GTE LENKURT

DEMODULATOR

JANUARY/FEBRUARY 1975

PCM Update

PARTS 1 AND 2

NEW *DEMODULATOR* CIRCULATION PROCEDURE

Since the *Demodulator* was established in 1952, it has shown a steady increase in subscribers. By the end of 1974, the number of subscriptions, including both foreign and domestic, had reached 45,450 (the Spanish translation has an additional 5,000 subscribers). It is unfortunate that, while the number of subscribers has steadily increased, so also have the costs of producing and circulating the *Demodulator*, especially the mailing costs. Because these are times when economizing is a must in practically any endeavor, a decision has been made to modify the *Demodulator* circulation procedure.

Beginning with the January and February issues of the *Demodulator*, two monthly articles will be combined in one mailing so that there will be six, rather than twelve, mailings a year. In this way, mailing costs will be substantially reduced, but the number of monthly articles published yearly will remain the same as before. GTE Lenkurt hopes that this economizing step will not be a great inconvenience to the readers of the *Demodulator*.

Jose C. de Leon, Editor
The GTE Lenkurt Demodulator

PCM Update - Part 1

Pulse code modulation techniques have established a significant foothold in the field of communications. Here is an update on some of the recent developments, concepts, and possibilities in this field.

Pulse Code Modulation (PCM) was conceived by Alec H. Reeves in 1937 and patented in 1938. Although some of its potential as a telecommunications technique was recognized then, the concept nevertheless remained dormant a whole decade for lack of the necessary electronics. With the development of high-speed, solid-state switching devices shortly after World War II, the possibility of a working PCM system stirred interest in the telecommunications industry, and 1948 saw an increase in PCM studies. Still, it wasn't until the mid-fifties – when the use of semiconductors became economically feasible – that PCM became a practical possibility. The first commercial U.S. system was installed in Akron, Ohio, in 1962 for Bell Telephone of Ohio.

The original Western Electric T1 carrier system, which included the D1 PCM channel bank, was conceived as a short-haul, heavy-route carrier system designed to relieve cable congestion in metropolitan areas. The T1 system offered the reliability and quality of voice transmission necessary for use on direct trunks between Class 5 end offices and toll-connecting trunks, or tandem trunks between Class 5 end offices and Class 4 toll or tandem offices (see Figure 1). After the introduction of the T1 system by the Western Electric Company, many independent manufacturers began to produce similar T1-type equipment.

In spite of its popularity, however, the D1-type channel bank did not have the necessary transmission quality to be widely applied in the intertoll networks, where there could be as many as seven trunks in tandem, from an originating Class 4 office to offices 3-2-1 and back down 1-2-3 to the terminating Class 4 office. The D2 channel bank was subsequently designed and introduced into the telephone networks, particularly for service in the intertoll network (see Figure 2). Besides better transmission characteristics, it offered features tailored to intertoll and special service usage, and also offered some maintenance and alignment simplifications. Independent manufacturers again produced equipment that was end-to-end compatible with the D2 channel banks.

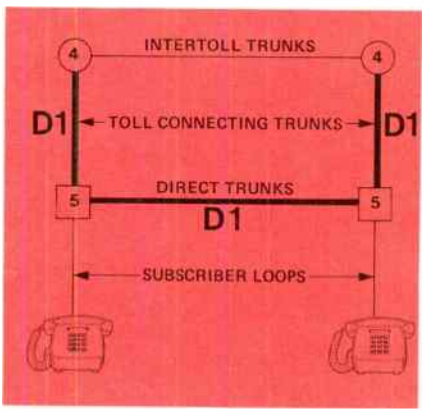

Figure 1. D1 channel bank service.

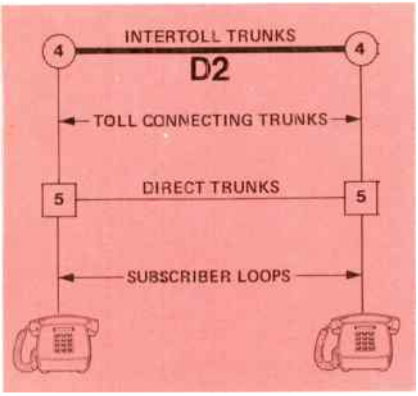

Figure 2. D2 channel bank service.

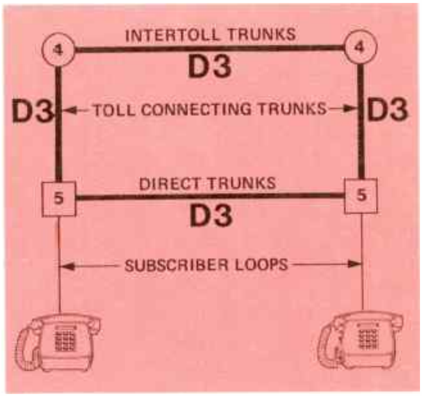

Figure 3. D3 channel bank service.

While the D2's higher quality transmission performance would allow it to be applied to direct or toll connecting trunks, its intertoll service features made it too expensive to apply in the lower steps of the network, a condition which led to the further development of the D3 channel bank. The D3 channel bank was originally conceived as a technological update of the D1 channel bank, to be used in the same type of service. However, it was soon realized that, with the new technologies available, such as integrated circuits, there was little or no cost penalty in making the D3 operation compatible with D2. This would afford greater flexibility for D3, allowing it to be used in intertoll service as well as direct and toll connecting service (see Figure 3). While channel bank designations D1, D2, and D3 are Western Electric designations, they have almost become generic terms used to describe D-type compatible equipment manufactured by independent telephone companies.

The Western Electric D2 is packaged as a 96-channel terminal with some circuits shared over all 96 channels. But electrically it functions as four nearly independent 24-channel terminals. Each 24-channel transmission package, called a "Digroup," operates over a separate 1.544 Mbs T1 line. The independent manufacturers, however, have all designed D2-compatible channel banks as 24-channel terminals, completely self-contained. These terminals all have the higher transmission qualities of the D2, and features applicable to intertoll and special service usage.

The W.E. D3 was also designed as a 24-channel terminal, completely self-contained, with the same transmission qualities as the D2. Besides the 96- versus 24-channel package, the main differences between D3 and D2 are different channel sampling sequence, slightly different alarm sequence, and different maintenance features. Since the independent manufacturers' channel banks were already in 24-channel packages, very little was required to make them completely end-to-end compatible with either W.E. D2 or D3. Most manufacturers have options available which give their channel banks D2/D3 capability. This makes these channel banks applicable in all types of trunk service because of the flexible maintenance and interface features in the various types of channel units.

PCM Subscriber Carrier

PCM subscriber carrier has been a development of independent carrier manufacturers, with no equivalent offering, as yet, from Western Electric. Consequently, no industry-wide standardization has been set, and some manufacturers offer PCM subscriber carrier based on their D1, while others offer subscriber carrier based on their D2/D3.

In this context, "subscriber carrier" is defined as a point-to-point system, in contrast to "station carrier," which is defined as a distributed system. Subscriber carrier allows the extension of central office service to a single remote point for distribution (see Figure 4). Station carrier allows the extension of central office service to several distributed points along the extension route. PCM "subscriber" carrier generally allows subscriber loops at its outward (or subscriber) end to be equal to or longer than the loops allowed at Class 5 end offices (up to 1600 ohms); other types of "station" carrier generally limit the subscriber loop to 200 ohms or less. Therefore, PCM subscriber carrier actually adds to the telephone network, for a possible total of 11 links instead of the traditional 9 links previously allowed between subscriber loops.

There is a possible disadvantage in using PCM subscriber carrier based on the D1 format as opposed to that based on the D2 format, since the D2/D3-based subscriber carrier has better transmission performance, thereby maintaining subscriber-to-subscriber performance at a higher level for longer distances. Another major advantage of the D2/D3-based PCM subscriber carrier is that its line format is compatible with digital time division multiplex (TDM) switching, which is to be introduced into Class 5 offices in the not-too-distant future.

TDM Switching

Time division multiplex switching equipment for Class 4 toll offices or tandem offices is now being developed by both Bell Labs (No. 4 ESS) and Automatic Electric (No. 3 EAX); these will be introduced into the telephone network sometime between 1977 and 1980 (see Figure 5). These switching centers operate at 1.544 Mbs to provide high-capacity switching from one PCM line to another. This is accomplished without demultiplexing from 1.5 Mbs down to voice frequencies for switching, and back up to 1.5 Mbs for

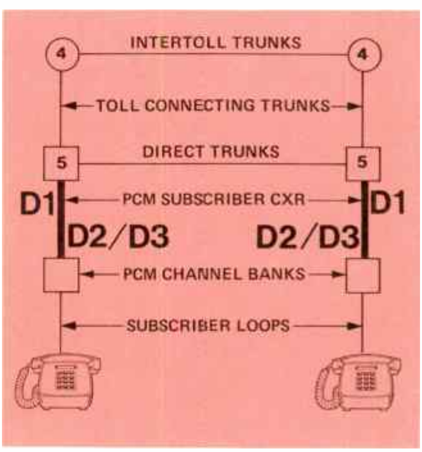

Figure 4. PCM subscriber carrier service.

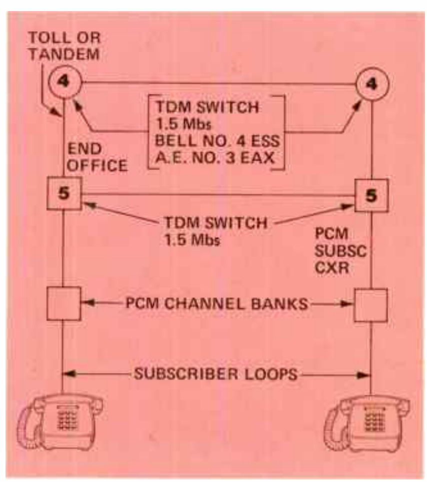

Figure 5. Digital switching capability.

re-transmission. Only the D2/D3 PCM 8-bit word format will be allowed in these switching centers, so connection of D1 channel banks directly to a switch at 1.5 Mbs is prohibited.

Soon after introduction of digital switches into the toll or tandem offices, TDM switches will be introduced into Class 5 end offices. It appears now that these switches will also allow only the D2/D3 format. Therefore, PCM subscriber carrier based on the D2/D3 channel bank has a distinct advantage.

Digital switching is accomplished by the switch breaking down the received 1.5 Mbs pulse stream into individual channel 8-bit PCM words, storing them until the appropriate channel time comes along in another addressed 1.5-Mbs pulse stream, and then transmitting them in the appropriate time slot (see Figure 6). Thus, information in a channel 2 slot coming from an end office to a No. 3 EAX may be switched to a channel 15 slot going to another office; this connection is maintained through the switch as long as the conversation lasts. However, the actual "connection" is only made at the appropriate time slot,

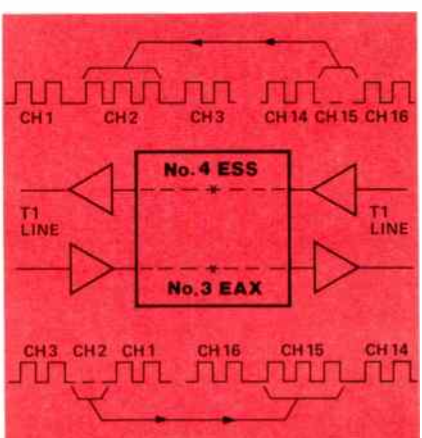

Figure 6. Toll or tandem TDM switching (8 bit PCM).

leaving the 23 other time slots available for other channel connections.

In addition to the voice transmission considerations for each of the types of telephone trunks, the signaling requirements of each must be considered. In the intertoll network, E&M signaling is most prevalent, with loop signaling capability also required. When common channel interoffice signaling (CCIS) is introduced into the network, a flexible PCM carrier must have the ability to accommodate this type of signaling. In direct or toll connecting trunk use, the most prevalent type of signaling is loop signaling, with some requirements for E&M and Foreign Exchange (FX).

A truly flexible PCM channel bank to be used for PCM subscriber carrier should be able to accommodate many types of signaling, including bridged, divided and superimposed ringing, paystation, and foreign exchange. Channel banks of the D2/D3 type provide improvements over the D1 channel banks, including a greater signaling flexibility, improved noise and distortion performance, and a standardized digital switching format.

PCM Format

The three basic operations necessary for PCM voice operation are sampling, quantizing, and encoding. Basic sampling theory indicates that an analog wave, such as a voice representation, can be reproduced with very little distortion if the sampling frequency is at least twice as high as the highest frequency to be transmitted. Since voice transmission of about 4-kHz bandwidth is to be transmitted in the telephone network, the sampling rate of 8 kHz has been standardized in the United States and is used for D1, D2, and D3 operation.

