand, parallel systems are convenient to use because many business machines have parallel inputs and outputs. Though a serial data set has the added converters for parallel interface, the parallel transmitter requires several oscillators and filters to generate the frequencies for multiplexing each of the side-by-side channels and, hence, is more susceptible to frequency error.

Standards

Because of the wide variety of data communications and computer equipment available, industrial standards have been established to provide operating compatibility. These standards have evolved as a result of the coordination between manufacturers of communications equipment and the manufacturers of data processing equipment. Of course, it is to a manufacturer's advantage to provide equipment that is universally acceptable. It is also certainly apparent that without standardization intersystem compatibility would be almost impossible.

Organizations currently involved in uniting the data communications and computer fields are the CCITT, Electronic Industries Association (EIA), American Standards Association (ASA), and IEEE. (See Table B.)

A generally accepted standard issued by the EIA, RS-232-B, defines the characteristics of binary data signals, and provides a standard interface for control signals between data processing terminal equipment and data communications equipment. As more and more data communications systems are developed, and additional ways are found to use them, the importance of standards will become even more significant.

The second part of this two-part article will appear in the October issue of the Demodulator. Subjects to be covered include error detection and correction, transmission methods, and signal impairments.

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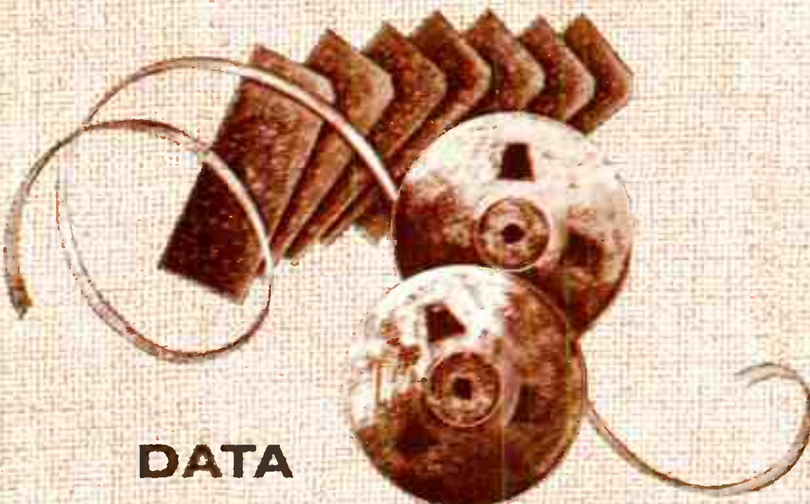


DATA COMMUNICATIONS

Part 2

Today, telephone companies are faced with the challenge of providing reliable data transmission at higher speeds over voice facilities. Because data signals do not possess the built-in redundancy of speech signals, data accuracy is very important. A single error could render a message incorrect. This article, the second of two parts, describes some of the major causes of errors and discusses some of the considerations involved in transmitting data over telephone lines.

ONE of the most important considerations in transmitting data over communications systems is accuracy. Data signals consist of a train of pulses arranged in some sort of code. In a typical binary system, for example, digits 1 and 0 are represented by two different pulse amplitudes. If the amplitude of a pulse changes beyond certain limits during transmission, the detector at the receiving end may produce the wrong digit, thus causing an error.

It is very difficult in most transmission systems to completely avoid such errors. This is especially true when transmitting digital signals over an analog transmission system designed for speech signals. Many of the inherent electrical characteristics of telephone circuits have an adverse effect on digital signals, often making the circuits unsatisfactory for data transmission—especially at high speeds. Frequently, these circuits must be specially treated before they can be used to handle data at speeds above 2000 bits per second.

Voice channels on the switched (dial-up) telephone network exhibit certain characteristics which tend to distort typical data signal waveforms. Since there is random selection of a particular route for the data signal with each dialed connection, transmission parameters will generally change, sometimes upsetting the effect of built-in compensation networks. Often data communications operations require speeds or types of signals which cannot be handled by the switched network. In addition, the switched network cannot be used for large multiple address data systems using time sharing. Because of these considerations, specially treated voice bandwidth circuits are made available for data use (see Table A). The characteristics and costs of these point-to-point *private lines* are published in documents called tariffs, which are merely regulatory agreements reached by the FCC, state public utilities commissions, and operating telephone companies regarding charges for particular types of telephone circuits. The main advantage of private or dedicated facilities is that transmission characteristics are fixed and remain so for all data communications operations.

Signal Impairments

Probably the most critical circuit qualities affecting data transmission are attenuation-frequency response, phase-frequency characteristics, and impulse noise. In addition, echoes and circuit net loss or over-all attenuation tend to degrade data pulses and usually have to be considered when selecting a voice circuit for data transmission.

The attenuation-frequency response limits the bandwidth for transmission. This characteristic causes the transmission loss of a circuit to vary with frequency. Ideally, the attenuation-frequency curve should be "flat" across the band required for data transmission. Most modern voice transmission circuits are sufficiently "flat" between 600 and 2700 hertz so that low-speed data is not seriously affected by this characteristic. However, for high-speed data transmission, it often is necessary to provide some sort of compensation in order to equalize the attenuation-frequency response.

The phase-frequency characteristics of a circuit cause what is known as *delay distortion*. (See *The Lenkurt Demodulator*, June, 1965.) Delay distortion results from the capacitive and inductive reactances common in all communications circuits, causing the propagation rate of signals to vary non-linearly with frequency over the desired bandwidth. This is not a problem in speech transmission because the human ear is not very sensitive to phase-frequency varia-

TABLE A. Specially Treated Voice Bandwidth Circuits for Data Use

	Use	Interstate Tariff FCC No.
Schedule 2 Telephoto	Alternate voice/facsimile (telephoto) or FAX only	140
Schedule 2 Telephoto with Special Conditioning	Alternate voice/facsimile (telephoto) or FAX only	140
Remote Operation & Control (FAA)	Simultaneous voice/remote operation and control or remote operation and control only	135
Schedule 4 Type 4 Data	Alternate voice/data or data only	237
Schedule 4 Type 4A Data	Alternate voice/data or data only	237
Schedule 4 Type 4B Data	Alternate voice/data or data only	237
Schedule 4 Type 4C Data	Alternate voice/data or data only	237
Schedule 5 Data	SAGE data circuits digital data	237

tions. But for data, such variations limit the speed of transmission and reduce the margin for error.

Delay distortion becomes more critical as data speeds increase. Typically, higher data speeds are achieved by shortening the width (duration) of each pulse. Because of the shorter pulse, slight shifts in relative phase between frequency components have a greater effect in distorting the signal.

Some type of compensating network or *delay equalizer* must be employed when the phase-frequency characteristics of a circuit are unsuitable for data transmission (see Figure 1). Such devices introduce a controlled amount of phase shift into the circuit at various frequencies.

The most difficult type of signal impairment to overcome is *impulse noise.* Such noise is extremely unpredictable and is commonly caused by electrical storms and the operation of switching equipment in the telephone plant. Quite often impulse noise will raise or lower the amplitude of a data pulse above or below a fixed detection or slicing level, causing the wrong binary symbol to be indicated at the receiver. One way of overcoming the problem of impulse noise would be to raise the signal power. However, communications systems have limited power handling capabilities which cannot be exceeded.

White noise, on the other hand, has a relatively uniform distribution of energy. Caused by the thermal agitation of electrons in resistances, white noise is always present in electrical circuits and cannot be eliminated. Since this type of noise is predictable, its effect can usually be overcome in the design of a data communications set.

Signal impairment also may be caused by variations in the carrier frequencies between the transmit and receive termi-

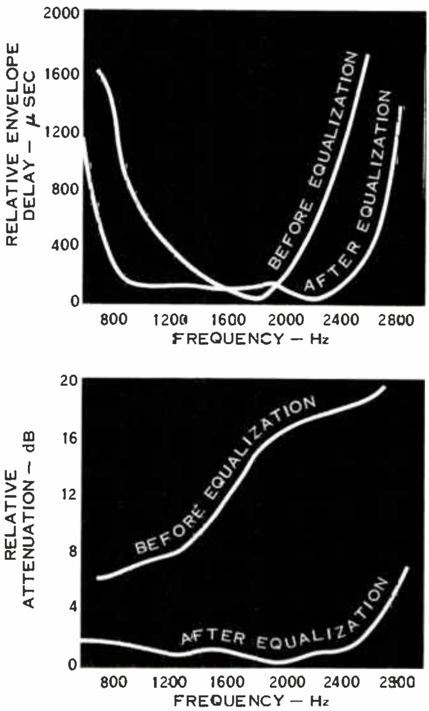


Figure 1. Both delay and amplitude distortion affect the quality of data transmission. Top panel shows the relative envelope delay of a typical voice channel both before and after equalization. The result of amplitude equalization is shown in bottom panel.

nals of a single-sideband suppressed-carrier multiplex communications system. In such systems, carriers removed at the transmit terminal must be reinserted at the receive terminal. Any change in frequency between the transmit and receive carriers will shift the various frequency components of the data signals. If the frequency difference is great enough, the data signals can become so distorted that they cannot be correctly detected. This problem is overcome by synchronizing the carrier generators of the two multiplex terminals, usually with a *pilot* signal which is transmitted through the system. Also, the use of a subcarrier in the modulation scheme of a data set helps to minimize the effect of frequency shift when data signals are fed through multiplex systems.

Error Detection and Correction

As data signals leave the business machine or computer and enter the data transmitter, they are essentially free from error. Since telephone systems were originally designed for voice transmission, various characteristics of the telephone circuit can impair the quality of a data message, possibly changing its context and rendering it unusable. Therefore, from the standpoint of service, the actual error rate or probability of error, which is determined statistically, is the most important factor in evaluating the overall performance of a data set.

Because of the significance of errors in data communications and since it is not practical to build error-free transmission circuits, the usual course is to equip data sets with some type of error control. Typically, error control systems in use today are capable of both error detection and correction, or error detection only.

Techniques which only detect errors are generally less complex than those which detect and correct errors. The simplest and most widely employed method of error detection is simple *parity check coding*. This technique uses redundant bits of information inserted into the digital message so that there is always an odd or even number of mark or space bits transmitted. Parity check coding, though vulnerable to

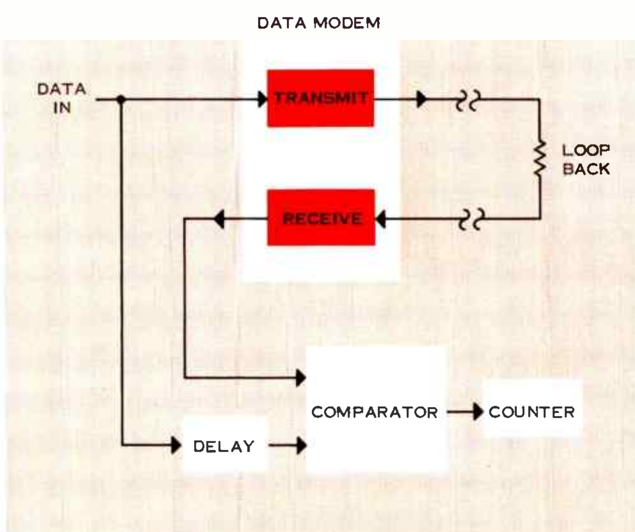


Figure 2. Typical method for measuring error performance of a data communications set. Data signals are looped back from transmitter to receiver. Delay corresponding to the amount in the path is added to the original data signals which are then compared bit-by-bit with the received data signals. An error counter keeps track of the number of errors.

many kinds of multiple errors which overcome the capacity of the coding method, has found wide use because the data receiver can be arranged to check parity without the need for complex circuits.

