Digital Transmission of Analog Signals

The digital transmission of analog information is an old idea which has always had a certain amount of appeal to telecommunication system designers. If a minimum level of signal-to-noise ratio is maintained, then it is possible to operate a digital transmission system almost error free.

It is the intent of this article to provide a brief tutorial on digital telecommunications for personnel not already familiar with this subject. As background material, some modulation and multiplexing techniques will be covered initially. The focus of the article will be a presentation of the two major telecommunication hierarchies found in today's networks and then digital transmission via microwave modulation techniques will be briefly covered.

ANALOG PULSE MODULATION

The continuous variation of the amplitude, width, or position of the pulses in a pulse train to represent an information signal is defined as analog pulse modulation (examples are shown in Figure 1). The examples are, respectively, pulse amplitude modulation (PAM), pulse width modulation (PWM) and pulse position modulation (PPM). These modulation formats were among the first of the pulse techniques to be utilized, and can still be found in existing telecommunication and telemetry networks. But, more significant, is that PAM is utilized as the first step in many digital pulse modulation formats.

Digital transmission is performed by discrete bundles of energy called pulses. The foundation for this concept arises out of the work done in sampling theory by Henry Nyquist of the Bell Telephone Laboratories. Briefly, Nyquist explained that if a signal, f(t), was bandlimited to B Hertz, then the signal could only change at a maximum rate of B Hertz. He further showed that if a sample of the band-limited signal is taken every $1/(2B)$ seconds, no information is lost. The minimum
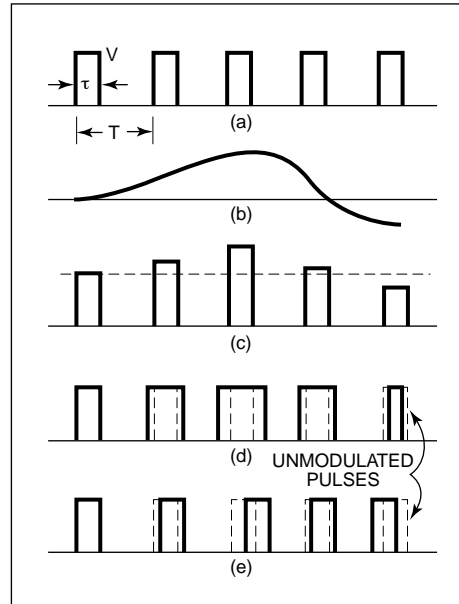

Figure 1. Pulse modulation format. The carrier pulse train "a" can represent the analog signal "b" by continuous variation of the pulse amplitude "c," the pulse width "d," or the pulse position "e."

sampling rate of 2B samples per second is called the Nyquist sampling rate and $1/(2B)$ is called the Nyquist sampling interval. Figure 2 illustrates the sampling process.

Analog pulse modulation is attractive for many data-handling applications due to the ease with which it can be implemented. Analog pulse modulation is also attractive because some modulation formats like PWM and PPM show a signal-to-noise improve-
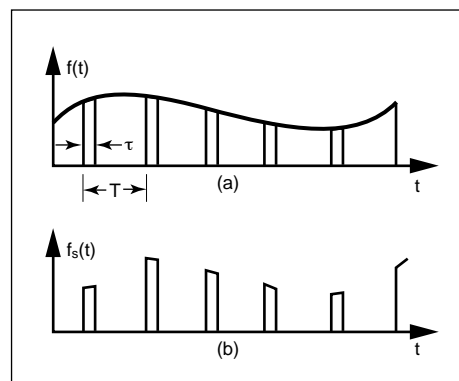

Figure 2. The sampling process. The analog signal f_t is sampled every T seconds for a time of τ seconds (a), which generates the PAM signal (b).

ment that is similar to the improvement found when wideband FM systems are compared to AM systems.

DIGITAL PULSE MODULATION

Pulse code modulation (PCM) and delta modulation (DM) are the major digital pulse formats. Digital pulse modulation is characterized by the representation of the information signal as a discrete value in a finite set of values. Pulse code modulation begins with a sampled information signal (PAM) whose sample amplitudes are quantized and encoded into a finite number of bits or into an n-bit word. The implementation of PCM is more complicated than analog pulse modulation formats, but PCM's transmission and regeneration capabilities are more attractive. The sequence leading from an information signal to a PCM word is depicted in Figure 3. The transmission and regeneration capabilities of PCM lie in the way in which information is carried in the presence (or absence) of pulses, and not in the amplitude of the pulse or the location of the edges of the pulse. Analog pulse modulation formats can withstand only a limited number of repeaters in a noisy system, while the PCM format is generally immune to system noise as long as the repeaters or regenerators are properly spaced. The price paid for transmitting information almost noise-free lies in the quantization distortion generated and the larger system bandwidths required (as in wideband FM).