The number of steps used to represent the sample (or quantize it) determines the quality of the recovered voice signal at the receive output of the system. Or, put another way, the number of steps determines the "quantizing distortion." While the D1's use of 128 quantizing steps was adequate for direct or toll-connecting trunks, the D2 (and D3) required 256 quantizing steps to provide adequate quantizing distortion for the intertoll network.

The number of binary bits necessary to give a digital code representation of 128 steps is 7, while to represent 256 steps requires 8 bits. Therefore, the D1 encoded "word" is a 7-bit word and the D2 or D3 word is an 8-bit word (see Figure 7).

The most important and most basic characteristic of a PCM system is its frame format. Once this is understood, the whole system can be understood. Since the sampling rate is 8 kHz, the time between successive samples of a given channel is 125 μsec. The "frame" length, then, is 125 μsec, during which all channels in the system must be sampled, quantized, and encoded in sequence. Then, the first channel is sampled again, and the sequence repeated (see Figure 8).

For the D2/D3 system, 24 channels with 8-bit words are included in one frame, giving 192 bits. Then a 193rd bit is added for framing and signaling control. Since it is "shared" between

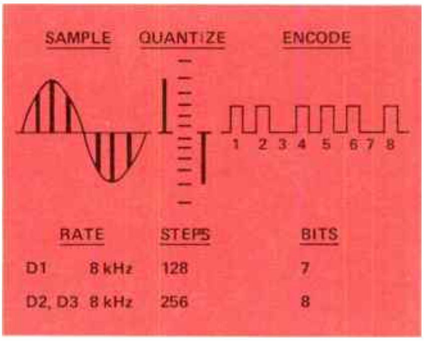

Figure 7. Pulse code modulation coding.

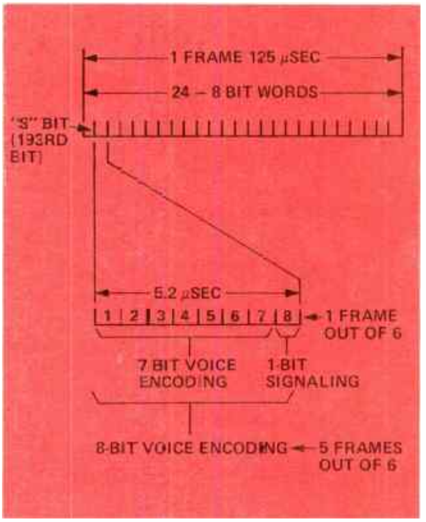

Figure 8. D2/D3 PCM frame format.

these two functions, it is called the "S" bit.

Each of the 8-bit words in the 125 μsec frame length is allowed 5.2 μsec. For the least amount of quantizing distortion, all 8-bits should be used for voice encoding. However, signaling for each channel (such as dial pulsing and supervision) must also be transmitted. But, since dial pulsing and supervision are of such low frequency compared to voice transmission, it can be sampled and transmitted at a slower rate than the voice. Therefore, a compromise is reached to transmit 8-bit voice samples for 5 frames out of 6, and in the 6th frame, transmit a 7-bit voice sample and a 1-bit signaling sample.

Future plans call for common channel interoffice signaling (CCIS), whereby signaling for all channels is carried by a single data channel. This will allow 8-bit encoding at all times, and improve quantizing distortion by about 2 dB.

Coding

The D2/D3 coding utilizes a nonuniform coder-decoder (Codec) transfer characteristic (see Figure 9). That is, low level samples at the input of the coder are compared against (quantized by) small steps, whereas high level samples are quantized by large steps. This is a compression technique which maintains relatively constant quantizing distortion over a wide range of talker volumes.

For a given size of coder input sample, one size of decoder output sample is produced. However, for several samples close to the size of the first sample, the same decoder output sample will still be produced. This is quantizing distortion, in that not all samples can be reproduced exactly. But, by using small quantizing steps for small samples and large quantizing steps for large samples, the same relative error (in dB or %) can be maintained for small and large samples. Since this is a compression technique, a typical compression curve is described, where small input signals are reproduced nearly linearly, but large signals are attenuated to form the output compressed signal. Since this is a digital system, it is desirable to have a segmented compression characteristic rather than a continuous one. Therefore, from subjective talker-listener considerations, the curve is made up of 8 positive segments for positive samples and 8 negative segments for negative samples. The 8th segments, positive and negative, are co-linear, so the curve is called a 15-segment curve (see Figure 10).

If a continuous curve were fitted to the segmented one, and its shape described by the equation,

$$Y = \frac{\ln(1+\mu x)}{\ln(1+\mu)}$$

the closest fit occurs when μ = 255. This compares to μ = 100 for D1.

The D2/D3 digital code for a sample is arranged as shown in Figure 11. Bit No. 1 of the 8 bits is used to designate the sign of the sample. If it is a positive sample, bit 1 is a 1. If it is a negative sample, bit 1 is a 0. Bits 2, 3, and 4 indicate on which of the eight segments of the compression curve the sample lies, after it is known whether the sample is positive or negative. Bits 5, 6, 7, and 8 tell which of the 16 steps within a segment most closely represents the height of a sample.

For example, if a small sample is to be encoded, it would be accomplished as shown in Figure 12. The sample at the coder input is positive, so bit 1, the sign bit, is a 1. Next, the sample falls within the coding range of seg-

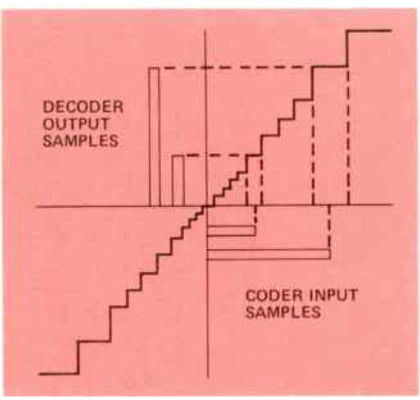

Figure 9. Non-uniform Codec transfer characteristic.

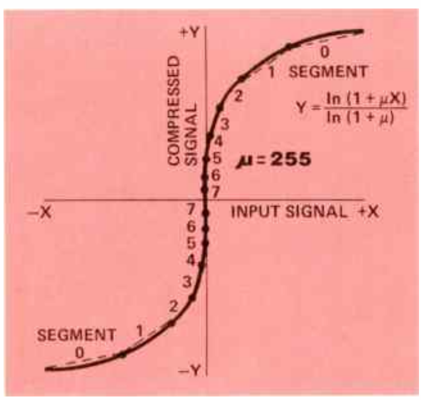

Figure 10. D2/D3 segmented compression characteristic.

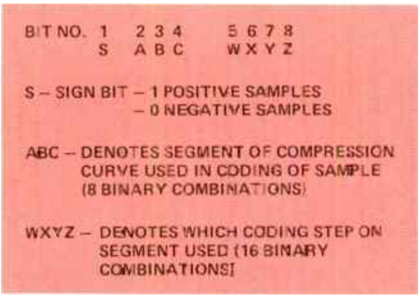

Figure 11. D2/D3 compression code (8-bit).

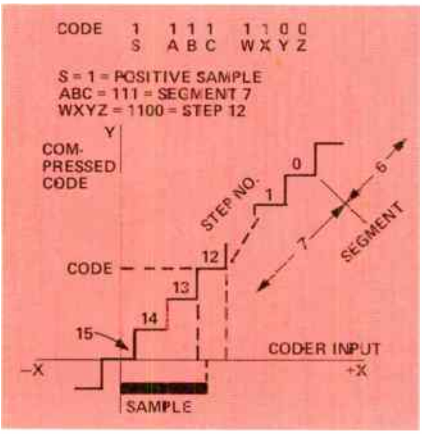

Figure 12. D2/D3 coding characteristic example.

ment 7, so bits 2, 3, and 4 are used to give the binary representation of 7, which is 111. Then, the final sample height is best described by step 12 of segment 7. Therefore, bits 5, 6, 7, and 8 are coded with the binary representation of 12, which is 1100. The 8-bit PCM word representing this sample is therefore: 1 1 1 1 1 1 0 0

The result of the 8-bit code and compression characteristic with μ = 255

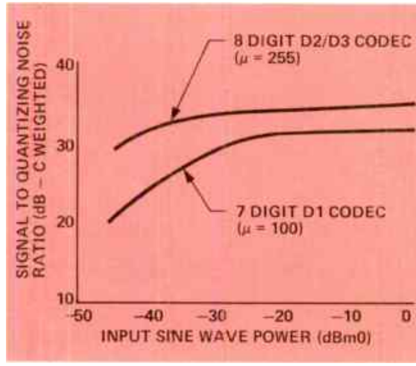

Figure 13. Signal-to-distortion performance (D1 and D2/D3 logarithmic codecs).

gives a substantially better signal-to-quantizing distortion ratio for the D2/D3 systems compared to that of the D1 system.

With 8 bits for coding vs 7 bits, signal-to-quantizing distortion ratio is improved by about 6 dB in the range of higher level signals. Then, the compression characteristic with μ = 255 gives the D2/D3 a relatively constant signal-to-quantizing distortion ratio over about a 40-dB range of input talker levels. This is an improvement over the μ = 100 compression characteristic for D1, which gives only about a 30-dB range over which the signal-to-quantizing distortion ratio is relatively constant (see Figure 13). This is one of the major improvements in the D2/D3 systems which was required to provide toll-quality service.

The following part of this series will deal with common equipment, channel-unit functions, and with some of the various signaling arrangements possible with modern PCM equipment.

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2. de Leon, Jose C., ed. *Readings in Pulse Code Modulation.* San Carlos, California: GTE Lenkurt Inc., 1973.

PCM Update - Part 2

The signaling that has to be transmitted between two telephone offices takes two forms; Pulsing and Supervision. A PCM system is an ideal transmission medium for those signaling forms because it is a digital system transmitting digital information.

Part 1 of this series described some of the concepts and applications of PCM in subscriber carrier and TDM switching. This issue deals with PCM signaling arrangements, common equipment, and channel unit functions.

Telephone dial pulsing is relatively slow, compared to sampling rates in the PCM system. Therefore, the pulsing digits can be sampled, transmitted, and reconstructed with ease (see Figure 1). The reconstructed pulse is never perfect, but the distortion can be held as low as desired by adjusting the signaling sampling rate. In the D2/D3 carriers, this has been optimized to accommodate many different types of interoffice signaling.

Framing and Signaling Frame Identification

In the D2/D3 frame format, the 193rd bit – the timeshared or "S" bit – is reserved for terminal framing functions and signaling framing identification (see Figure 2). Terminal framing is required to align the receiver timing with the incoming bit stream from the remote transmitter so that channel word decoding can be accomplished in the proper sequence. To do this, terminal framing is sent as a unique repetitive pattern of 1 0 1 0 1 0---, and steps are taken in the transmitter to prevent any simulation of this pattern, on a continuing basis, by any voice or signaling transmission.

Since signaling is transmitted only once every sixth frame, a pattern is needed for the receiver to identify

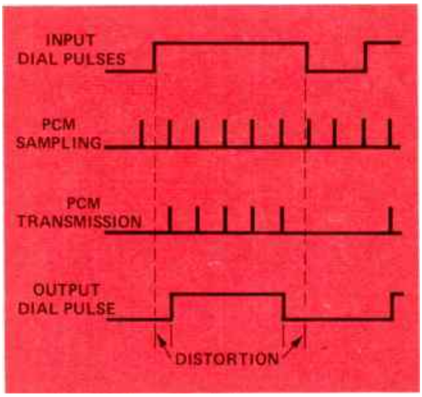


Figure 1. PCM signaling sampling.

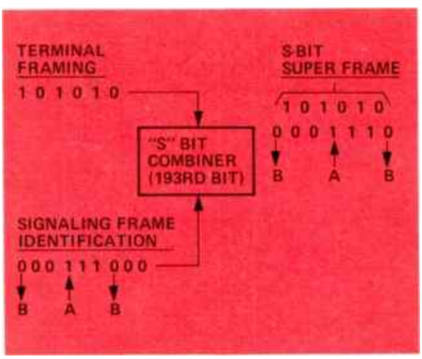


Figure 2. D2/D3 framing and signaling identification.

which of the six frames contains the signaling. This pattern is the signaling frame identification pattern, which is again a unique repetitive pattern of 0 0 0 1 1 1. These two patterns are combined to time-share the 193rd bit in each frame. A complete sequence of these time-shared bits requires 12 frames, and is called a "Super Frame."