Error correction techniques use redundancy on a larger scale than the parity check method. The amount of redundancy determines the maximum number of errors that can be corrected. Error correction arrangements are extremely complicated, and the reduction in effective transmission speed necessary to accommodate redundant bits can become excessive. Furthermore, a substantial percentage of errors tend to come in bursts, so the utility of many correction schemes may be somewhat limited.

Data transmission systems using the Lenkurt - developed Duobinary technique (see *The Lenkurt Demodulator*, February, 1963) provide error detection without the need for adding redundant information. Instead of adding redundant bits to the message which, in effect, lowers the transmission rate, Duobinary coding follows a systematic pattern that provides a more powerful error detection method than the simple parity check.

For business purposes, data systems normally transmit information in sections or blocks. When errors in a block are detected, the system automatically keeps retransmitting the block until there are no errors. For real-time systems, such as high-speed telemetering, where data is obsolete almost immediately after it is received, duplicate or diversity transmission techniques are about the only practical solution to offset the problem of errors.

Error rate may be determined by comparing each and every bit of the received message with the transmitted message. This can be accomplished by either looping-back the received signal to the originating transmitter, or by transmitting a predetermined pattern of data signals known to the receiver. When the data signal is looped back (see Figure 2), a variable delay, corresponding to the transmission time of the data message in both directions, is inserted into the original message before it is applied to a bit-by-bit comparator. By

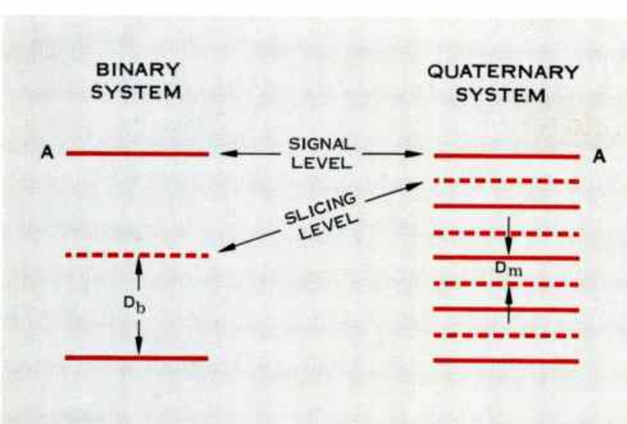


Figure 3. Determining noise penalty compared to binary. Because multilevel systems require more slicing points than binary systems, they are more susceptible to errors caused by noise.

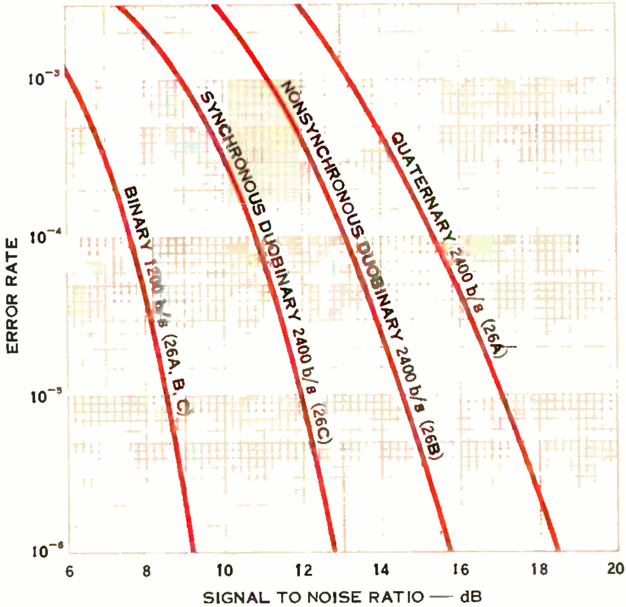


Figure 4. Comparison of error rate versus signal-to-noise ratio. Without normalization synchronous Duobinary has a noise penalty of about 3.4 dB compared to binary. This is a marked increase in performance over nonsynchronous Duobinary and quaternary.

analyzing the transmitted message and the received message together, the comparator signals an electronic counter when errors are present.

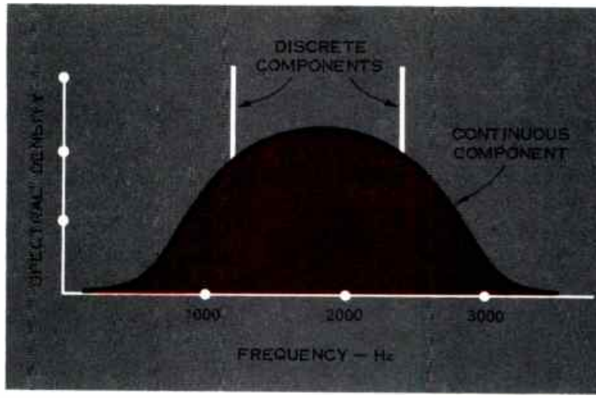
Modulation Methods

Selecting a suitable type of modulation is extremely important in the design of data transmission systems to provide simplicity and to achieve optimum performance. The three basic methods of modulation in data transmission are amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM). Frequency modulation of a binary signal results in the shift of a two-state or binary signal about an FM carrier, and thus is usually referred to as frequency shift keying (FSK) rather than FM.

The most commonly used technique is binary FM (or FSK) because this type of signaling offers a good signal-to-noise ratio, and is not affected by changes in amplitude level. However, the signaling speeds achieved with binary modulation are limited and generally inadequate to meet present-day data requirements.

Vestigial sideband AM and techniques using more than two or *multilevel* signal states increase the amount of data that can be transmitted over communications facilities. Quaternary FSK and quaternary phase shift keying (PSK) are examples of multilevel systems which effectively double the data rate compared to binary that can be transmitted in a given bandwidth. But there is a greater sensitivity to noise with both vestigial sideband and multilevel systems, and error performance is usually poorer. Furthermore, distortion in the absence of noise, termed inter-

Figure 5. Spectral distribution of a binary FM signal. Discrete components present in the binary signal represent wasted power; these are not present in the synchronous Duobinary signal.



symbol interference, is increased with multilevel signals. This type of distortion is caused by the overlap of positive or negative overshoots of the past pulses into the time slots of other pulses. Experimentation on multilevel systems continues and new approaches have been investigated to improve their performance.

Correlative Technique

Correlative data transmission techniques, particularly the Duobinary principle, have aroused considerable interest because of the method of converting a binary signal into three equidistant levels. This correlative scheme is accomplished in such a manner that the predetermined level depends on past signal history, forming the signal so that it never goes from one level extreme to another in one bit interval.

The most significant property of the Duobinary process is that it affords a two-to-one bandwidth compression relative to binary signaling, or equivalently twice the speed capability in bits per second for a fixed bandwidth. The same speed capability for a multilevel code would normally require four levels, each of which would represent two binary digits.

Noise Penalty

Generating a signal with correlated levels permits overall spectrum shaping as well as individual pulse shaping, thus minimizing intersymbol interference. However, there is a noise penalty with respect to binary systems, and this applies to level-coded correlative systems as well as other multilevel systems. Though exact mathematical calculations of such penalties are usually complicated, there is a quick method of finding an approximate value. In Figure 3, both binary and multilevel representations are shown. Assuming an equal peak voltage of *A* volts for both cases, the noise penalty is the ratio of the distances between any signal level and the adjacent slicing level for each of the two systems.

The corresponding distances are $Dm = A/2(m - 1)$, where m is the number of levels

and for binary

$$Db = A/2$$

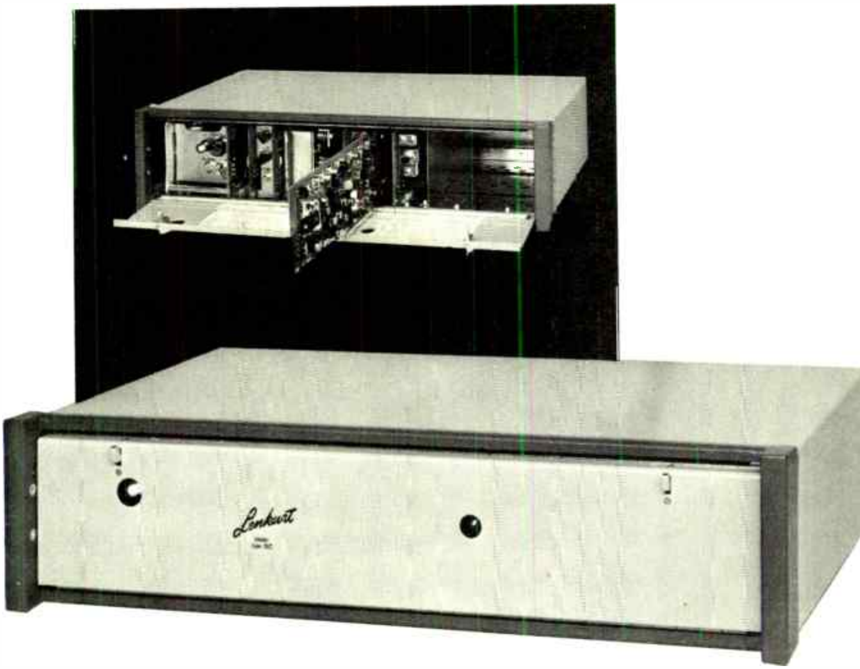


Figure 6. A typical data set shown here is Lenkurt's 26C which transmits serial digital data over standard 3-kHz voice channels at 1200 or 2400 bits per second. Other practical applications include telemetry, digitized voice or facsimile data transmission, and air-to-ground data communications over UHF or VHF radio links. Transmission at 2400 bits per second is achieved through the use of the Lenkurt synchronous Duobinary coding technique. Low error rate coupled with automatic error detection are added features of the Duobinary technique.

The approximate noise penalty (in decibels) for an m-level system relative to a binary system is then

$$20 \log Db/Dm$$

Substituting the values of Db and Dm gives

$$20 \log \frac{A/2}{A/2(m-1)} = 20 \log (m-1)$$

For a binary system, the noise penalty is 0 dB, since binary serves as a reference. The calculation for a quaternary system reveals that the noise penalty is approximately 9.4 dB relative to binary. The Duobinary signal has three levels ($m=3$). Calculating the noise penalty for a Duobinary system from the above formula results in a 6-dB value with respect to binary.