Delta modulation is a digital pulse modulation technique which has found widespread acceptance in the military sector because of the requirements for low bit-rate digital systems. Delta modulation is differential in nature; it transmits a signal which is related to the difference between successive signal values, as opposed to the actual value of the information signal at a given time. The difference signal, $V_d(t)$ (shown in Figure 4), is developed by comparing the value of the
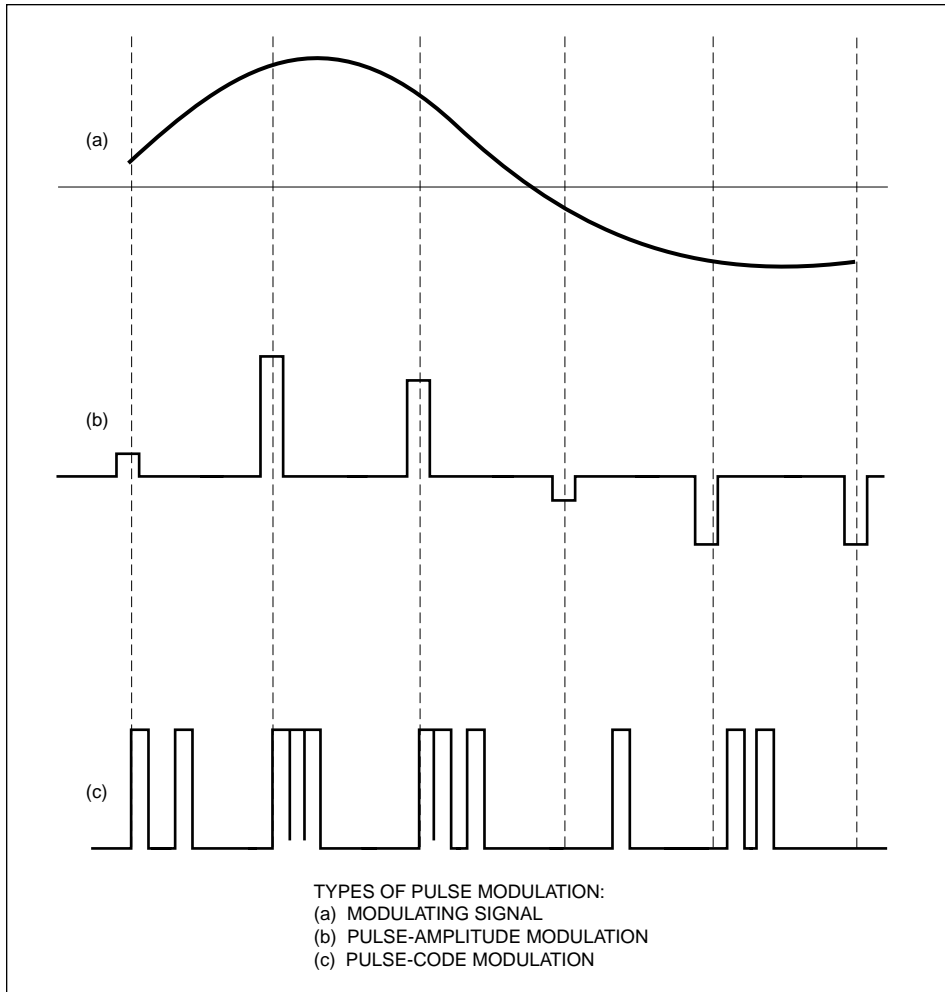
Figure 3. PCM word generation. The modulated signal (a) is sampled and produces a PAM signal (b). Each PAM pulse is then coded into a PCM word (c).

input, $V_{in}(t)$, with the estimate, $V_i(t)$, of the input. The polarity of the difference signal at the sampling instant determines the polarity of the pulse transmitted, $V_{out}(t)$. The difference signal is a measure of the slope of the signal.

A delta modulator is shown in block diagram form in Figure 4. The pulse generator furnishes a regularly recurring train of pulses, $V_p(t)$, of fixed amplitude and polarity. To simplify the illustration, the pulses are assumed to be impulses. The operation of the modulator may be seen using the waveforms of Figure 4. In this figure, t = 0 has been selected to occur at a pulse occurrence. The initial value of $V_{in}(t)$ is 2 units and the initial value of $V_i(t)$ is assumed to be 0. The polarity of the difference signal is positive, and a positive impulse is transmitted. At the next sampling instant, T, the impulse is positive. At the sampling instant, 2T, the impulse is negative because the integrated pulse is 2 units and the value of $V_{in}(2T)$ is less than 2 units, which results in a negative polarity for the difference signal. Examination of the waveforms of Figure 4 reveals two points: **(1) In the presence of zero slope, the modulator's output is a pulse train of alternating polarity.** Demodulation is accomplished by integration and low-pass filtering. The pulse train of alternating polarity will integrate to its average value. If no signal is put in, then the pulse train will have an average value of zero. **(2) The modulator cannot respond quickly to signals whose amplitudes change by more than a unit step from one sampling instant to the next.** This is

evident in the first two intervals of the waveforms.

The delta modulators in use today change their step size in response to the steepness of the slope so that they can respond quickly to large dynamic changes in signal amplitude. Another attractive feature of the delta modulator is that the circuitry is easier to implement than PCM circuitry.

INCREASED UTILIZATION THROUGH MULTIPLEXING

One method for increasing the utilization of a communication channel is by sharing the channel among two or more signals. Sharing may be done on either a phase, frequency, or time basis, or combination thereof. Multiplexing is the sharing of a communications path among signals, and in telecommunications, the more commonly found multiplexing systems are implemented on either a frequency or a time basis.