With this signaling frame identification pattern, two signaling conditions, A and B, can be sent in alternating 6th frames. That is, conditions A and B are both sent within 12-frame increments, each condition appearing six frames apart from the other. These are then interleaved to send a signaling condition every 6th frame. This is accomplished in the following manner:

1. Signaling condition A is transmitted in the 8th bit of all 24 PCM channel words in the frame whose 193rd bit is a 1, preceded by the framing pattern 1 0 0 0 1.
2. Signaling condition B is transmitted in the 8th bit of all 24 PCM channel words in the frame whose 193rd bit is a 0, preceded by the framing pattern 0 1 1 1 0.

This framing scheme allows each voice channel to send up to two different conditions of signaling. This gives a capability of two signaling channels per voice channel, in each direction (see Figure 3). Since each signaling channel is sampled every 12th frame, this gives a sampling rate of 1.5 milliseconds (12 x 125 μs).

In the frames where signaling is allowed, each signaling channel uses the 8th bit of the PCM word. Since this 8th bit can be either 1 or 0, each signaling channel can send two states of signaling. Therefore, a total of four states of signaling can be sent in each direction of transmission.

These signaling channels can then be used either separately or combined, depending on the signaling requirements imposed by the service to which

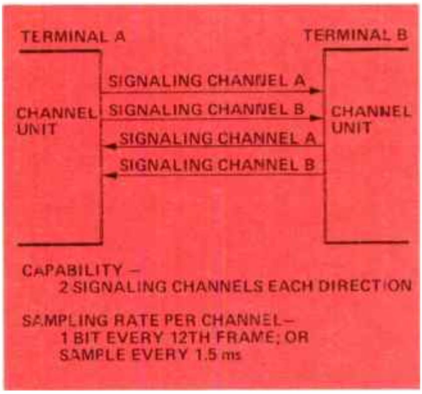


Figure 3. D2/D3 signaling channel capability.

the channel bank is committed. This feature is one of the greatest contributors to the flexibility of the D2/D3 PCM channel bank, allowing it to be used for signaling in subscriber, direct, toll connecting, tandem and intertoll applications merely by assigning these signaling channels in the appropriate way. For example, in E&M signaling, only two signaling states, on hook and off hook, are required in each direction (see Figure 4). This requires only one signaling channel; therefore, the two signaling channels available are combined to give a faster sampling rate of 0.75 milliseconds, or signal every 6th frame. Since low sampling distortion (less than 2%) is required in E&M

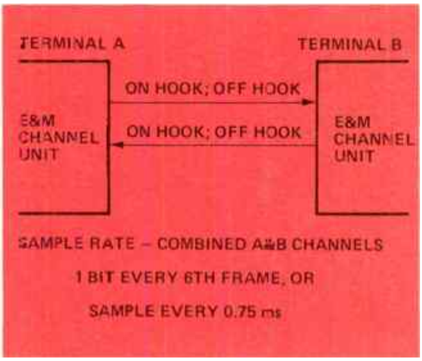


Figure 4. D2/D3 E&M signaling.

signaling, this rate of signaling transmission is required to accomplish it.

In contrast to E&M signaling, foreign exchange signaling requires up to four states of signaling in one direction. Also, two separate conditions, such as loop closure and ring ground, require transmission simultaneously (see Figure 5). Here, the two signaling channels can be used separately to send one condition every 12th frame; the signaling sampling rate of 1.5 milliseconds is adequate for the type of functions being transmitted.

Common Channel Interoffice Signaling

Use of the 8th bit in the PCM channel word for signaling in every 6th frame causes a degradation of about 2 dB in the signal-to-quantizing distortion ratio. Provision has been made in the D2/D3 format to upgrade the channel performance in the future by this 2 dB. This can be accomplished by allowing 8-bit voice encoding in all frames, including every 6th frame.

Signaling for all voice channels may be transmitted over a "common signaling channel" instead of using the 8th bit every sixth frame for each voice channel. First, all signaling for the voice channels is gathered into external equipment which multiplexes and codes it into a 4-Kbs data stream. This 4-Kbs data stream is substituted into the portion of the "S" bit stream previously occupied by the signaling frame identification pattern (see Figure 6). There is no frame now reserved for signaling, so the signaling frame identification pattern is not needed. If its share of the S bit data capacity is looked at as a 4-Kbs data stream, the common channel interoffice signaling (CCIS) data stream can be directly substituted. This flexibility of the D2/D3 format allows for future use of CCIS as it is introduced into newer type switching offices.

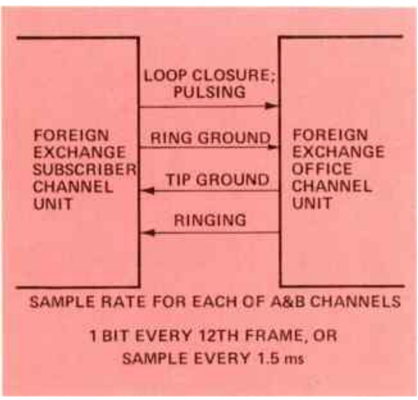


Figure 5. D2/D3 foreign exchange signaling.

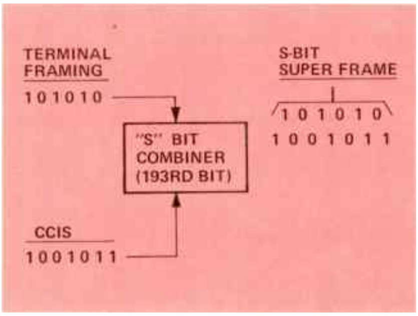


Figure 6. D2/D3 framing and common channel interoffice signaling.

Physical Organization of D2/D3 Carriers

In its simplest form, a PCM terminal contains the following functions:

(1) channel units to provide the individual voice channel interface between the PCM terminal and the telephone office, (2) a transmit common unit to provide all necessary functions to encode all voice channels for PCM transmission, (3) a receive common unit to recover the PCM transmission and decode it back to individual voice channels, and (4) an alarm and control unit to remove the terminal from service when a system failure occurs. If the interface between channel units

and common equipment is chosen carefully, both the simplest packaging of the terminal and the greatest flexibility can be achieved at the same time (see Figure 7). In all of the latest generation PCM terminals, this interface is chosen so that each channel unit contains all those functions which are truly limited to only that one voice channel; allowing common units to contain all those functions which are truly common to all voice channels in the terminal.

Common Equipment

The transmit common equipment (see Figure 8) contains a clock (at 1.544 MHz for D2/D3 terminals) and all the timing counters required for both common coding and timing functions and for individual channel sampling control. The coder converts channel PAM (pulse amplitude modulation) samples from a common interface bus to PCM words on a time multiplexed basis. Signaling is added to each channel PCM word at the appropriate bit time, and a bipolar converter converts from the unipolar pulses used in the terminal logic circuits to a bipolar signal more suitable for transmission on a repeated line.

The receive common equipment (see Figure 9) starts the recovery by converting from the bipolar line transmission format to unipolar pulses which can be processed by logic circuits. A 1.544 Mbs clock is recovered from the incoming bit stream to synchronize the receive operations to the far-end transmit operations. Timing counters provide timing both for common decoding and for individual channel reconstruction of the transmitted signal. Signaling bits are recovered from the PCM stream and routed to the appropriate individual channels for signaling recovery.

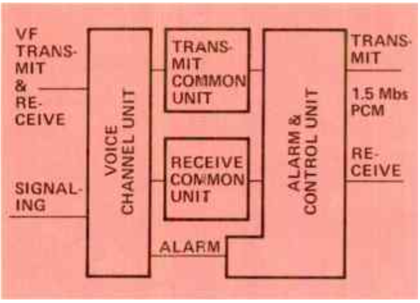


Figure 7. PCM terminal functions.

Channel Units

The real flexibility of the PCM system is achieved in the channel units. The majority of the voice transmission functions are common to all types of channel units. By changing

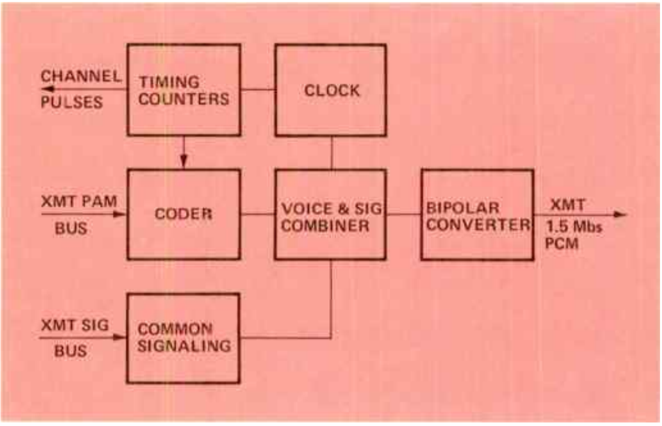
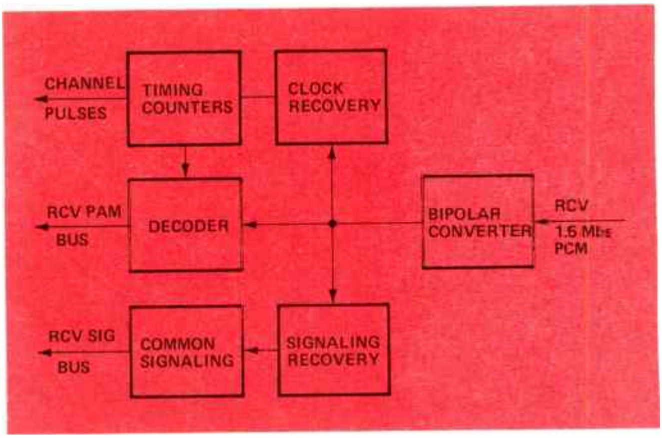


Figure 8. PCM common transmit functions.

Figure 9. PCM common receive functions.



the signaling functions and office interfaces, many types of channel operations can be created, thereby creating many types of system applications. For example, in intertoll usage, the most common type of channel operation required is to provide 4-wire voice transmission and E&M —independent transmit and receive— signaling (see Figure 10).

In the transmit voice path, an amplifier provides level adjustment and office impedance matching. A low-pass filter limits the incoming frequency band to less than half the 8 kHz sampling rate, for proper sampling. Finally, a transmit gate connects samples of the vf signal to the common pulse amplitude modulated bus at the appropriate time to time multiplex all channel PAM samples.

In the receive voice path, a receive gate connects PAM samples from a common bus to the individual channel at the appropriate time to time demultiplex all channel PAM samples. The low-pass filter reconstructs the original vf signal, and the amplifier provides level control and office impedance matching.

In the signaling portion of the unit, a signaling transmitter converts from the signaling interface levels of the office, to logic levels usable in the terminal. Then a transmit signaling gate connects samples of the office pulsing or supervision to a common signaling bus at appropriate times to time multiplex all channel signaling samples. In the receive direction, a receive signaling gate connects samples from a common bus to the individual channel at the appropriate time to demultiplex all channel signaling samples. Then, a signaling receiver reconstructs the pulsing or supervision and provides the proper interface to the office.

To create another type of channel

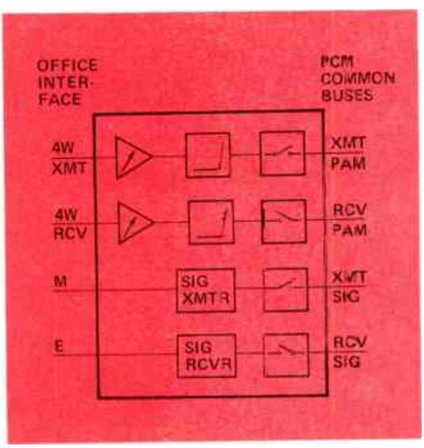


Figure 10. Four-wire PCM channel unit.

unit for different operations—direct trunk usage, for example—many of the channel functions stay the same. The requirements for different functions are only in the voice-path-office interface and in the signaling interface to the office. For instance, in a 2-wire loop signaling channel unit (see Figure 11), only a hybrid function is added to the amplifier, filter, and gate functions already available in the 4-wire design to provide for a different office interface in the voice path. Then, different signaling logic and office signaling interface is used with the signaling gate design already available to produce the required office interface. This same technique is also used to produce the many different channel units used in toll connecting and subscriber applications.