Synchronous Duobinary

A special situation occurs when data transitions are *synchronized* with the carrier phase in FM data transmission at 0°, 180°, or ±90°, as opposed to where the data transitions bear no relationship to the FM carrier phase. It would appear that because there is still the same number of slicing levels, the *synchronized* Duobinary signal has a 6-dB noise penalty. Yet, in practice, it is closer to 3 dB. This situation can be analyzed mathematically, but is more clearly demonstrated by comparative testing of actual working systems under identical conditions.

If error rate is measured as a function of signal-to-noise ratio, individual noise penalties can be determined by subtracting the resultant signal-to-noise ratio of nonsynchronous Duobinary, synchronous Duobinary, and quaternary from the signal-to-noise ratio of a binary system.

It is easy to establish identical conditions to determine the noise penalty by using three systems that have the same peak-to-peak FM deviation, namely, 1200 Hz, which has been proven to be optimum for binary transmission. Three such systems are the Lenkurt 26A, a quaternary system; the Lenkurt 26B, a nonsynchronous Duobinary system; and the Lenkurt 26C, a synchronous Duobinary system.

The conditions for plotting the curves shown in Figure 4 were established by supplying a random binary data signal to each system, introducing white noise with flat weighting over a 3.4-kHz bandwidth, and retiming and sampling the data with a clock derived from data transitions.

For clarity, the signal-to-noise ratio values shown are not *normalized* (put on the same speed basis), since the most significant factor is the performance of Duobinary and quaternary relative to binary rather than the absolute signal-to-noise ratio values. From Figure 4 and Table B, it can be seen that the experimental results and calculated values from the formula *20 log (m - 1)* are relatively similar for quaternary and nonsynchronous Duobinary when compared at an error rate of 10^{-5} (1 error in one hundred thousand bits), with a sufficiently long-time error averaging period.

For synchronous Duobinary, there is only a 3.4-dB noise penalty with respect to binary for random input data. This is because the synchronous system results in a power spectrum that does not have discrete spectral lines as does the nonsynchronous system. (See

TABLE B. Signal-to-noise ratio and noise penalty comparisons with respect to binary of synchronous Duobinary, nonsynchronous Duobinary, and quaternary systems for an error rate of 10^{-5}.

System	S/N Ratio (dB) from Figure 4	Noise Penalty (dB) relative to binary
Binary	8.5	-
Synchronous Duobinary (Lenkurt 26C)	11.9	3.4
Nonsynchronous Duobinary (Lenkurt 26B)	14.5	6.0
Quaternary (Lenkurt 26A)	17.0	8.5

Figure 5.) Discrete components are two steady tones (sinusoids) at frequencies of 1200 and 2400 Hz, respectively, and which appear in binary FM. If the ratio of frequency difference in hertz between mark and space to the bit speed in bits per second (or deviation ratio) is unity, the total power is equally divided by the continuous component and the two discrete components. This phenomenon is inherent to binary FM transmission and there is nothing that can be done about it. Unlike this situation, the synchronous Duobinary signal contains only the continuous information carrying component of spectral density, but no discrete components. Consequently, the total power can be increased by 3 dB compared to nonsynchronous Duobinary FM which contains discrete components that do not carry information.

The Future

It is universally recognized that communications is essential at every level of organization. The United States Government utilizes vast communications networks for voice as well as data transmission. Likewise, businesses need communications to carry on their daily operations.

The communications industry has been hard at work to develop systems that will transmit data economically and reliably over both private-line and dial-up telephone circuits. The most ardent trend in data transmission today is toward higher speeds over voice-grade telephone channels. New transmission and equalization techniques now being investigated will soon permit transmitting digital data over telephone channels at speeds of 4800 bits per second or higher.

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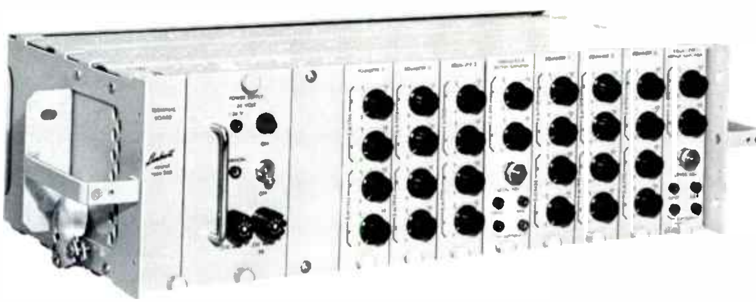
Correction: *In the September, 1966 issue, Figure 1, the lower "Transmit" and "Receive" functions within the blocks marked "Data Modem" should be reversed. Each data modem should have a transmit and a receive section.*

Lenkurt Electric Co., Inc.
San Carlos, California

DELAY EQUALIZER



Lenkurt's 30231 Delay Equalizer corrects for both attenuation and delay distortion on voice frequency circuits used for data transmission, generally providing better equalization than fixed equalizers. Each fully self-contained Delay Equalizer consists of two sets of seven equalizer sections for two separate voice frequency lines. Adjustments are simple and do not require complicated calculations. A typical voice frequency circuit can be equalized to have a relative delay of 50 microseconds and a relative attenuation of ±1 dB between 700 and 2800 Hz.

For more information on this economical equalizer, write Lenkurt, Dept. B720.

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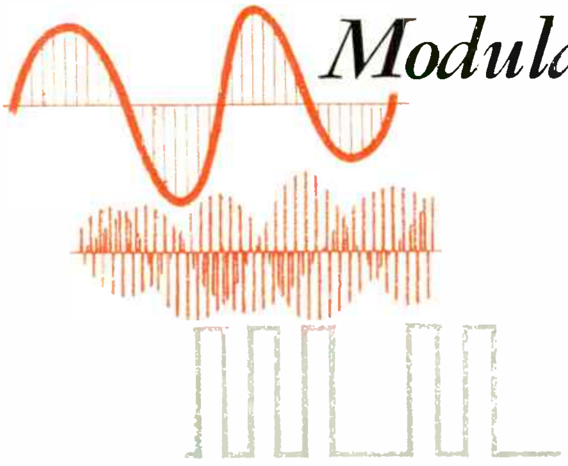
NOVEMBER 1966
VOL. 15, NO. 11

the Lenkurt Demodulator

Pulse Code Modulation



The constant search for better communications at lower cost has led the telephone industry to a radically different method of transmitting speech information. This method, using binary digital pulses rather than conventional analog signals, provides high quality transmission and has proven to be very economical in short-haul carrier systems.

Up to the present time, telephone communications has been accomplished almost entirely with an analog electrical carrier wave which is varied continuously in proportion to the speech signals. Such systems have always been haunted by noise and crosstalk.

In recent years, the telephone industry has shown great interest in a method of coding speech information into digital electrical pulses. Unlike analog signals, these pulses, after becoming distorted by noise during transmission, can be completely regenerated at repeaters along the transmission path and at the receive station.

In the nineteenth century there were many attempts to code speech and music into digital electrical signals for transmission, using the techniques employed in telegraphy. Unfortunately, early experimenters did not have the mathematical tools provided by what is now termed information theory and were denied success because their coding schemes were too simple and did not convey enough information. Before they were able to advance their coding techniques, Alexander Graham Bell successfully transmitted speech using an analog electrical signal. The success of Bell's experiment was so immediately overwhelming that an immense telephone communications industry revolutionized around analog speech transmission.

Because of the outstanding success of analog transmission techniques, such as frequency division multiplexing (FDM), many years passed before serious attention was given to other methods of transmitting speech signals. However, with the ever-present problems of noise and crosstalk and the rising complexity and cost of electrical filters and other devices found in frequency division systems, it was certainly natural for engineers to search for more practical and efficient transmission methods. One of the most significant methods under investigation has been *time division multiplexing*.

It was demonstrated experimentally even before the development of FDM that time division techniques could be used to transmit many speech messages simultaneously over the same circuit. But such techniques could not be put into practical use at the time because of the limitations of mechanical devices for high-speed switching. The invention of the vacuum tube and the electric wave filter made frequency division multiplexing much more attractive for use in telephone transmission systems. However, researchers continued to investigate time division methods.

The first useful time division multiplex systems were developed in the early 1930's. In these systems a number of circuits share a common transmission path but at separate time intervals. Time division systems employ some type of pulse modulation, in contrast to the more familiar amplitude and frequency (AM and FM) techniques used in FDM.

The most popular type of pulse modulation has been pulse amplitude. In pulse amplitude modulation (PAM), a continuous signal, such as speech, is represented by a series of pulses called samples. The amplitude of each sample is directly proportional to the instantaneous amplitude of the continuous signal at the time of sampling. Since the amplitudes of the samples are continuously variable, the problems of cumulative noise and dis-

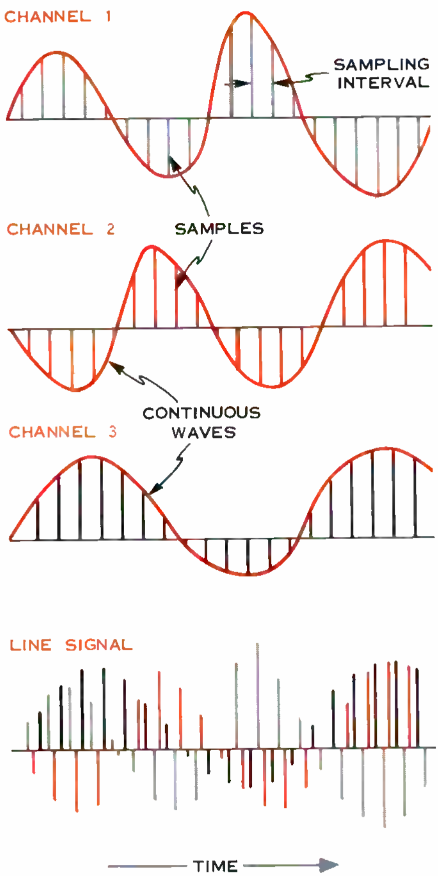


Figure 1. Example of time division multiplexing using pulse amplitude modulation. A pulse amplitude sample is placed on the line as each channel is sampled in turn.

tortion associated with analog signals are present in pulse amplitude modulation systems.

In 1937, Alec H. Reeves, then a member of the International Telephone and Telegraph Corporation Laboratory in Paris, resolved that the problems of cumulative noise and distortion could not be overcome in pulse modulation systems using pulses of varying amplitude. This prompted him to review the early idea of transmitting speech using coded pulses of constant amplitude, similar to those used in telegraphy. His investigation resulted in the invention of a radically different approach to transmitting speech signals. In 1938, Reeves patented his invention which became known as *pulse code modulation*. Unfortunately, the development of practical pulse code modulation systems had to await the arrival of high-speed solid-state switching devices, which occurred after World War II.