Important Factors

The principal factors contributing to the increasing flexibility of PCM carrier include a standardized D2/D3 PCM coding format which allows high enough voice quality for the most demanding portions of the telephone network, while allowing low enough cost to apply to the least stringent portions of the network, and time

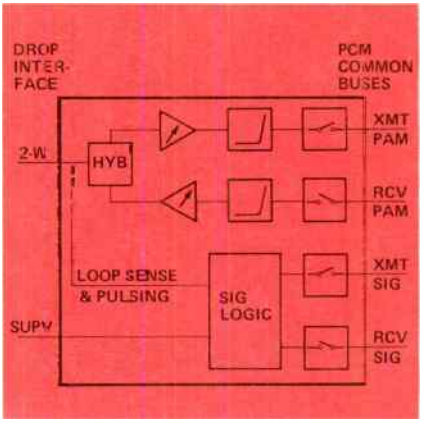


Figure 11. Two-wire PCM channel unit.

division multiplex digital switching machines to be applied at the network switching points. Additional factors that enhance the operation of PCM are the nature of PCM, which lends itself ideally to providing signaling and supervision transmission as a part of the PCM bit stream, and the physical organization of the latest generation D2/D3 channel banks, which allow standard interface PAM and signaling buses to provide maximum flexibility of application of signaling and voice functions.

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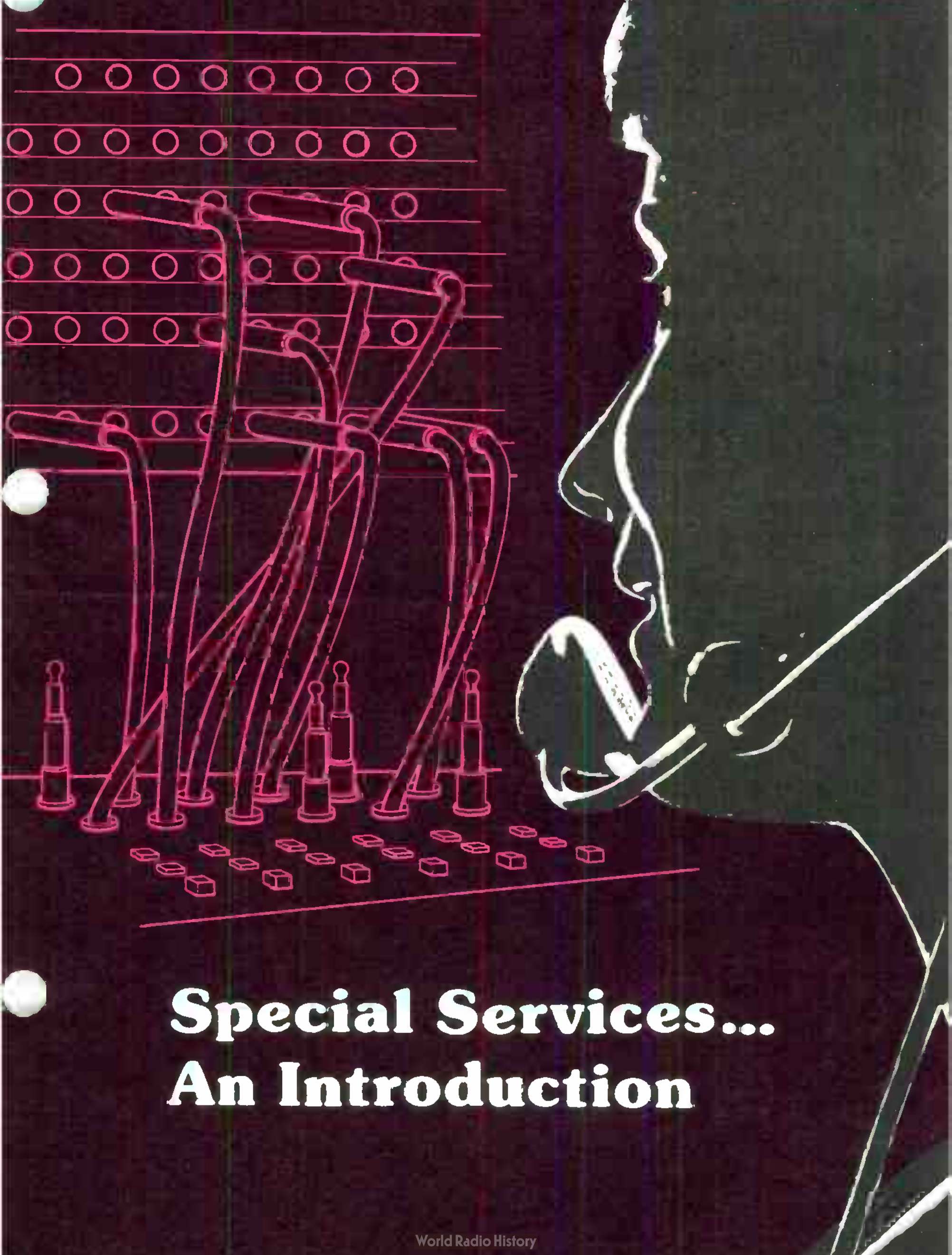
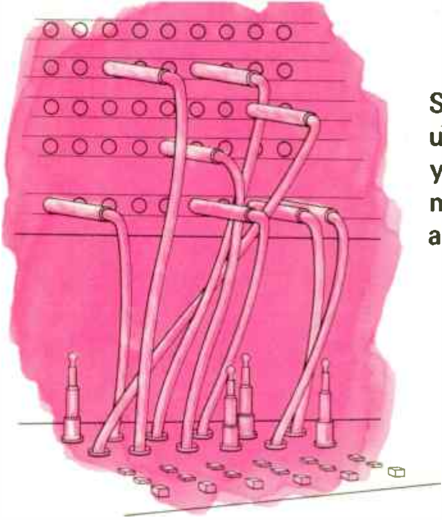
DECEMBER 1974



Special Services... An Introduction



Special telephone services are widely used throughout the business world, yet their versatility and multiplicity make understanding their technical aspects a difficult task.

"Special services" is a nebulous term which may be used in reference to a range of literally thousands of telephone applications, or to a portion of that range; the exact definition of the term depends upon what telephone company, customer, or equipment supplier is doing the defining. It is generally accepted, however, that special services are supplements to the traditional standard services known as "plain old telephone service," or POTS, and that they encompass all applications other than residence, coin, and non-PBX business telephones.

POTS

Standard telephone service accepts and routes the telephone number codes, ranging from the single-digit zero (operator) to the seven-digit local and ten-digit direct distance dialing numbers, that are used to identify non-special service telephone subscribers, or customers.

The telephone service with which most people are familiar is the simple residence phone (see Figure 1), consisting of a subscriber station set connected by a 2-wire loop or cable pair to a central office (C.O.), and generally using a loop signaling technique. Equipment in the C.O. provides switching arrangements that enable connection of the subscriber station to lines leading to other subscriber stations. When both are within the same telephone exchange area, the connection would be to the called station's subscriber loop. For calls outside of the local area, the connection would be to a trunk leading to the central office serving the called station (a trunk is a communications channel commonly used to carry all calls of the same class that are generated between two switching centers, such as central offices; a loop is associated with one particular subscriber location). Signaling functions, also provided by the C.O. in a POTS system, allow a caller on the originating end of a line to inform a particular party at the receiving end that a message is to be communicated, and also monitor the on-hook/off-hook condition of the stations involved (supervisory signaling).

In this simple form, when a subscriber removes the handset from the cradle of his instrument (the station goes "off-hook"), a switch contact in the instrument closes, completing a dc path through a loop consisting of the

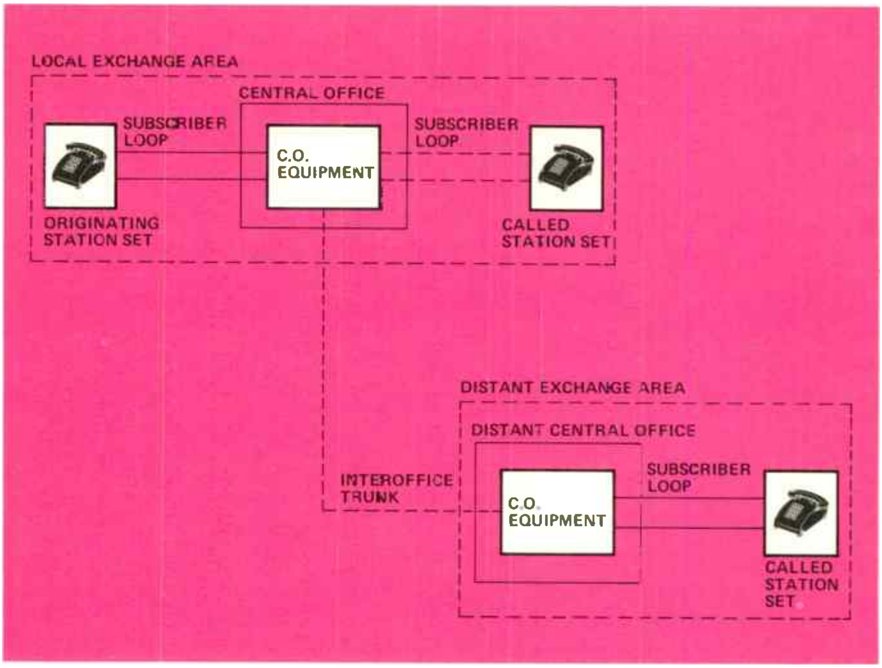


Figure 1. In POTS, C.O. equipment connects an originating station set directly to any other station within the same local exchange area (exchange service), or to a trunk circuit for connection to a distant station (toll service).

subscriber station set, a line relay, and either a battery or ground in the central office.

When this loop is completed, the line relay is energized, connecting the subscriber's line to the C.O. equipment. The C.O. then extends dial tone to the station; the dialing process generates pulses which indicate to the C.O. equipment what station is being called. The originating station is connected to the line associated with the desired station; when the called party answers, the two stations are linked and the signaling function becomes merely supervisory until the call is terminated.

Besides this loop signaling technique, which places the signaling current on the same conductors as those used for voice transmission, the telephone network uses a technique known as "E&M" signaling, which is characterized by the use of separate facilities for the signaling and voice transmission. This method employs two leads, designated E and M, to connect the signaling equipment to the signaling facility. The M lead transmits a battery or ground indication from the central office to the distant end of the circuits, while incoming signals are received on the E lead. (See the July, 1966, *Demodulator* for a more extensive discussion of signaling techniques.)

Special Service Categories

Special services are usually some form of extension of the local-telephone type of service, and almost always make use of the same facilities and techniques provided for regular telephone service. In the broadest sense, they are intended to extend signaling functions to subscriber stations whose locations preclude standard telephone loop service.

A residence phone would be provided special service, for example, if the station were beyond the pulsing limit of the standard C.O. equipment. This limit is the maximum amount of distortion — caused by the loop line impedance — that a central office can tolerate in the dial pulses generated by a subscriber's telephone before the C.O. equipment begins to make errors. To compensate for this distortion, a long-line adapter can often be placed in the 2-wire loop at the central office as an impedance-matching device, extending the dial pulsing and ringing ranges. When the station is so far from the C.O. that the voice transmission and signaling function are both impaired, a voice-frequency (VF) repeater is commonly used in conjunction with a long-line adapter to keep the over-all system operation within acceptable limits. However, if the loop is so long that the repeater-adapter combination cannot make up for all of the line loss, the 2-wire loop can be replaced by a 4-wire circuit. In this application, two cable pairs — one for each direction of transmission — are used between the station and the central office. Because the station set and C.O. are essentially 2-wire facilities, a 4-wire terminal set is generally installed to act as a 2-wire to 4-wire conversion device in this special service.

Special services find their greatest application in the business community, where their uses extend beyond voice transmission to include such services as data, facsimile, and television transmission. It is possible, despite the number of variations and the frequent overlapping of implementations, to group all of these special services into four very general categories: foreign exchange (also known as special access), private lines, tie trunks, and common control switching arrangements (switched service networks).

A private branch exchange (PBX) switching system is very often included in the implementation of a special service. PBX systems are placed on subscribers' premises to allow limited-digit dialing—typically two or three digits are used—between telephones on the premises, and to provide access to the central office equipment.

Foreign Exchange

The foreign exchange (FX) category includes services similar to those provided by regular subscriber lines (POTS), but differs from these in transmission and signaling requirements, and in the location of the station relative to the central office serving it. A few of the classes of service in the FX category are PBX foreign exchange, wide area telecommunications service (WATS), and off-premises extension.