Pulse code modulation involves transforming continuously variable speech signals into a series of digitally coded pulses and then reversing the process to recover the original analog signals. This procedure can be carried out in three successive operations.

The first operation is to *sample* the speech signals at a suitable rate and to measure the amplitude of the signal at the time of sampling. This operation is equivalent to pulse amplitude modulation (PAM). Next, the voltage amplitude of each sample, which may assume *any* value within the speech range, is assigned to the nearest value of a set of discrete voltages. This process is known as *quantizing* and is equivalent in mathematics to rounding off to the nearest whole number or integer. The final step is to *code* each discrete amplitude value into binary digital form, similar to coding the letters of the alphabet for telegraphy. Now a series of binary coded digital pulses can be used to carry the message over a transmission line. These binary pulses are in fixed and predetermined time positions and only the presence or absence of a pulse determines the information content of the signal. Since the precise magnitude of the pulses is no longer critical, the problems of cumu-

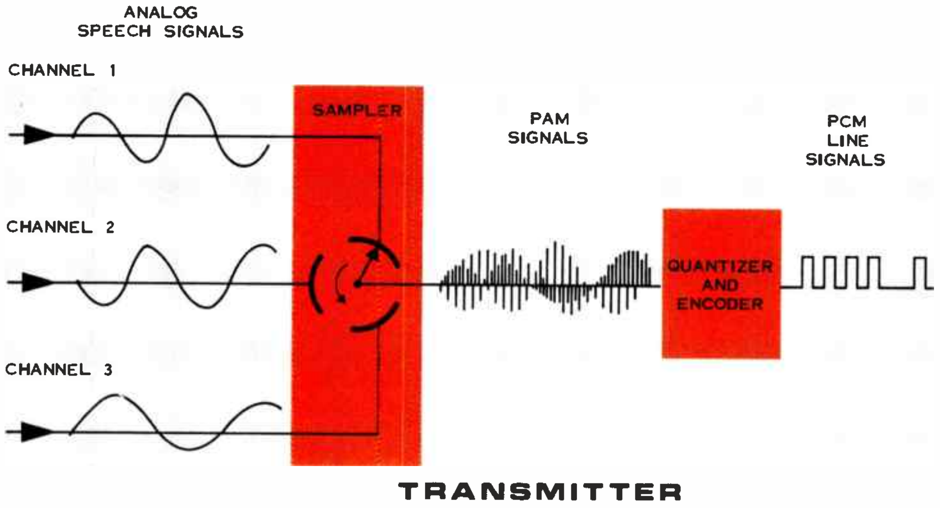


Figure 2. Simplified time division multiplex PCM system.

lative distortion and noise associated with pulses of varying amplitude are greatly reduced.

Sampling

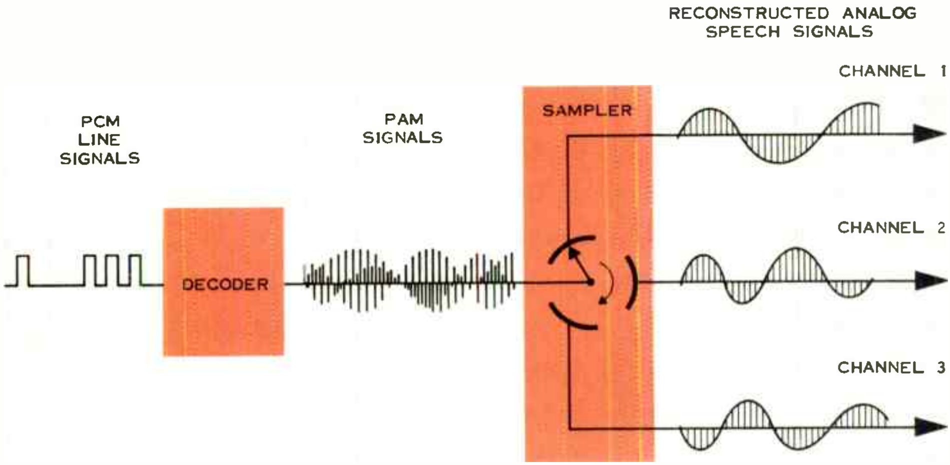
It has been proven mathematically that if a continuous electrical signal is sampled at regular intervals at a rate of at least twice the highest significant signal frequency, then the samples contain all of the information of the original signal. This principle is known as the *sampling theorem*. A continuous signal waveform, therefore, can be represented completely if at least two amplitude samples are transmitted for every cycle of the highest significant signal frequency.

In PCM systems designed for speech signals, a sampling rate of 8000 Hz, or one sample every 125 microseconds (1/8000 second), is ordinarily used. This sampling rate is sufficient since the bandpass of ordinary speech or telephone channels has an upper cutoff frequency below 4000 Hz. The 125 microsecond interval between samples of one voice channel can be allocated to other voice channels by means of time division multiplexing.

The number of channels that can be time division multiplexed using an 8000-Hz sampling rate depends, of course, on the duration of the time slot assigned to each sample — the shorter the duration, the greater the number of channels. In practice, the duration of the samples depends upon the operation and characteristics of a physical circuit. Thus, the number of time division channels is limited by the performance requirements and capabilities of a particular transmission system.

Quantizing

As previously stated, sampling a continuous speech signal at regular intervals results in a series of pulses whose voltage amplitudes are proportional to the level of the signal at the



RECEIVER

time of sampling. The amplitudes might be any of an infinite number of values within the intensity range of speech. The usual intensity range encountered in telephone systems is about 60 dB, or a voltage ratio of 1000 to 1.

After sampling, the next step in the PCM process is to divide or quantize the 60-dB intensity range into increments or amplitude levels to permit binary digital coding. These discrete levels, known as quantum steps, are used to represent any level within the speech range. This is accomplished by using the quantum step nearest to the actual amplitude value of the pulse sample. For example, an actual amplitude sample with a value of say 8.24, would be represented by quantum step 8. A sample value of 8.61 would be represented by quantum step 9, and so on.

Since the quantum step only approximates the actual value, there is always some error. The maximum error is equal to one-half the size of the quantum step. In speech signals, such errors are random and cause what is usually referred to as *quantizing error or noise.* Quantizing noise is the major source of signal distortion in PCM systems. The degree of quantizing noise is mainly a function of the number of quantum steps used—the more quantum steps, the less the quantizing noise. However, increasing the number of quantum steps increases the bandwidth required to transmit the coded signals.

It is, of course, necessary that the quantizing process detect all of the positive and negative amplitude levels within the dynamic speech range. Experiments have shown that approximately 2048 *uniform-size* quantum steps are required to cover the speech range and to provide sufficient signal fidelity. An excessively large bandwidth is required to transmit the coded line signals representing such a large number of uniform quantum steps.

One way of reducing the number of quantum steps without sacrificing quality is to make the size of the quantum steps *non-uniform,* thereby taking advantage of the statistical distribution of speech amplitudes. Most of the information in speech signals is concentrated at low amplitude levels. If the quantum steps are all equal in size, then low level or weak signals suffer the greatest amount of quantizing error. Therefore, small quantum steps are needed more at the low amplitude levels than at the higher levels. If very small quantum steps are assigned where most of the speech information is concentrated, that is at low amplitude levels, and larger steps assigned to the rest of the amplitude range, then the total number of steps required can be greatly reduced. Varying the size of the quantum steps requires sophisticated coding techniques which are presently under development.

Another method is to *compress* the amplitude range of the pulse samples before uniform quantization and then to expand the range back to normal at the receiving end of the circuit. This technique, called instantaneous compression and expansion, or *companding*, achieves the same results as varying the size of the quantum steps. Instantaneous companding, which must be very fast-acting to respond to the short pulse samples, should not be confused with the slower-acting syllabic companding technique used in certain analog telephone circuits — although the principles are the same. The syllabic compandor responds to the envelope of analog speech signals directly while the instantaneous compandor responds to PAM samples of the analog signals.

Signal compression modifies the normal distribution of speech amplitudes by imparting more gain to weak signals than to strong signals. In typical applications, the technique reduces the amplitude ratio from 1000 to 1 to 63 to 1. Using a certain compression characteristic that reduces the speech range from about 60 dB to about 36 dB, and one that varies logarithmically with signal amplitude, the number of quantum steps can be reduced from

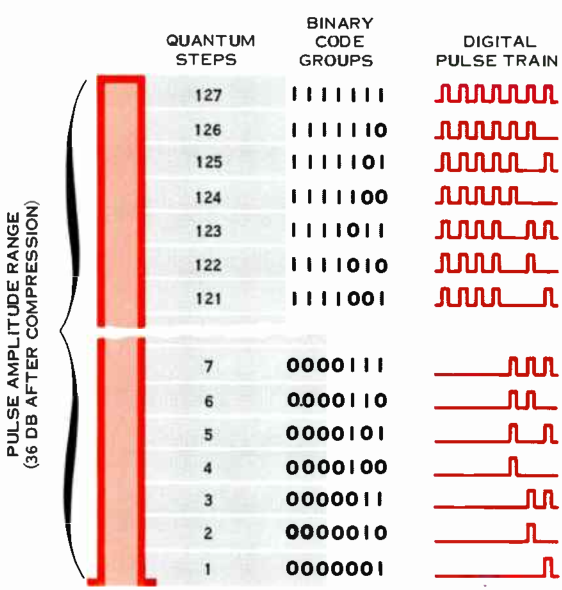


Figure 3. In PCM, amplitude samples of speech signals are compressed, quantized, coded into binary form, and placed on the line as digital pulses.

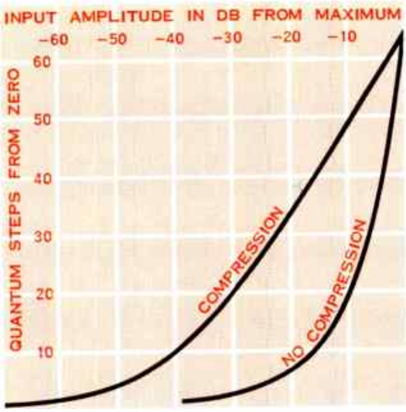


Figure 4. Signal compression using typical logarithmic compression characteristic.

2048 to 128 while maintaining the same quantizing noise performance. Signal compression using a typical logarithmic compression characteristic is shown in Figure 4.

Coding

The final step in the PCM process is to code the quantum steps into digital form. If each quantum step is numbered in decimal form, then some type of digital code can be developed to represent each of the numbers. Ordinarily, a binary code is used that consists of a combination or *code group* of binary l's and O's, each group representing a decimal number. Once the code is established, a series of on-off binary pulses, representing the code groups, can be used for transmission.

The number of quantum steps that can be represented with a binary code is 2^n, where n is the number of binary digits, or bits, required in each code group. Thus, a 5-bit code is required for 32 (or 2^5) quantum steps, while a 7-bit code is needed for 128 (2^7) steps. In systems using a 7-bit code, the speech amplitude range is normally divided into 127 quantum steps; step 64 is zero reference, with 63 steps positive and 63 steps negative.

The bandwidth required to transmit digital pulses is directly proportional to the number of bits in the code group. A code representing 2048 uniform quantum steps would have required 11 bits per code group ($2^{11} = 2048$). Compressing the amplitude range of the sample pulses before quantization, therefore, reduces the number of bits per code group required for quality speech transmission from 11 to 7.

With a 7-bit code, the first bit position has a value of $2^6 = 64$, the second has a value of $2^5 = 32$, and so on. The value of all seven positions is shown in the following table.

Bit Position	1	2	3	4	5	6	7
Value	2^6	2^5	2^4	2^3	2^2	2^1	2^0
	64	32	16	8	4	2	1

Typically, the coded line signal consists of a train of pulses in which binary l's are represented by positive or negative pulses and binary O's are represented by spaces (or no-pulses). A binary l in any of the bit positions means that the value of the position is to be summed. A binary O in any of the positions means that the value of the position is *not* to be summed. As an example, the pulse train and code group representing quantum decimal step number 100 would be:

Bit Position	1	2	3	4	5	6	7
Pulse Train							
Binary Code	l	l	O	O	l	O	O
Value 100	64 +	32 +	0 +	0 +	4 +	0 +	0

Present Applications

One of the most outstanding features of PCM systems is that the coded line pulses can be regenerated at repeater stations. Since only the presence or absence of a pulse determines the message, the line signal can be completely renewed each time it passes through a repeater. This allows a high signal-to-noise ratio to be maintained through a long string of repeaters, thus overcoming most of the problems of cumulative noise which characterize analog transmission systems.

Unfortunately, the advantages of PCM are obtained at the expense of increased bandwidth. For example, the bandwidth of a voice channel in a PCM system using an 8000-hertz sample rate and a 7-bit code would be approximately 56 kHz compared to 4 kHz required for a single-sideband suppressed-carrier FDM system.

In typical long-haul high density transmission systems, especially microwave radio systems, the availability of bandwidth is usually very critical. Presently, PCM does not provide sufficient economical or technical improvements over analog techniques to justify its use in these long-haul systems. But the same is *not* true in short-haul cable transmission systems. There have been continuing efforts by the telephone industry to shorten the economical *prove-in* distance of multichannel carrier systems since they were introduced into the short-haul cable plant. The tremendous population growth around urban areas has greatly increased the need for low cost carrier systems in short-haul interoffice trunks. This need has stimulated

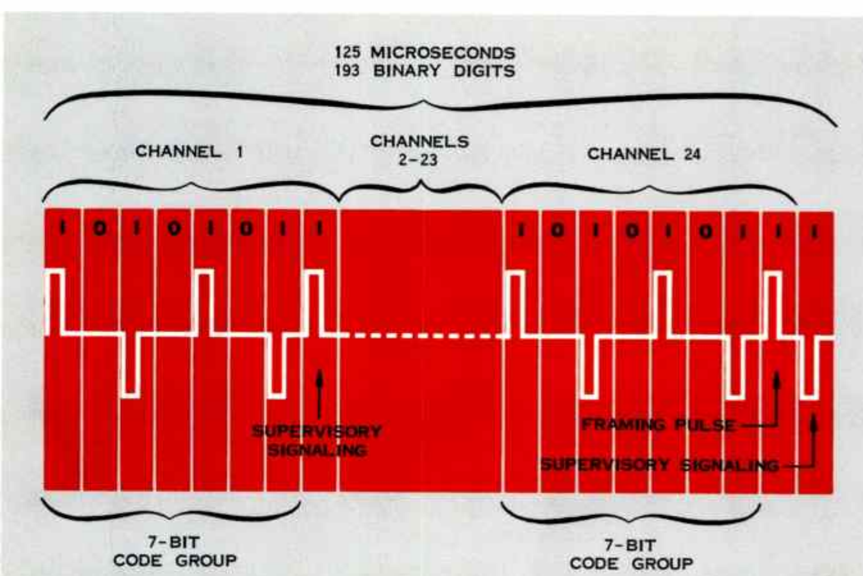


Figure 5. In the 24-channel T1 carrier system, 125 microsecond sampling interval or frame is divided into 193 time slots — 168 slots for speech, 24 slots for supervisory signaling, and 1 slot for synchronization.

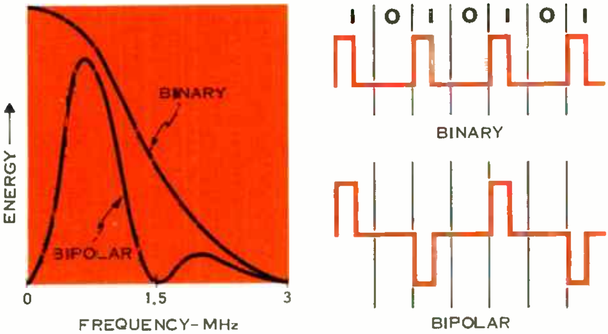


Figure 6. Energy distribution for binary and bipolar pulses with 50-percent duty cycle. In bipolar pulses most of the energy is concentrated at half the pulse repetition frequency and there is no dc component.

interest in PCM systems for use in telephone exchange cable trunks ranging in lengths from about 6 to 50 miles.

In 1962, the Bell System began production of a 24-channel PCM carrier system called the T1. Installation of T1 carrier systems in Bell's exchange plant marked the first large-scale use of time division multiplexing in commercial telephony.

The T1 carrier system was designed primarily for use with two non-loaded 22-gauge cable pairs in exchange area trunks. One cable pair is required for each direction of transmission. Regenerative repeaters, used with the T1 system, are spaced at intervals of about 6000 feet. This interval corresponds to the spacing of Western Electric's H-88 load coils on 22-gauge cable pairs. Since the load coils must be removed when the line is to be used for PCM operation, it is convenient to replace them with a regenerative repeater.

Each voice-frequency input channel in the T1 is sampled once every 125 microseconds or 8000 times per second. The variable amplitude pulses resulting from the sampling process are then compressed and quantized into one of 127 quantum steps coded into 7-digit binary code groups. An eighth digit or bit is added to the code group for each channel sample and is used to carry supervisory signaling information.

The time slots which make up one 125-microsecond period constitute what is called a *frame*. An additional bit time slot is added to each frame for use in synchronizing the two system terminals. This makes a total of 193 time slots per frame (24 channels x 8 bits per code group + 1 synchronizing slot). Multiplying the 193 time slots times the 8000 hertz sampling rate provides an output pulse train with a maximum bit rate of 1,544,000 bits per second.

The binary coded pulses transmitted to the cable pair have a fifty percent duty cycle, which means the width of the pulses is one-half the time slot allocated to each pulse. Bipolar transmission is used with successive pulses, representing binary 1's, alternating in polarity. Figure 5 illustrates a pulse train representing one frame.

There are several advantages of the bipolar pulse pattern over straight binary or unipolar transmission. As shown in Figure 6, most of the energy of bipolar signals is concentrated at frequencies of about half the pulse repetition frequency. Accordingly, there is much less energy coupled into other systems in the same transmission cable because of increased crosstalk coupling

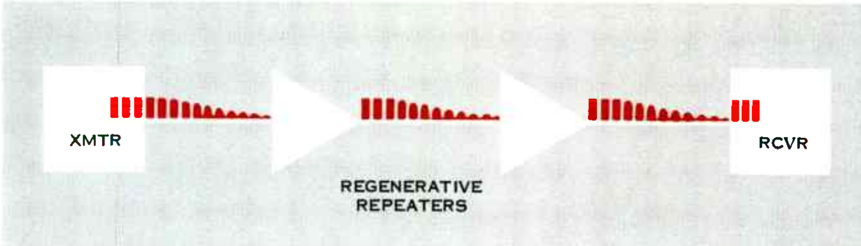


Figure 7. One of the most outstanding features of PCM systems is that the binary coded digital line pulses can be regenerated at repeaters and at the receiver station.

loss. Also, bipolar pulses do not have a dc component, thus permitting simple transformer coupling at repeaters. The unique alternating pulse pattern can also be used for error detection since errors tend to violate the pattern.

As the line signals travel along the cable pairs the pulses become distorted by the usual signal impairments such as noise and attenuation. When the pulses reach a repeater, they are retimed and reshaped so that a new undistorted pulse is produced for each pulse received. At the receive terminal the line pulses are again reconstructed before they are fed into the receiver detection and decoding equipment. The PCM coding process is reversed in the receiver in order to recover the original continuous speech signal. The continuous signal at the output of the PCM receiver should be a replica of the original signal, except for some distortion resulting from quantization.

However, noise troubles, like energy, seem to be conserved and only changed from one form to another. So it is with pulse code modulation. Although noise in PCM systems does not accumulate, it does prevent the perfect timing of regenerated pulses and shows up as jitter on the retransmitted pulse train. Successful practical solutions to this timing problem are the key to successful PCM cable carrier transmission.

Conclusions

New digital technology promises much more than just carrying out the tasks of transmission systems developed in the past. PCM systems will eventually handle all of the transmission functions of today's frequency division multiplex systems more efficiently and more economically. The development of digital techniques will enable different types of services such as voice and facsimile to be treated alike in transmission systems. Once the various types of analog signals are formed into digital signals they are all similar.

The new PCM cable carrier systems must carry on the tradition of the telephone industry. They must be used with the telephone plant that exists today, complete with its inheritance of old cables and switching systems produced by many different manufacturers.

In addition to operating over existing cable systems, there are already means to interconnect FDM systems with PCM networks using a device called an en*co*der-*dec*oder or codec. Network television in color will also be handled. Such high density systems will require line transmission rates close to 300 megabits per second. Also, the use of a technique called *pulse stuffing synchronization* will permit adding and dropping systems by digital means, and provide an easy method of interconnecting PCM systems.

High density PCM systems will use thousands of transistors in circuit configurations where switching times may be as fast as a fraction of a nanosecond. In the future, the use of integrated circuits instead of discrete components promises great economies as well as excellent reliability.

PCM systems are not without problems. Long chains of repeaters in tandem challenge the ingenuity of engineers to produce reliable repeaters at economical prices. The T1 system, for example, may use 50 repeaters in tandem for a 50 mile link. Further problems arise because PCM requires so much bandwidth. Bandwidth is readily available on cable but is not easily obtained from the available microwave spectrum. This fact ensures that single sideband frequency division multiplex will be around for a long time. PCM systems are also vulnerable to impulse noise which may prohibit their use in situations where cable plant and switching machines are not up to modern standards. In such situations present day FDM cable carrier systems, which do not exhibit the noise threshold characteristics of PCM, may do a better job. Also, cables already carrying FDM carrier systems will have to be filled out with the same type of systems since it is presently not possible to mix T1 and FDM systems in the same cable.