In a typical application, a subscriber's station is served by a switching system remote from its local central office, providing service to and from a telephone exchange other than the one which would normally serve the station location. This allows calls to and from the distant C.O. area without a toll charge; the subscriber typically pays the normal rate for local service plus a fixed monthly fee for the interoffice mileage. If, for example, the subscriber were a business

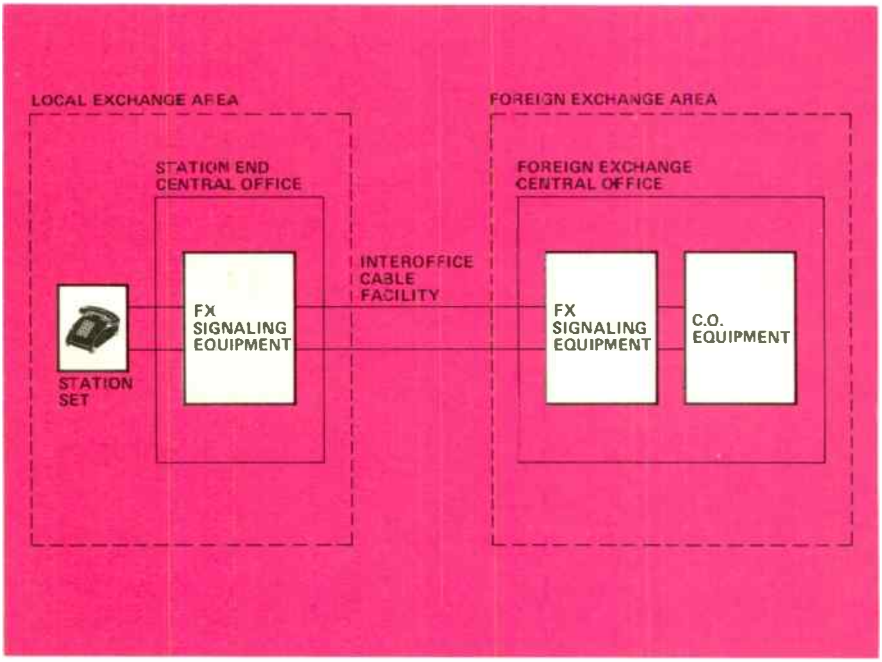


Figure 2. Implementation of foreign exchange service means that a subscriber's station set is served by equipment in a remote telephone exchange area.

with a large number of customers in an area not connected to his local central office, foreign exchange service would allow the customers to dial a number in their area and reach the business without paying toll charges.

Figure 2 shows one form of FX application. In this instance, a local cable pair connects the subscriber's set (which may actually be a PBX system) to the station-end central office, and a cable facility links the station-end and foreign exchange offices. When the subscriber's set is a telephone instrument, the service is generally referred to as a "foreign exchange line"; when it is a PBX system, the service is a "foreign exchange trunk."

Assuming that E&M signaling is being used, when the subscriber goes off-hook, the E lead at the FX end receives the signal and causes the line relay to energize, just as in POTS. This completes the loop, allowing the FX office to extend dial tone back to the subscriber set. Dial pulses are passed through the facility to the central office equipment, which determines what station is being called, and completes the call.

A call to the subscriber station set causes the FX office equipment to send ringing current through the station-end C.O. to the set. When the subscriber lifts the handset, an off-hook signal is returned to the FX office, which then closes the loop to link the two voice paths together.

The off-premises extension (OPX) class of service shown in Figure 3A allows connection of an OPX station set to the same line that serves a main

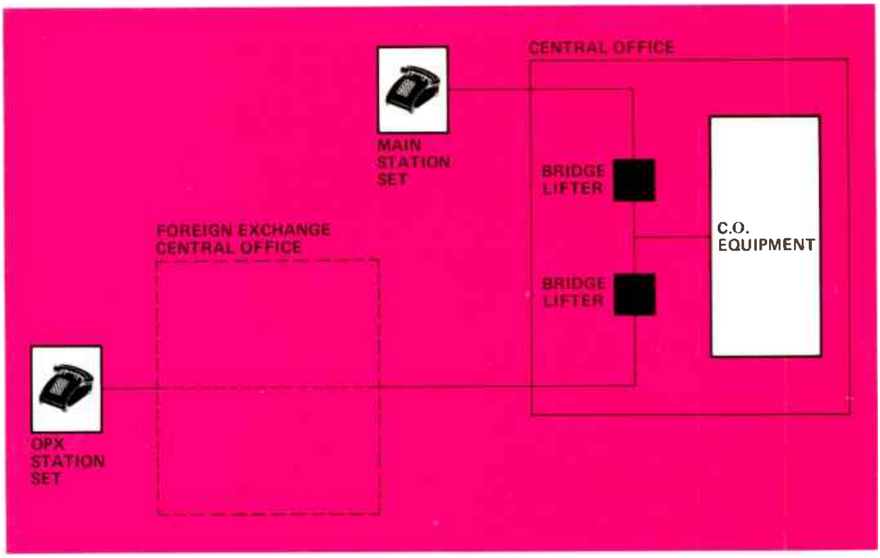


Figure 3A. An off-premises extension is commonly bridged at the central office. If the OPX station is in a foreign exchange area, it is implemented as an FX service.

station set. Typical of this application is the small businessman who, upon leaving work, throws a key switch to route calls from the office directly to his home where the OPX is located. An off-premises extension is commonly bridged at the central office rather than at the primary station location, although when used with a PBX switching system—in which the extension is generally assigned its own number, and is referred to as an off-premise station (OPS)—the bridging is often done at the main PBX station, as shown in Figure 3B.

Private Lines

Private line services are those which employ dedicated transmission facilities to link two or more subscriber station locations. The simplest private line service consists of two subscriber station sets permanently linked together by a communication channel, and provided with a signaling function.

The methods for implementing signaling in private lines are as varied as the applications, and are determined by customer requirements. These requirements may vary from no signaling at all, through signaling in one direction only, to the case where each station in a network is able to signal any one of the other stations individually, or all of the stations simultaneously.

Ringdown signaling is frequently used in private line applications; it utilizes a continuous or pulsing ac signal transmitted over the line to ring the other station or stations on the line. A simple private line using automatic ringdown signaling is shown in Figure 4. When one of the stations in this "hot line" application goes off-hook, it causes the other station to be automatically rung. This service is used a great deal at airports and other locations to allow rapid connection with taxicabs and hotels.

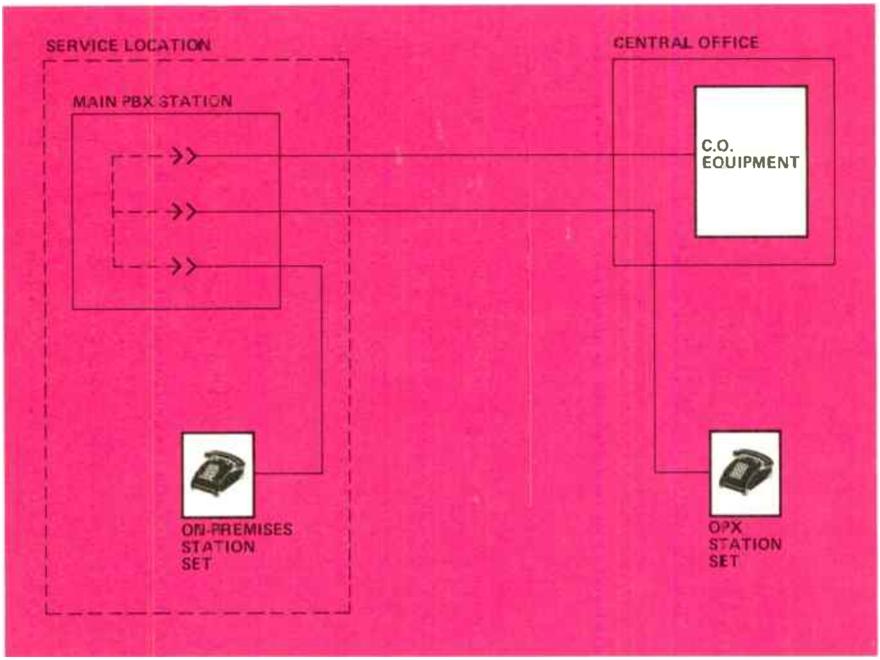


Figure 3B. An off-premises extension in a PBX system is commonly bridged at the main station location.

Another common application of private lines is the multi-station network. Generally designed for conference-type service, this application allows a large number of stations to be interconnected, with any number off-hook, without interfering with the transmission quality.

Besides the applications outlined here, private lines also are used in such areas as telemetry, mobile radio, and alarm circuit extension.

Private line service can offer both economic and technical advantages over standard telephone service. If the subscriber's telephone traffic is heavy enough between the dedicated locations, the fee for private line service could be significantly less than the charges for a large number of toll calls. Also, because they are dedicated and their performance characteristics known, private line communication channels afford a subscriber a great deal of flexibility in establishing his system.

The nature of private lines is such that they cannot be switched directly to the exchange and toll networks. However, a PBX termination provides access to these networks, as well as to other private line circuits and services (such as off-premises extensions) that may terminate at the PBX station.

Tie Trunks

Basically, a tie trunk or tie line is a communications channel used to directly connect two PBX switching systems; it is, in effect, a private line terminated at both ends by PBX facilities. Tie trunk applications can range

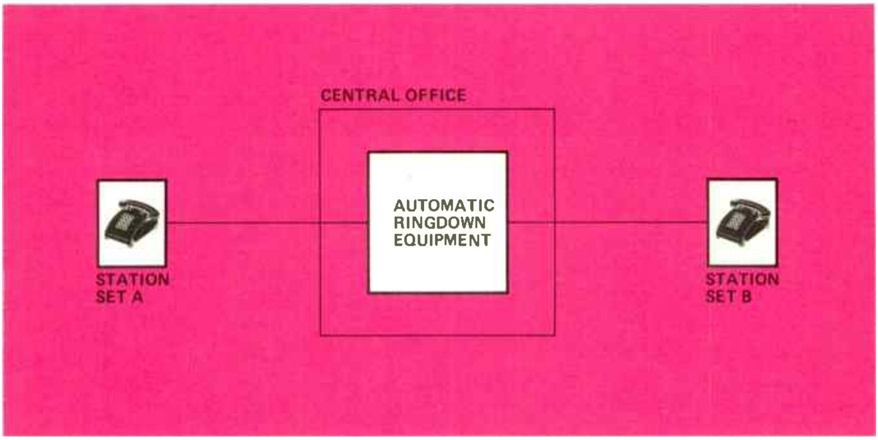


Figure 4. An automatic ringdown private line allows one station to ring the other as soon as the handset is lifted. One application of this service is in airports and similar facilities to summon taxicabs and make hotel reservations.

from a simple one-link, voice-only connection to a complex and flexible tandem network for both voice and data transmission.

Classification of tie trunks is generally in terms of their signaling and switching requirements. For example, a "tandem PBX network" employs a specific type of switching at the PBX locations, but may use any suitable signaling technique; a "ringdown tie trunk" uses a ringdown signaling technique, but may employ any suitable switching method.

The non-tandem tie trunk application shown in Figure 5 is used mainly to connect two PBX systems that are not associated with other PBX's. Besides making this connection, the trunk may also be linked to the exchange and toll networks through C.O. equipment.

If the tie trunk in Figure 5 were a dial repeating tie trunk, a user at either one of the PBX stations could pick up a handset and dial an access code – generally a single digit – to receive dial tone from the distant PBX. He could then dial the number of a station at the distant end and be automatically connected to the desired station. In some applications, the caller could access the distant PBX and place a call to any station within the distant exchange area. In a ringdown tie trunk, dialing the distant access code would connect the caller to a PBX attendant, who would then manually complete the call.

The tie trunk provides for very rapid connection of PBX stations at both ends of the line; many subscribers, however, have PBX systems at widely separated locations. Such a subscriber may have a high community of interest between these locations and require more than individual tie trunks between pairs of systems. For such a subscriber, a common control switching arrangement (CCSA), or switched service network (SSN), is available.

Common Control Switching Arrangement

The common control switching arrangement (CCSA) is a private line

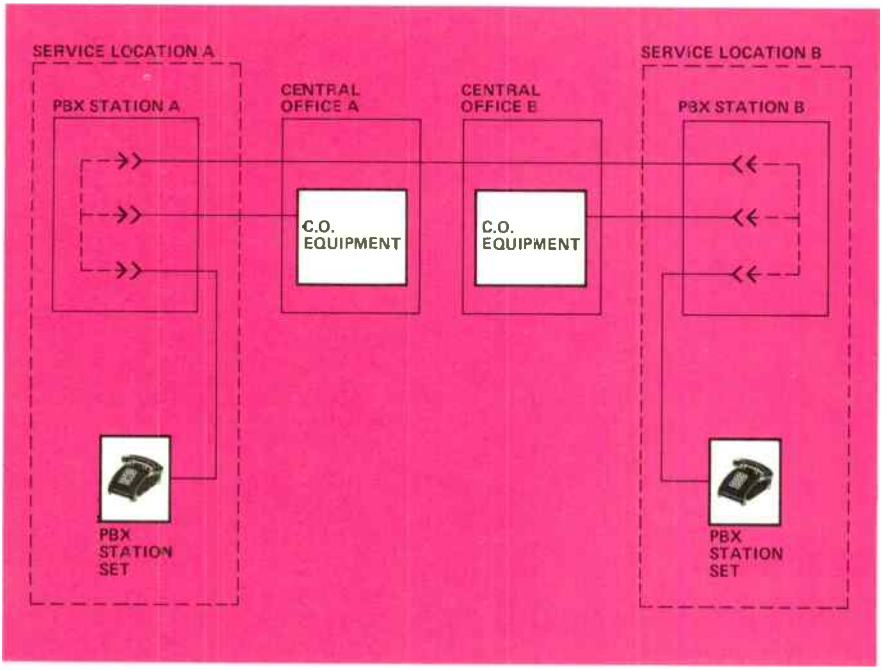


Figure 5. A tie trunk links two PBX systems, and may also provide access to the regular switched networks through the central office equipment.

service designed to allow dialed connection between scattered subscriber PBX switching systems. As shown in Figure 6, each PBX system is connected by dial tie lines to a CCSA switching machine in its local central office. Thus, a user in one location can dial an access code, receive dial tone from the nearest CCSA switching machine, and dial a 7-digit code to reach any PBX station in the network. The first three digits are used by the machine to determine which PBX location is being addressed; the last four digits are then extended to the selected location, enabling the system there to ring the desired station set.

Just as the direct distance dialing (DDD) network contains all of the voiceband trunks needed to provide toll and exchange message service between central offices, the CCSA network includes the access lines and trunks necessary to connect the PBX stations and systems into the switching network. Any service normally provided by a PBX system on the toll and exchange network can be provided by the CCSA, including both voice and data transmission.

Special Service Equipment

Throughout this discussion, the various special services have been considered in simplified terms, with blocks representing equipment and single lines representing communication channels. In practice, however, implementation of a special service may entail use of a large number of features and considerable amounts of equipment; a foreign exchange link, for

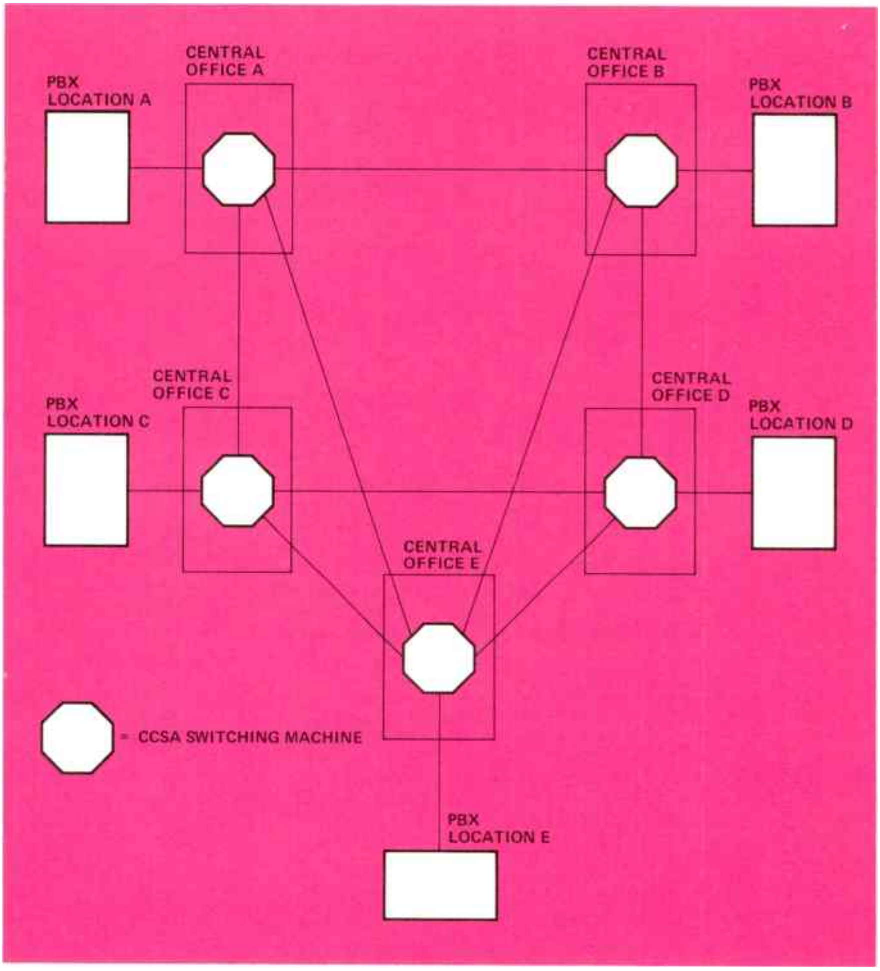


Figure 6. A CCSA network provides interconnection of subscriber PBX systems which are spread among several locations.

example, may actually extend through many central offices, employing several types of carrier and physical facilities in its application.

Traditionally, the individual circuit components of a given transmission facility terminal have been grouped together in a central office area dedicated to that function, and have been connected through distributing frames to the other equipment required for the circuit. This method has a certain amount of flexibility, but also creates problems such as noise and crosstalk aggravation by congested office cabling, and difficulty in maintaining components which are scattered through a large building.

To overcome these problems, plug-in units and direct wiring have come into use, providing all of the circuit features required at a transmission facility terminal in standard shelf arrangements that require no local cross-connection. The end objective of this technique is to eliminate the need for any distributing frame connections except where the circuit leaves and enters the plant building.

Implementation of this "consolidated equipment" concept is achieved by using plug-in units designed to meet the facility interface needs of the various special services; equipment shelves provide a direct-wired position and socket for each special service function, allowing the function to be changed or modified by simply removing a plug-in unit from the socket and inserting a different unit.

This approach has been extended to the point where terminal equipment, such as GTE Lenkurt's 11A Signaling Equipment arranged for private line and special service applications, can be installed on a subscriber's premises rather than in a central office. By placing an equipment housing on the premises and equipping it with appropriate plug-in units, any desired special service can be implemented.

Voice frequency equipment and the services it afford have both evolved considerably since the telephone system first began to extend beyond the capabilities of a simple pair of wires. Because of the progressing state of the communications art and the pressure created by the growing demands of communications users, special services have been developed which can meet virtually any need, whether VF or data.

The multiplicity and overlapping of the various special service applications can be overwhelming if one attempts to view each service individually; looking at the way in which the services interrelate, even in such a cursory manner as has been adopted in this discussion, can put the entire field into its proper perspective.

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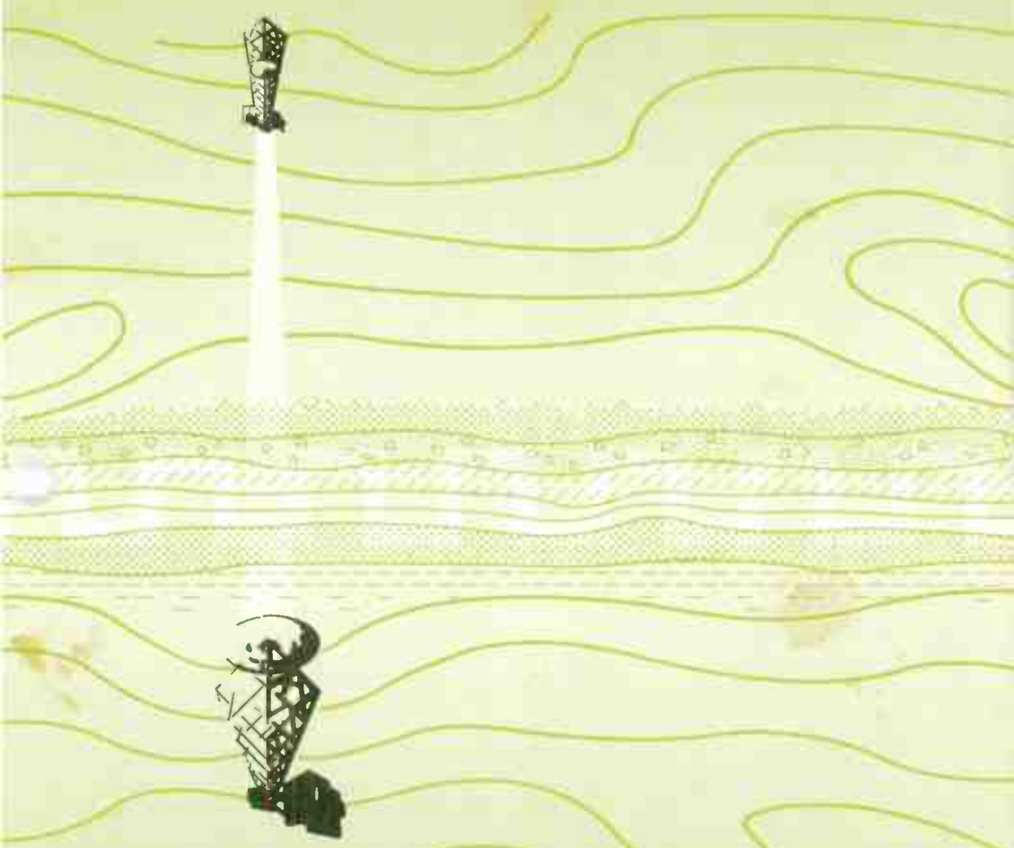


PCM OVER MICROWAVE PART 3

The main impairments to good overall performance of a PCM-over-microwave system are far different from those that degrade either PCM cable carrier or FDM-over-microwave systems.

The first two parts of this series on the transmission of PCM over microwave radio have dealt mainly with the internal operation of a digital multiplexer (the GTE Lenkurt 9120A), which processes the PCM information into an analog signal suitable for transmission over a standard FM microwave radio system. This issue discusses the overall performance of a PCM-over-microwave system, using as a hypothetical example, a low density, 2-GHz system intended for short haul local exchange service.

The performance of a PCM cable system may be degraded by such nongausian electrical events as office noise, lightning hits, switching noise, and crosstalk between cable pairs. FDM-over-microwave is affected by delay and linearity inequalities within the radio system, by thermal or fluctuation noise, by path propagation fades to the noise threshold or mute point of the radio receiver, and by low levels of co-channel rf interference. PCM-over-microwave system performance is influenced primarily by multipath propagation fades that introduce excessive hits (errors) into the PCM bit stream, and by high levels of co-channel rf interference.

System Performance Measurement

The most meaningful measure of performance in a digital system is the bit error rate (BER) at the receiver digital demultiplexer output. The bit error rate is given as the number of bits in error, divided by the total number of bits sent. For example, a BER of 10^{-6} means that one error occurs in every one million transmitted bits.

The bit error rate depends primarily on the value of the signal-to-noise (S/N) ratio at the receiver. The S/N ratio is the difference between signal and noise strength in dB. A S/N ratio of 10 dB, for example, means that the rms value of the signal is 10 dB greater than the rms value of the thermal noise. System performance is determined by measuring signal-to-noise ratio at a particular bit error rate. Generally, the voice quality at the receiving end of a typical PCM system carrying voice traffic begins to deteriorate appreciably when the BER exceeds 10^{-3}, a point at which "cracking and popping" sounds may be heard due to bit errors. A BER of 10^{-6} is an acceptable minimum standard for a PCM system carrying 2400-baud data channels. PCM systems are therefore designed so that under normal conditions, the BER lies well below 10^{-9} most of the time and only reaches a 10^{-6} threshold level for very small percentages of the time. The measured value of BER versus S/N ratio is shown in Figure 1. From the graph, it can be seen that a S/N ratio of 20 dB would correspond to a BER of 10^{-6}.

PCM vs. FDM

In a fade-free environment, typical of short paths or paths that traverse rough, nonreflective terrain in dry or elevated climates, low (near threshold) rf received signal levels may be specified for a PCM-over-microwave link. This is contrary to the engineering requirements of an FDM-over-microwave link, whose signal levels must be quite high even in such a suitable environment to provide the thermal noise quieting required to meet the high quality, low noise specifications of a modern communications system. As in cable systems, however, the BER

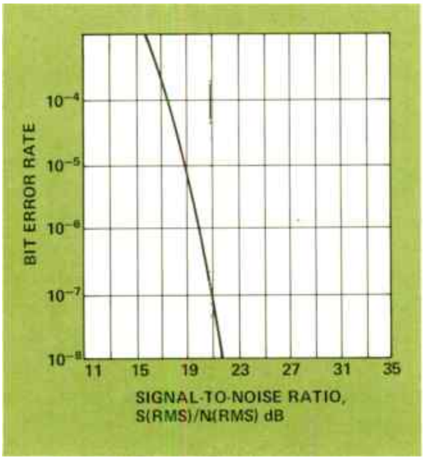


Figure 1. Measured values of BER versus S/N in the receive interface unit of the digital multiplexer.

in a PCM-over-radio system is affected primarily by the introduction of noise spikes or interference pulses into the data stream, which are decoded as legitimate bits of information (bit errors).