Although the T1 carrier system was developed primarily for the transmission of analog information in the form of processed voice signals, its repeatered line is a very fine high-speed digital transmission facility. Techniques have been developed to use these digital transmission systems to handle up to eight 50-kilobit data channels or two 250-kilobit data channels.

Pulse code modulation systems provide better handling of telephone supervisory signaling than the usual in-band methods used with FDM systems. The systems employ time division signaling methods which are very economical and avoid the problems of speech simulation or *talkdown* inherent in in-band signaling systems.

Some small switching machines employ time separation instead of the familiar space separation techniques used for so many years by electromechanical machines. Digital transmission is used with these time separation switching machines, and it is only a short technical step to join digital transmission and switching into an integrated communications system. It seems likely that in the 1970's integrated electronic switching and PCM transmission systems will be operating both in the United States and Europe.

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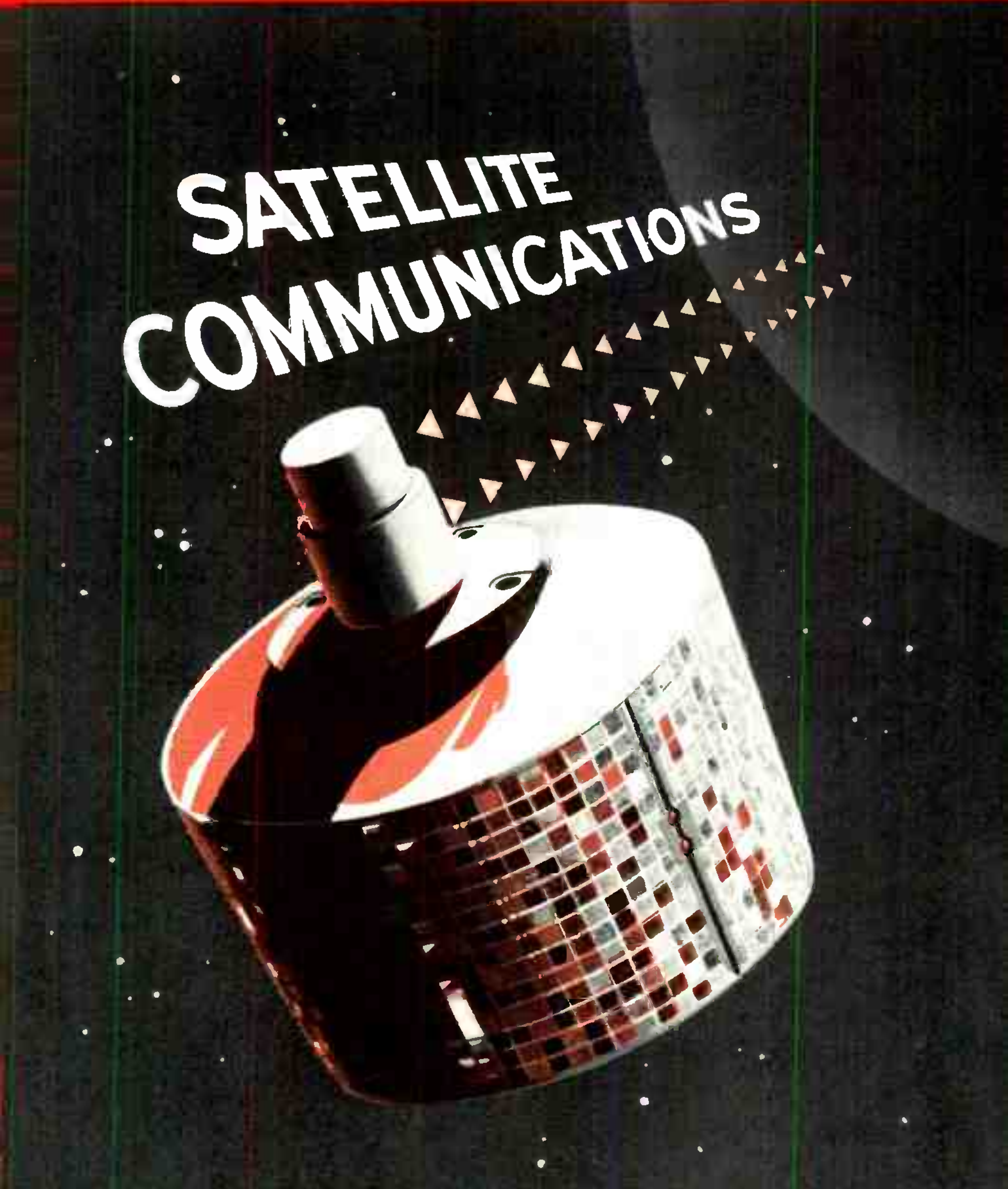

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The first commercial use of man's ability to travel in space — not yet a decade old — has come in the field of communications. Soon, over 90 percent of the world's telephone facilities may be joined in a global network through communications satellites.

Long before communications-relaying space vehicles became a reality, scientists eyed the natural satellite of the earth—the moon. Early in 1946 Project Diana bounced the first radar signals off the moon. Twenty years later artificial satellites circle the globe, but the moon has not been forgotten. A ship-to-shore communications link using the moon as a reflector is expected to be operational sometime in 1967.

The first man-produced "beeps" directly from space were heard on October 4, 1957 from the Russian Sputnik I. The United States entered the space age on January 31, 1958 with Explorer I. Today, in addition to the spectacular manned exploration of space, commercial satellites stretch telephone circuits across the oceans, and telecasts from other continents are common.

American scientists placed the first communications satellite in orbit a year after Explorer I. Score—a short lived but highly successful "bird"—relayed messages up to 3000 miles and broadcast to the world a tape recorded Christmas greeting from President Eisenhower.

The 1960 flight of Echo I was witnessed around the world as the 100-foot balloon-like reflector satellite provided a "radio mirror" for powerful ground stations. Echo II went up in 1964 as experiments with passive reflectors continued.

An active repeater, Courier, extended the knowledge of space communications in 1960 with successful transmission of of high speed teletype, voice, and facsimile.

The commercial value of communications satellites was accentuated in 1962 with Telstar I, the joint project of NASA and AT&T. The first live telecasts between Europe and the United States added to the satellite's performance in transmitting high-quality voice, data, teleprint and other signals. Telstar II, and NASA's Relay I and II added more data.

By mid-1963, the first of three Syncom satellites was launched and communications milestones began to pile up. Syncom III brought the Tokyo Olympic games to the United States, and went on to demonstrate its value for all types of telecommunications. NASA has since concluded its planned tests, and both Syncom II and III are now working for the Defense Department, parked over the Pacific and Indian Oceans.

How High?

Most alternatives faced in the design of communications satellites are centered around the choice of orbits. Orbital mechanics govern precisely the height and period of a satellite. A satellite circles the earth in a period directly related to the satellite's altitude. (The mass of the satellite is negligible and can be ignored in most calculations.) A satellite 100 miles high circles the earth in about 87 minutes; at 1000 miles the period is 118 minutes. As the altitude increases, the orbital period becomes longer, until at an altitude of 22,300 miles a satellite orbits the earth in exactly the same time as one rotation of the earth—that is, every 24 hours. Placed in an easterly orbit over the equator, such a "synchronous" satellite appears stationary in the sky.

The first communications satellites—the Telstars and Relays—were in nonsynchronous orbits. The new breed, lead by Syncom and Early Bird, are synchronous and remain in precisely fixed positions. But depending on the application, each plan has advantages and disadvantages.

The lower the satellite, the shorter its period, and likewise the less time it will be in the simultaneous view of any two ground stations. For example, a 3000-mile-high satellite can be tracked for only 24 minutes by stations located 3000-miles apart—*if* the satellite passes directly over both stations.

Military Plan

A low-flying random orbit is especially appealing to the military, interested in the security of its communications system. The quasi- or nonsynchronous satellite does not require orbit-control commands from the ground, and therefore cannot be tampered with by an enemy. In the Initial Defense Communications Satellite Program (IDCSP), with up to 24 satellites placed in an 18,000-mile orbit, if a satellite fails for any reason another will soon move into view. The satellites drift around the earth at about 30° per day—any single satellite is in view for over four days at a time.

Commercial Advantage

Commercial systems planned through 1968 will be synchronous; the reasons are mostly economic. Fixed position synchronous satellites greatly simplify tracking, thus reducing the cost of ground stations. In developing a truly worldwide system, where each additional ground station may open communications to an entire region or country, the installation cost of these stations becomes increasingly important.

Tracking becomes a relatively simple function of the ground station of a synchronous system. As satellites are pushed by "solar wind" and pulled by gravity from the earth, moon, and sun, periodic adjustments in position are made by onboard thrusters. Between correction intervals typical 85-foot parabolic ground antennas track minor variations—measured in hundredths of a degree.

The synchronous satellite, at 22,300 miles above the earth, is visible to almost half of the globe at one time. Three satellites, spaced equally around the earth, would provide contact with any country served by an adequate ground station (Figure 1). An exception exists at the poles where signal strength is at a minimum. In practice, more than three satellites undoubtedly will be used to increase the number of circuits available in high density areas, and to ensure greater flexibility.

Long transmission delay time is the one serious disadvantage at synchronous altitudes. A one-way telephone path through such a satellite is about 50,000 miles—a delay of 265 milliseconds. Since delays of over 400 ms are unacceptable for voice communications, circuits probably will be limited to only one satellite hop in spanning the globe. The problem, however, does not concern the transmission of television or data.

The products of new technology are easily phased in with a synchronous system. Three or four satellites can replace an entire synchronous system, and older low-capacity units can be moved to areas where traffic is lighter.

The ability to relocate synchronous satellites is a needed feature should a failure occur. An extra satellite, parked in a low traffic area, could be moved in as a replacement in much less time than it would take to prepare and launch a new vehicle. Using onboard thrusters, a satellite can move about 10° per day — from a station over the Atlantic to the Pacific, for example, in about 15 days.

Intelsat

Global communications is being established through the International Telecommunications Satellite Consortium (Intelsat), made up of 54 participating countries. Congress has franchised the Communications Satellite Corporation (Comsat) to establish service for American common carriers, and to represent this country in dealings with Intelsat. Comsat is the major shareholder in Intelsat and serves as its manager.

Any country can join Intelsat, agreeing to share the financing of satellites and tracking equipment. Each country has the responsibility for its own ground stations, with at least 25 countries expected to have working stations by 1971.

The first commercial communications satellite, popularly known as Early Bird, began operation over the Atlantic in mid-1965. Two advanced Intelsat 2 satellites (Blue Birds) will establish the first service over the Pacific and add to the channel capacity over the Atlantic. The third generation Intelsat 3 satellites are scheduled for launch in early 1968 to further expand the worldwide system.

Figure 1. Three satellites in synchronous orbit could cover the entire earth.

Power Source

A sizeable tradeoff between rocket booster power and payload weight must be made in orbiting any object. Since the communications satellite must carry its own power source into space, energy for all electronics is necessarily limited. In turn, the less radiated power from the satellite's transmitter, the lower the signal-to-noise ratio and the fewer channels that can be relayed.