In a PCM-over-microwave system, error rates exceeding the 10^{-6} threshold criterion are introduced with a decrease in the signal-to-noise (S/N) ratio either by rf received signal level fading to threshold, or by high level co-channel or in-band rf interference. The S/N ratio related to a 10^{-6} BER assigned threshold, or outage value, is a function of the type of PCM or PCM-to-analog modulation coding (three level duobinary, PSK, QPSK, multilevel PSK, etc.) and the receiver detection techniques (discriminator, coherent detection, etc.) employed. The modulation coding and detection techniques employed in the GTE Lenkurt 2-GHz, PCM-over-microwave system permit the use of a standard analog FM radio system (the 78F2). In a typical BER-vs-S/N characteristic curve such as shown in Figure 1, the S/N ratio is directly related, dB-for-dB, to the rf received signal level and, in PCM radio systems, is of significance only near the radio noise threshold. An unusable BER point is reached only near the receiver threshold, and rf fades to threshold result from changes in the propagation medium (the atmosphere and terrain below it).

Microwave Propagation

The short wavelength of microwaves gives them many of the same properties of light waves; they are therefore refracted or bent by the atmosphere, and are obstructed or reflected by such obstacles as mountains, buildings, bodies of water, and atmospheric layers. While microwaves travel at the speed of light in a vacuum, their speed in air is reduced and varies according to the varying density and moisture content of the air.

Gradual changes in air density may cause the radio wave to refract or bend continuously, so that the beam gradually curves toward the denser atmosphere. Because the atmosphere is increasingly thinner with higher altitudes, radio and, to a lesser degree, light waves, do not follow a straight path but are normally refracted downward. Radio paths extend beyond the visual "line-of-sight" horizon, since radio waves are more significantly affected by all three atmospheric density gradients (pressure, temperature, and humidity) than are light beams. The atmosphere is seldom homogeneous, but may be stratified or constantly changing (as evidenced by the twinkling of stars through what otherwise appears to be a distortion-free atmosphere). These atmospheric irregularities present a varying, nonhomogenous propagation medium to the microwave wavefront, which results in the propagation of not only the main body of energy but also many refracted and reflected secondary rays that arrive at the receive antenna at various phases and amplitudes. The amplitude of the resultant rf received signal—the sum of all of these main and secondary rays—could vary with time from 6 dB above normal to 40 dB or more below normal with signal cancellation. If the fade depth is greater than the fade margin provided in the transmission engineering process, a BER of greater than 10^{-6} is introduced into the PCM channel, resulting in an outage.

Fortunately, most fades of this magnitude result from atmospheric multipath and specular ground reflections, and are of extremely short duration. Only a small percentage of 2-GHz exchange plant PCM microwave paths would ever experience fading severe enough to require adding diversity protection or other special measures. Besides these short-term outages, long-term attenuation fades resulting from partial path obstruction or signal trapping may occur, but usually only in low clearance paths traversing shallow standing bodies of water (swamps, irrigated fields, lakes, etc.). An experienced transmission engineer can usually identify these unusual geographic conditions and route the microwave path over or around the suspected area.

The reliability (or availability) of a 2-GHz microwave path may be approximated with a suitable degree of accuracy from weighted Rayleigh distribution curves as shown in Figures 2A and 2B. These curves show that short 2-GHz microwave links only rarely experience a fade of such depth as to cause an outage, whereas longer paths may be subject to deeper and more frequent fading. The locality is of considerable importance, as shown in Figure 2B: long 2-GHz paths traversing reflective terrain or water in coastal or other highly humid regions will fade far more frequently than long paths over rough terrain in dry regions.

Most applications of the short haul PCM-over-microwave links are configured for single-channel operation, with no need for equipment or propagation redundancy. In longer systems in difficult propagation areas, space diversity may be used. Figure 2B shows that the outage time for a 30-mile path with 35 dB fade margin may be reduced from 48 to less than 2 minutes per year with a 40-foot diversity spacing of the receive antennas.

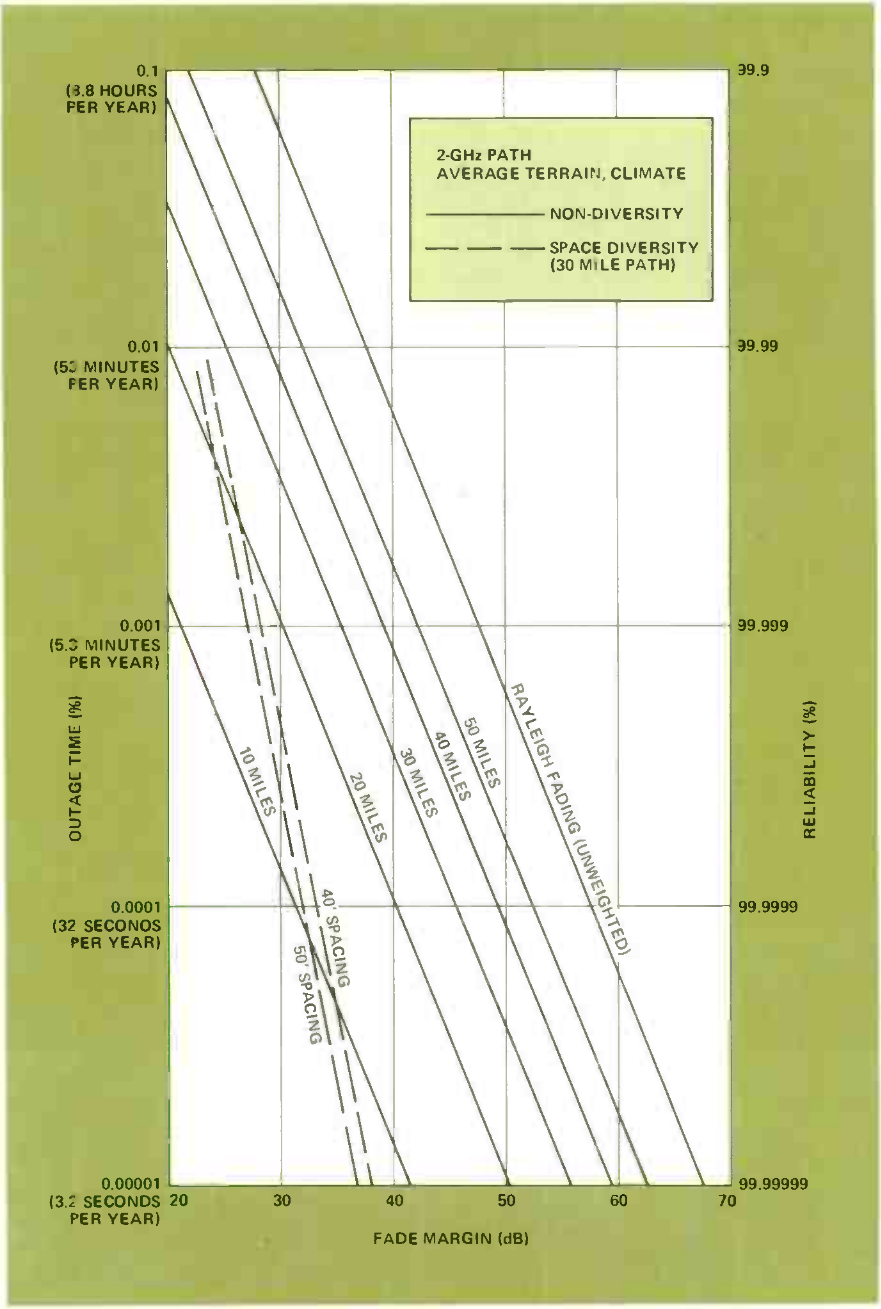


Figure 2A. Outage time due to fading, as a function of path length, can be estimated using weighted Rayleigh distribution curves.

Receiver Noise

In the receiver, the level fluctuations of the received rf signal caused by fading are removed by automatic gain control (agc) circuits before the signal is applied to the demodulator. In most microwave receivers, agc is provided at the intermediate frequency (70 MHz) to which the received signal is converted by the mixer. Thus, the receiver gain varies in accordance with the received signal level, the gain being high when the received signal is faded and low when it is not. Any noise entering the receiver input, as well as noise generated in the input circuit components, is amplified along with the desired signal, so that when the signal fades, the noise is proportionally higher and the signal-to-noise ratio is decreased.

In terrestrial line-of-sight microwave transmission, the limiting noise contributor is the thermal background noise of the warm earth (typically −114 dBm per MHz of receiver bandwidth). Added to this background noise is the receiver front end noise, characterized by the receiver noise figure indicating (in dB) how much more noise is applied to the receiver mixer compared to the −114 dBm/MHz warm earth background. Typical receiver noise figures range from 4 to 12 dB depending on receiver front end design and frequency band. The PCM signal must be 10-20 dB above the thermal noise level (a value determined by PCM coding and detection techniques) for a given threshold error rate of 10^{-6}.

Digital Transmission

A discussion of microwave propagation applies to any type of point-to-point microwave system, whether carrying digital or FDM channels. The information to be transmitted may be carried by the microwave signal in various forms of modulation such as frequency, phase, amplitude, or a combination of these; but, frequency or phase modulation is the preferred method because it facilitates provision of agc, and amplitude linearity is not required in the rf and if circuits.

When the information to be transmitted is in digital form—digital data or PCM voice, for example—the preferred method of modulation of the microwave carrier is still frequency or phase modulation, thereby keeping the envelope of the microwave signal as constant as possible. In this light, it can be seen that FM microwave equipment originally designed to transmit such analog signals as FDM voice or video, should also be suitable for the transmission of digital information. Everything necessary for satisfactory and reliable transmission is already present. All that is required is conversion of the digital signal into a form in which it can take the place of the normal analog modulating signal; that is, make it look similar with respect to level and frequency spectrum.

A typical example of this approach is GTE Lenkurt's 9120A digital multiplexer, which contains facilities to interface two low speed T1 PCM bit streams (24 vf channels each) with an existing FM radio of suitable bandwidth (such as the GTE Lenkurt Type 778F2A microwave radio operating in the 2-GHz common carrier band). A typical arrangement of a 2-GHz, PCM-over-microwave system is shown in Figure 3. At the transmitting end, the 9120A combines two asynchronous DS1 (1.544 Mb/s) bit streams into a single output unipolar digital signal of 3.156 Mb/s. This signal is converted into a "modified duo-