The solar cell remains the most practical power source in space, delivering about 6 watts per pound. Even at that, Intelsat 2 must make do with about 85 watts. Experts claim that up to 800 watts is possible using only skin-mounted solar cells, and deployable arrays might boost available power into the kilowatt area. But such arrays, like nuclear energy power for spacecraft, must remain in future plans.

Today's problem is doing the most effective job with the equipment available. With power output confined, attention has naturally turned to spacecraft antenna design—itself restricted by other physical considerations.

Satellites are prevented from tumbling uncontrollably through space by giving them a bullet-like spin of about 150 rpm. Spin stabilization eliminates some problems, but creates others, especially for antenna designers. With the satellite spinning, it is impossible to use a conventional directional antenna.

Present communications satellite antennas produce a toroidal or "doughnut shaped" pattern with about 9-dB gain (Figure 2). Better than a omnidirectional antenna, the method nevertheless "loses" considerable energy in the portion of the pattern not touching the earth.

Despun Antennas

Future spin-stabilized satellites will be equipped with devices for focusing this otherwise lost energy, increasing

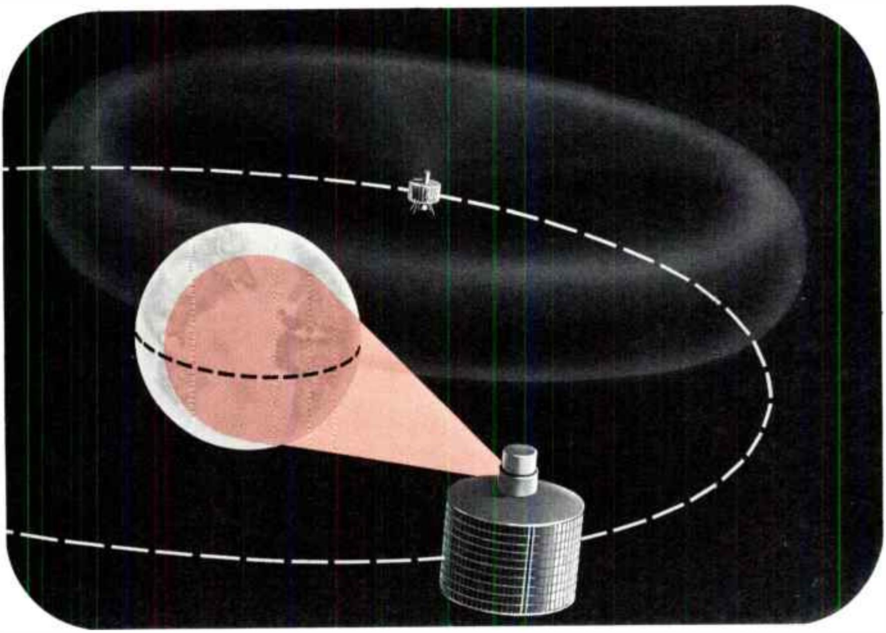

Figure 2. Toroidal pattern of first communications satellites (rear) loses much rf energy to space. Intelsat 3 will focus its communications beam toward earth, using new despun antennas, with up to 16 dB gain.

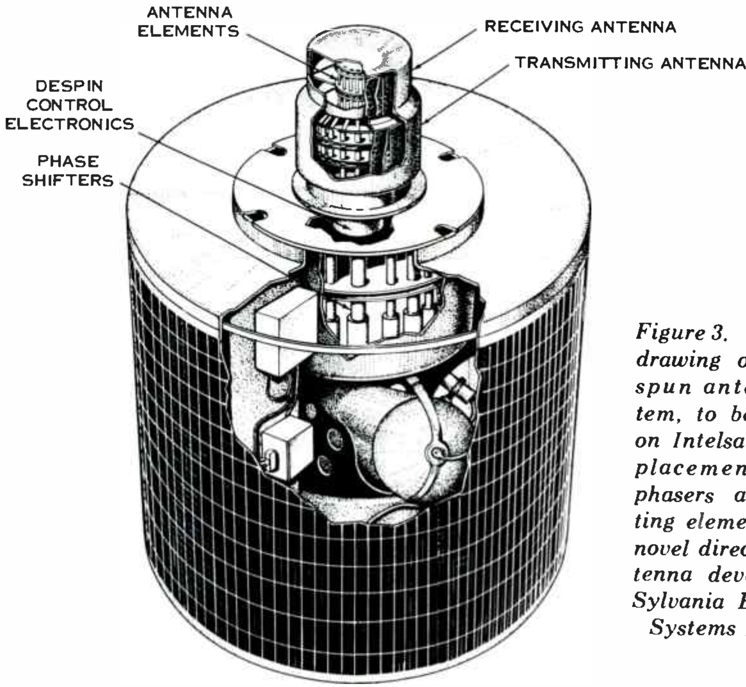

Figure 3. A cutaway drawing of the despun antenna system, to be initiated on Intelsat 3, shows placement of the phasers and radiating elements of the novel directional antenna developed by Sylvania Electronics Systems Division.

the gain considerably. Expected minimum gain will be 13 dB, with peak gain about 16 dB. The trick is to rotate the antenna in the opposite direction to the satellite, thereby keeping the "despun" beam pointed to earth.

Three types of despun antennas are possible: mechanical, electronically switched, and electronic. The mechanical method uses a directional antenna that is physically counter-rotated about the axis of the satellite. The greatest danger is mechanical failure. The electronically switched method systematically shifts rf power from antenna to antenna, keeping overlapping beams in the desired direction. The pure electronic approach—the one selected for the Intelsat 3 satellites—steers the beam by varying the phase of the signal as it feeds a series of radiating elements (Figure 3).

The electronically despun antenna has three major subsystems: an earth center reference system, control circuits, and the radiating assembly. Two redundant horizon sensors scan the earth as the satellite rotates. Control circuits regulate the action of phase shifters, which direct the rf energy to the radiating elements of the antenna. The result is a radio beam continually focused on the earth.

At synchronous altitudes, the earth's disk is just over 17° across. Allowing for satellite stabilization and antenna tracking errors, a beam approximately 19° x 19° would adequately cover most points on the globe. For specific purposes, the beam could be made more directional, thereby increasing the gain. For example, a satellite designed to relay traffic only from the United States to Europe might have a fan-shaped beam 19° x 10° (long to the east-west).

Minimum gain would be increased to 16 dB, with peak gain at 19 dB.

An alternative to spin stabilization is being tested, requiring no onboard thrusters or other control devices. Known as *gravity-gradient* stabilization, the method would maintain the same side of the satellite always facing the earth. A long object in space will tend to align itself vertically with the strongest source of gravity—in this case, the earth. Extendible arms could, in effect, make the satellite such a "long" object. Highly directional antennas could then be accurately pointed earthward.

Early Bird

Important advances are continually incorporated in new satellite designs. Our first communications satellite now seems small compared to vehicles being developed. Early Bird weighs 85 pounds, is 28 inches in diameter, has a solar power capacity of about 46 watts, and can relay 240 two-way voice channels, or two-way television. The communications system has two transponders (receiver-transmitters), one for each direction of traffic. The transmitter output comes from one 6-watt traveling wave tube; a second TWT is carried for redundancy. The bandwidth of the transponder is 25 MHz. Receiver frequencies are in the 6-GHz band, with transmission back to earth in the 4-GHz range. Telemetry and control signals to and from the satellite are at VHF frequencies in the 136 MHz range.

Intelsat 2

The latest addition to the global system, Intelsat 2, has five times the bandwidth (125 MHz) of the Early Bird, and three times the output power (18 watts). Increased power provides greater geographical coverage, while the wideband capability allows multiple access for the first time. Now a number of different ground stations can channel through the satellites at the same time. Capacity remains at 240 high-grade voice channels.

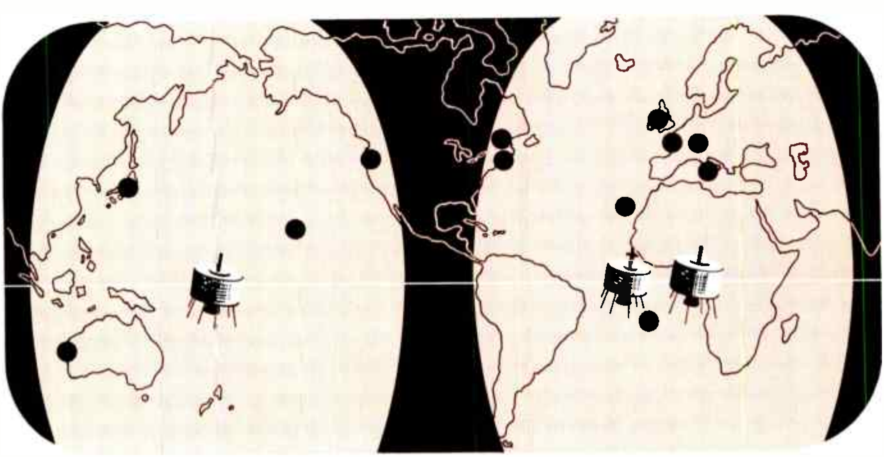

Figure 4. Ground stations on both sides of the Atlantic and Pacific will be connected through Early Bird and Intelsat 2 satellites.

So called "quasi-linear" transponders are a unique part of the multiple access design; transmitter output increases linearly with an increase in input power. There is no radiated power from the satellite until a signal is received from earth. If several signals are received simultaneously, transmitter power is divided among them proportionally according to the power of the received signal. The quasi-linear method reduces intermodulation products and crosstalk inherent in the Early Bird fixed-output transponder.

Four 6-watt TWT's are carried in the spacecraft. Three of them normally will work in parallel; the fourth is a spare. The Intelsat 2 satellite has an orbital weight of 165 pounds, is 56 inches in diameter, and produces 85 watts of power from solar cells. Communication and telemetry frequencies are virtually the same as in Early Bird.

Electronics

Within the Intelsat 2 transponder, the incoming 6-GHz signal passes through a low-noise tunnel diode rf amplifier to a directional coupler where command signals are extracted (see Figure 5). The communications signal is converted directly from 6 GHz to 4 GHz in a mixer section, then delivered to a driver TWT. The driver tube operates only on command, providing ground control selection of two redundant receivers. A beacon frequency (for tracking) is then added before the signal is supplied to the four output TWT's.

The satellite is capable of receiving a signal at any frequency between 6283

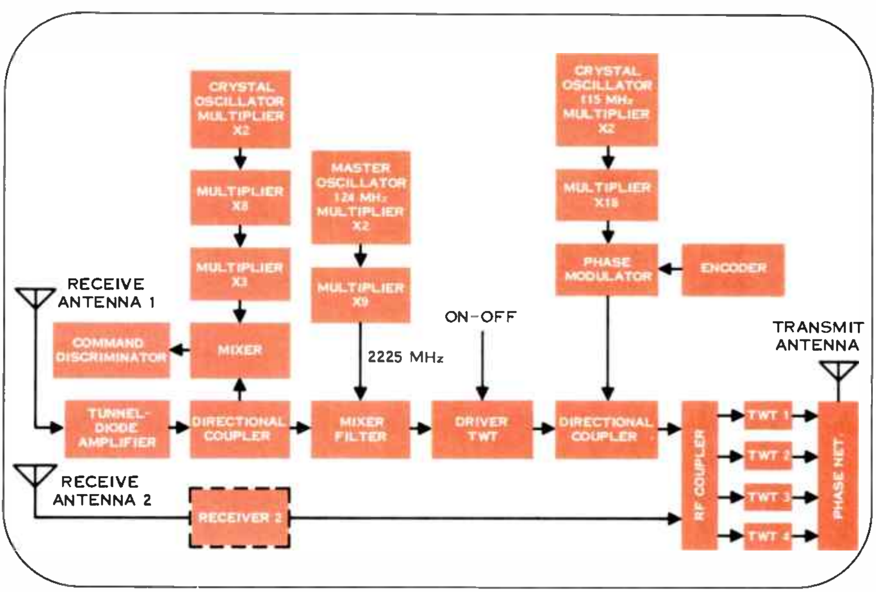

Figure 5. Communications system block diagram for Intelsat 2, built by Hughes Aircraft Co.

Figure 6. Seen inside Intelsat 3, to be launched in 1968, are many of the electronic and propulsion systems included in the third generation communications satellite being built by TRW Systems.

MHz and 6409 MHz. The incoming signal is translated at the mixer by 2225 MHz to the transmit band of 4058 MHz to 4184 MHz.

Telemetry and control signals are available on both VHF (136 MHz) and through the modulated beacon carried with the communications channels.

Together with adding satellite communications capability over the Pacific and increasing service in the Atlantic, Intelsat 2 will play a vital role in the Apollo space program. NASA will use a number of circuits for astronaut voice relay, spacecraft television, high-speed tracking data, and telemetry. Transmission path will be from the Apollo spacecraft to NASA surface stations (including special tracking ships at sea), and then via the communications satellites to mainland ground stations.

Intelsat 3

Even higher capability is being designed into the Intelsat 3 satellites, to be launched in 1968. Measuring 56 inches in diameter, weighing 250 pounds, and with solar power of 160 watts, the communications package will be able to handle at least 1200 two-way voice channels or four television channels (See Figure 6)

Each of the two transponders aboard Intelsat 3 will have a bandwidth of 225 MHz, with high-level 10-watt TWT output stages. Incorporating electronically despun antennas, the satellites will have an effective radiated power of about 22 dBw (decibels above one watt), compared to about 15 dBw for Early Bird. Like its predecessors, Intelsat 3 will use rf amplification, with translation from 6 GHz to 4 GHz.

Table A. Frequency allocations agreed on at the 1963 Geneva Extraordinary Administrative Radio Conference. Commercial bands, shared with terrestrial systems, are shaded: others are for special applications (including military).

Frequency Bands	Service	Frequency Bands	Service
1700-1710 MHz	Space Research (Telemetering & tracking) (shared)	5725-5850 MHz	Communication-Satellites (Earth-to-satellite) (shared)
1770-1790 MHz	Meteorological-Satellites (shared)	5850-5925 MHz	Communication-Satellites (Earth-to-satellite) (shared)
2290-2300 MHz	Space Research (Telemetering & tracking in deep space) (shared)	5925-6425 MHz	Communication-Satellites (Earth-to-satellite) (shared)
2690-2700 MHz	Radio Astronomy (exclusive)	7250-7300 MHz	Communication-Satellites (Satellite-to-Earth) (exclusive)
3400-4200 MHz	Communication-Satellites (Satellite-to-Earth) (shared)	7300-7750 MHz	Communication-Satellites (shared)
4400-4700 MHz	Communication-Satellites (Satellite-to-Earth) (shared)	7900-7975 MHz	Communication-Satellites (Earth-to satellite) (shared)
4990-5000 MHz	Radio Astronomy (shared in some areas)	7975-8025 MHz	Communication-Satellites (Earth-to-satellite) (exclusive)
5250-5255 MHz	Space Research (shared)	8025-8400 MHz	Communication-Satellites (Earth-to-satellite) (shared)
5670-5725 MHz	Space Research (Deep space) (shared)	8400-8500 MHz	Space Research (shared) (exclusive in some areas)

Life expectancy of synchronous satellites is about five years, governed by the onboard fuel supply for positioning thrusters. When the fuel is expended, the satellite will begin to drift slowly westward. Its communications capability, however, could continue for some years. The life of the electronics is primarily dependent on the source of electrical power—solar cells. These cells deteriorate with exposure to radiation, a common hazard in space, especially near the Van Allen radiation belts. Intelsat 3, for example, will begin its service with 161 watts of available power. After five years only about 105 watts can be expected from the solar cells. But this is still enough to support at least limited communications.

Modulation

The satellite microwave repeater has much more bandwidth than its terrestrial cousin. Bandwidth in satellites is needed not only to increase channel capacity, but to allow multiple access from many ground stations. Each ground station will use a discrete carrier frequency, with a number of multiplexed channels. Frequency division multiplex with frequency modulation (FDM/FM) is used in current systems, but other modulation techniques are possible. Time division multiplex (TDM) tests have been completed with the Early Bird satellite. Pulse code modulation (PCM) was used successfully to carry voice and data signals between two North American terminals.

Russian Satellites

Even though Intelsat countries have 90 percent of the international communications potential, considerable work is being done by nonmember countries—especially the Soviet Union. Russia has lofted several Molniya-class communications satellites in 12-hour elliptical orbits. Successful transmissions of many types of signals, including color television, have been carried out in joint experiments with the French. While the Russian satellites apparently lack channel capacity, they do boast high-powered transmitters and other "weighty" equipment. The Molniya has a command receiver, 40-watt transmitter, two reserve transmitters, and two steerable parabolic antennas. In addition, the satellite has orbital adjustment and three-axis attitude control capabilities. Apparently with room to spare, the Russians are also including meteorological equipment on board, returning cloud photos to weathermen.

Expanded Uses

Great potential in distant communications exists in many areas beyond the telephone and television industries. The Federal Aviation Agency is interested in establishing service for transoceanic flights, so often out of reach of HF and VHF radio. This could come by mid-1967. The same advantage would be available to ships on the high seas.

The possibility of broadcasting television directly to the home via satellite has received considerable attention recently. Though the practicality of such a system may be questioned for some years, technologically it is not difficult to imagine.

Satellites vs. Cables

Of immediate interest to international common carriers is the commercial value of satellites. Coaxial submarine cables—only ten years old themselves—remain as the backbone for communications across the oceans. More high-capacity cables will be installed in the next five, ten, or even more years. From the present 3500 voice channels, international service could jump to over 7000 in the next five years, and triple in 10 years. Much of this growth, especially between dense traffic areas, can be handled by high-capacity cable. Of course, satellites will absorb their share of the market.

Probably more important are the new markets satellites will create for international communications. Continent-to-continent television, for example, had to wait for satellites—cables lacked the necessary bandwidth. Satellites will likewise provide a logical medium for high-speed data transmission between commercial centers around the world.

We have obviously crossed the threshold toward complete global telecommunications.

(Editor's note: Additional information on specialized satellite subjects may be found in previous issues of the Demodulator. The May 1962 issue includes a more detailed examination of orbital mechanics in communications satellites. The August 1966 edition concerns echo suppression, a technique vital to communications through satellites. A discussion of ground stations is planned for the January 1967 issue.)

Lenkurt Electric Co., Inc.
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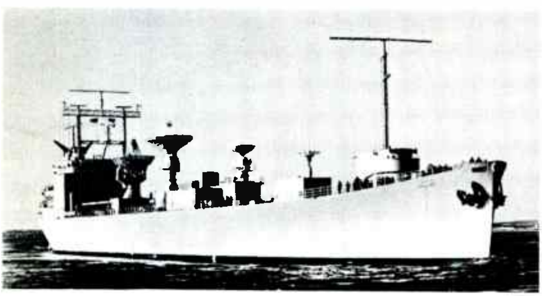
46A Links Apollo, Earth



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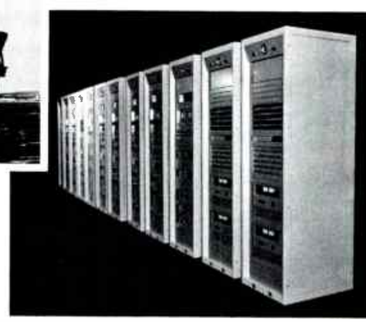

Communications between the Apollo moonship and earth will be conveyed by Lenkurt 46A multiplex equipment aboard tracking ships stationed around the world.

Signals from the spacecraft will be received at ships like the USNS Vanguard, and relayed by communications satellite to the mainland. The 46A also is used at a number of satellite ground stations.

Suited for small 4-channel systems, or high-density routes of 600 channels or more, the 46A multiplex system has the unmatched ability to handle voice and data signals in any proportion. For information, write Lenkurt, Dept. B720.

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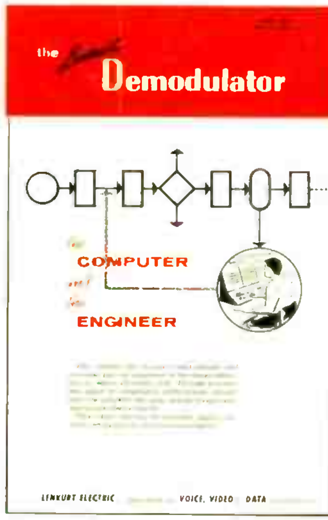
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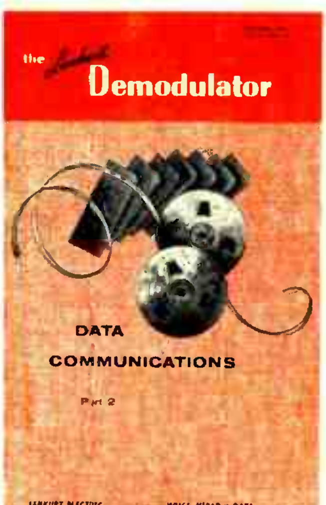
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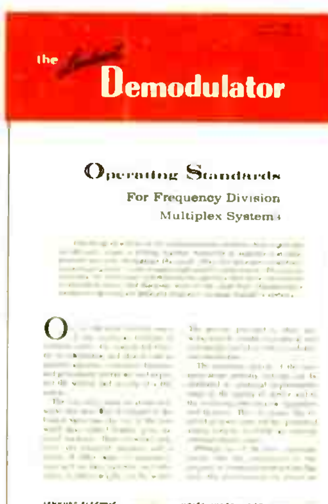
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