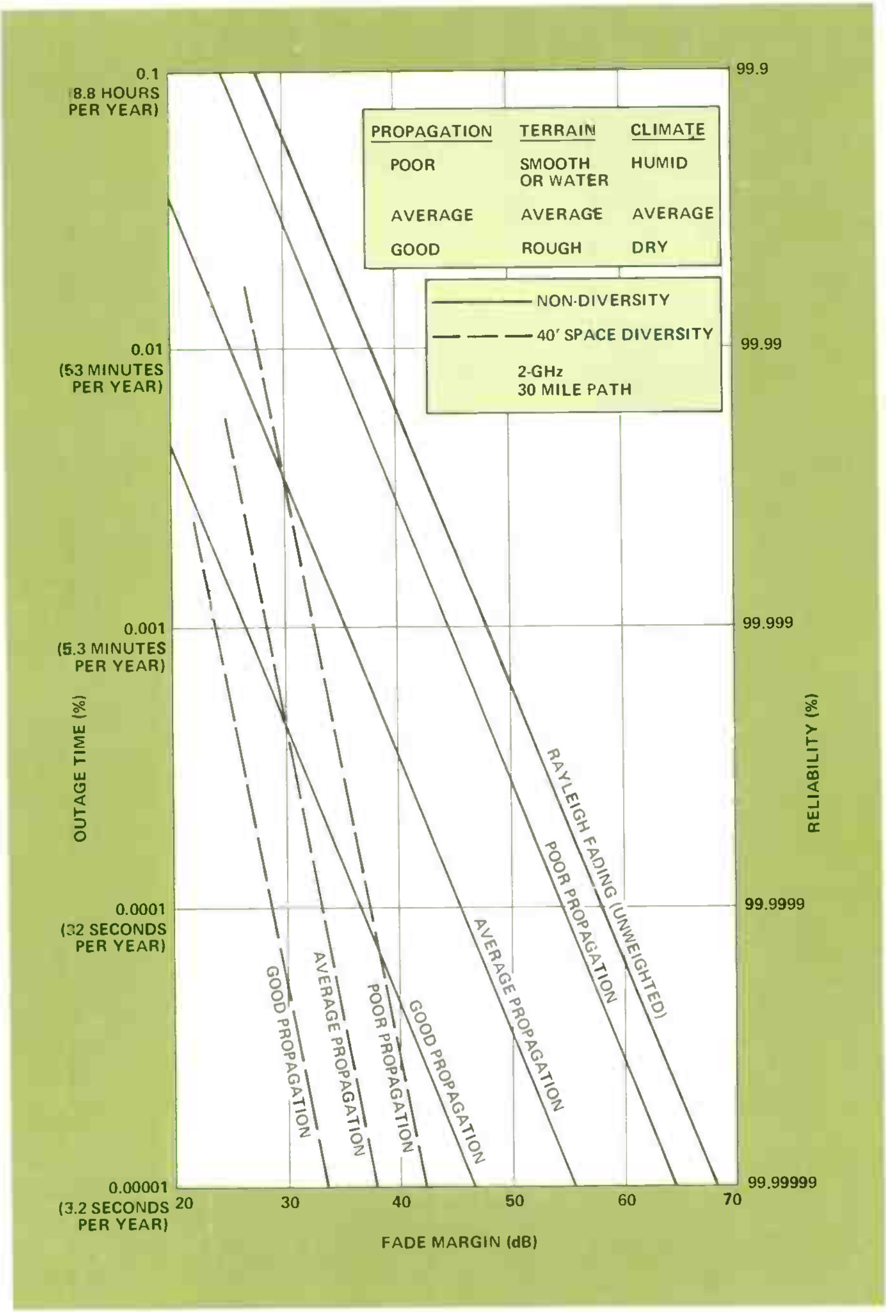


Figure 2B. Climate and terrain are important factors in determining overall microwave system performance.

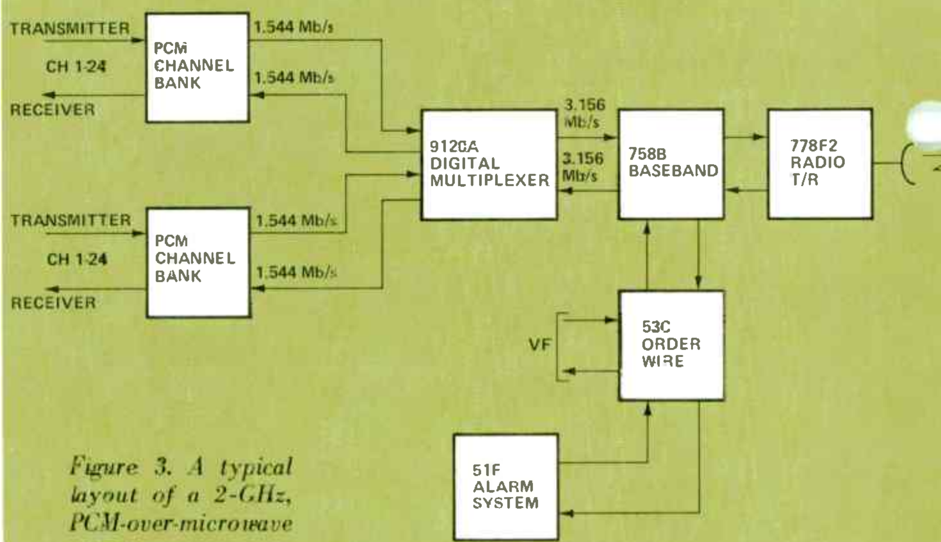


Figure 3. A typical layout of a 2-GHz, PCM-over-microwave system.

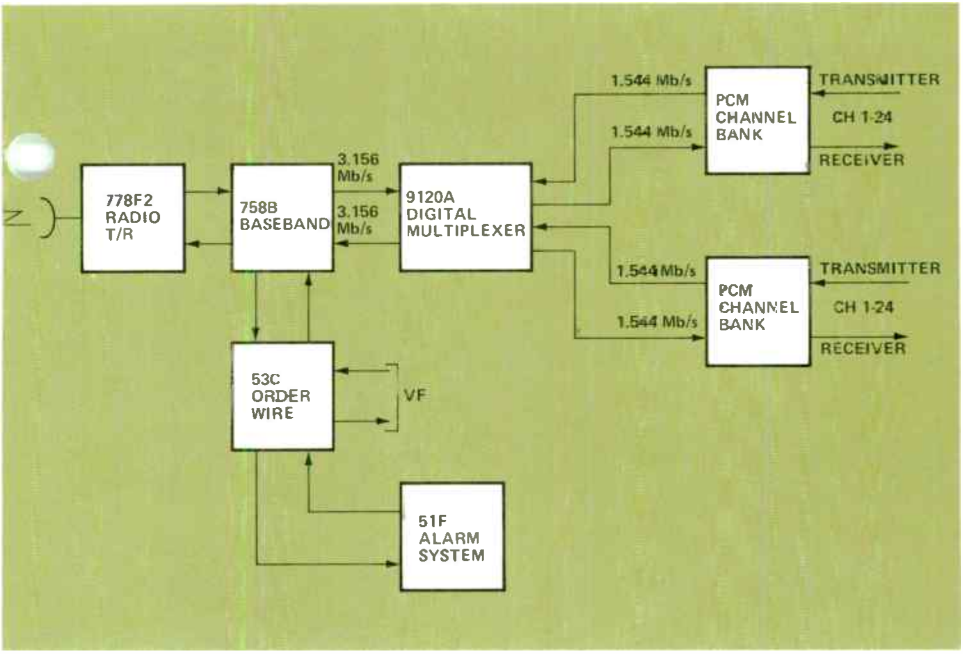
binary" signal in the radio-interface unit and applied to the baseband input of the radio, where it frequency modulates the microwave carrier.

At the receiver, the modulating signal is recovered in the conventional manner by the FM discriminator and then applied to the receive radio interface unit of the multiplexer. The interface unit extracts the clock from the modified duobinary signal and subsequently converts it back into a digital (binary) signal for further processing (de-multiplexing). Finally, the two original 1.544 Mb/s bit streams emerge from the multiplexer for connection to such T1-type equipment as channel banks, data terminals, or repeatered lines.

Fade Margin

Fade margin is one of the important factors that determine microwave system performance. Fade margin is the amount of reserve power in dB available at a receiver to overcome the effects of sudden atmospheric fades. If, for example, the normal receiver input level is -45 dBm, a 30-dB multipath fade will drop the receive level to -75 dBm. If the -75 dBm level corresponds to a S/N ratio in the receive interface unit of 20 dB and a BER of 10^{-6}, then any further increase in the intensity of the fade will exceed the limits of the bit error rate. For the conditions just described, the fade margin is 30 dB.

The fade margin depends on the arrangement of the system. The closer the transmitter is to the receiver, or the more suitable the climate and terrain is for microwave propagation, the smaller the fade margin necessary for required reliability. The minimum rf-transmit output power of the GTE



Lenkurt 778F2A radio transmitter (without hot-standby) is +36 dBm. The net path loss of a system is the total loss inflicted on the signal throughout the microwave path. It includes the loss of the path, the gain of the antennas, the loss in the coax between the antenna and the radio, and anything else in the rf path between the transmitter and receiver antenna ports.

A typical 2-GHz net path loss might be 71 dB if the transmitter and the receiver are approximately 30 miles apart, and 8-foot parabolic antennas are used. Such a loss means that normal receive carrier power will be approximately −35 dBm (+36 dBm − 71 dB = −35 dBm). The fade margin in this example is 40 dB, which means that the signal can drop 40 dB without exceeding 20 dB S/N ratio and a 10^{-6} BER.

The BER of a PCM microwave system is comprised of the total contribution of the digital multiplexer, repeatered line (if used), and the radio. The radio's threshold error contribution varies with the carrier-to-noise (C/N) ratio as well as the signal-to-interference (S/I) ratio at the microwave receiver. The C/N ratio varies with fading, and a 1 dB increase in noise will typically result in a ten-fold increase in errors. A safety or fade margin is required to avoid exceeding the error rate specified in the system reliability objectives more than a certain percentage of the time. The fading characteristics of a particular microwave path, the required propagation reliability, and the use or omission of diversity protection, determine the necessary fade margin.

The detected signal level is held constant by agc and amplitude limiting

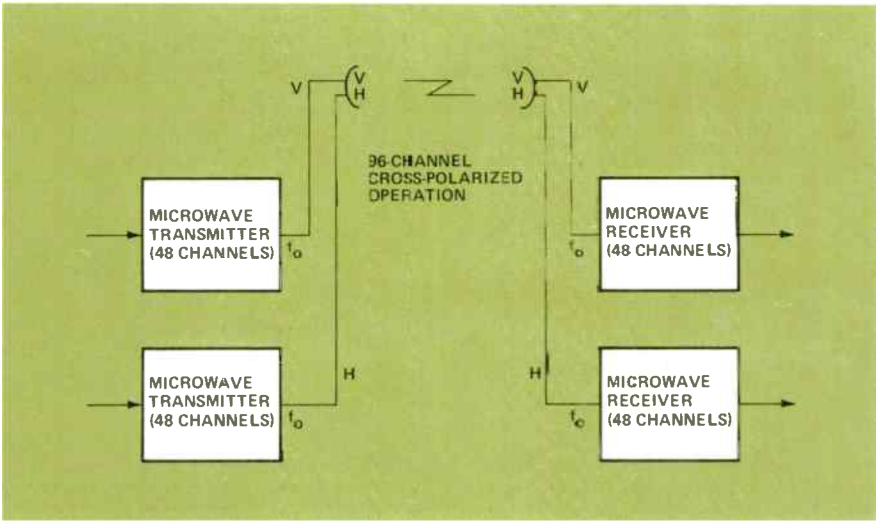


Figure 4. The use of cross-polarization can double the channel capacity of a digital transmission system (Figure shows one direction of transmission only.)

in the if section of the radio, and the noise level may vary over a wide range as a result of microwave signal fading. Over the normal operating range of rf input levels, the error rate is very low. In contrast to a conventional FDM system, noise under unfaded conditions or during moderate fades does not significantly degrade PCM system performance. The digital system is far more tolerant of interference, with error rate being significantly affected only when the power of the interfering signal approaches that of the noise at the error threshold. And, unlike frequency-modulated radio systems carrying FDM channels, the effect of interference on the digital signal is largely independent of the spectral characteristics of the interfering signal.

Cross-Polarization

Because PCM transmission over microwave radio is less sensitive to interference than is FDM transmission, both horizontal and vertical polarization of the same radio frequency can be used for two independent PCM systems over the same path in most propagation conditions, thereby doubling the route capacity. Typically, the value of XPD (cross-polarized discrimination) of a carefully aligned antenna system provides a 25- to 30-dB polarization advantage on a single hop. This means that the vertically polarized signal is attenuated 25 to 30 dB from the horizontally polarized signal (or vice versa). While this separation is unacceptable for transmission of FDM signals, it has an insignificant effect on the transmission of digital signals except near the receiver threshold. Both the horizontal and vertical polarization transmission of a single radio frequency can therefore be used for digital transmission. Thus, two signals from different radios occupying the same rf frequency assignment, but representing two different digital systems, can be transmitted over the same antenna and the same path (see Figure 4). Through the use of cross-polarization, the digital transmission channel capacity over 2-GHz radio using the 9120A digital multiplexer can be increased to 96 channels per radio frequency by using two radios.

While the use of multifrequency cross polarization has been an effective means of transmission for many years, it is not without limitations in single-frequency applications. The XPD may diminish during multipath fading, decreasing the S/I ratio to considerably less than the 20 dB that equates to a threshold of 10^{-6} BER. Similar loss of XPD has also been suspected to result from heavy rainfall, which otherwise is not a factor in 2-GHz propagation characteristics.

Frequency modulation (FM) techniques have been consistently used in commercial microwave radio systems for multichannel voice and data transmission. These systems are used in conjunction with the familiar frequency division multiplex (FDM) carrier system. Proven FM radio equipment can now offer an economically attractive alternative for the extension of digital systems over obstructions or into remote areas where cable plant costs are prohibitive. This three-part series has presented some of the many aspects of a new technique now available to meet the ever-expanding needs of the telecommunications industry.

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