The sharing of a telecommunication channel on a frequency basis is called frequency division multiplex (FDM). Each signal is nominally 3 kHz in bandwidth and, in FDM, each signal is translated in frequency to any one of many pre-assigned spectral locations. Translation is accomplished using single-sideband modulation (SSB) techniques because SSB format allows the signal to be transmitted without a redundant signal (sideband) or a carrier. The first level of FDM will create a composite signal containing twelve channels that are designated as a group. Before transmission, groups are usually combined, using FDM principles, into larger aggregates called super groups, master groups, and super master groups. At the receiving end, demultiplexing is aided by the hierarchical structure and the presence of single-tone signals called pilot tones.

Time division multiplex (TDM) increases the utilization of a telecommunication transmission channel by sharing that channel among many signals on a time basis. If the
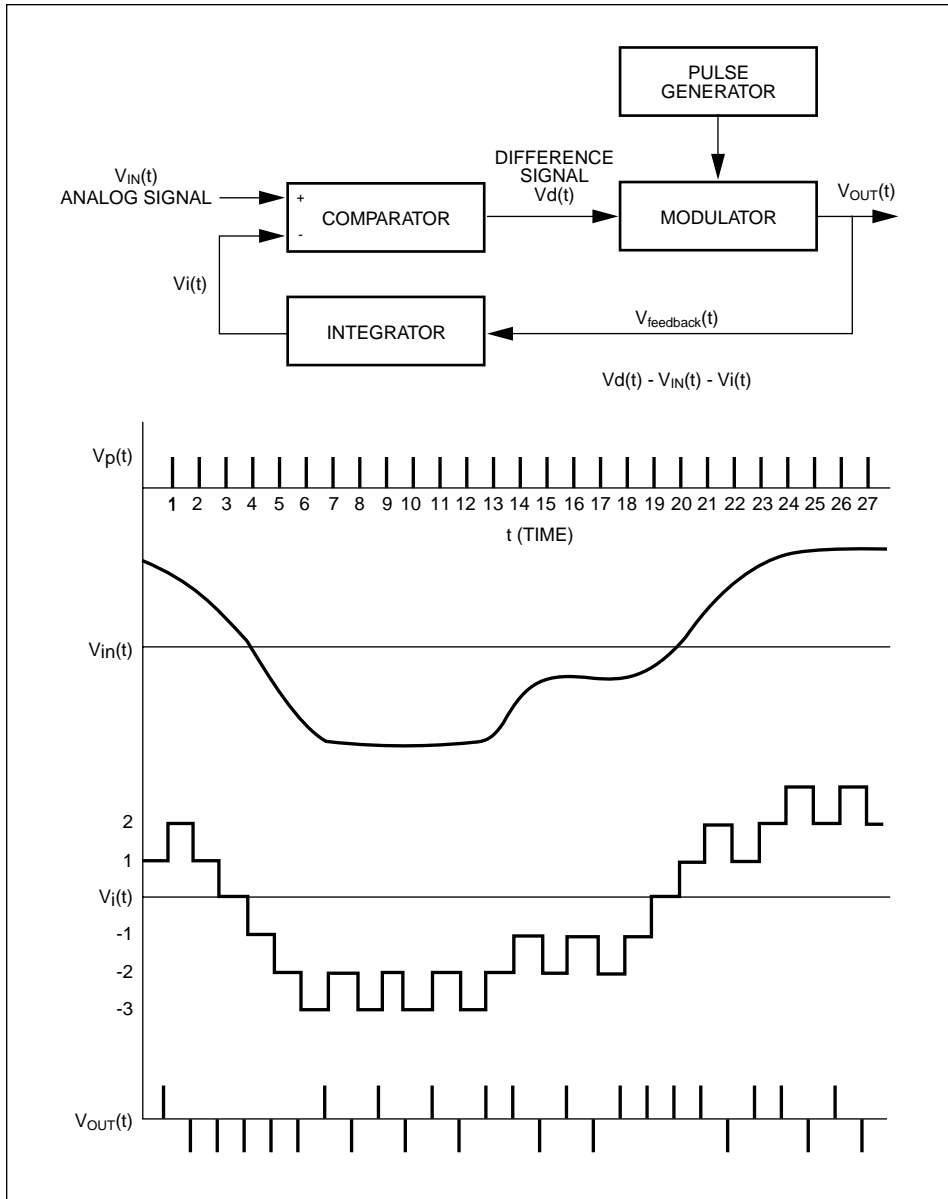
Figure 4. Block diagram of delta modulation.

Figure 5. The sampling process. The duration τ is a portion of the sampling interval (1/2B).

duration of the sampled signal in Figure 5 is allowed to decrease, then the bandwidth necessary for distortionless pulse transmission has to increase correspondingly. The top signal. $f_1(t)$, in Figure 6 will represent the signal of Figure 5. This signal f_1t is a pulse train where the ratio, called duty cycle, of pulse-duration-to-sample-interval is low. If more pulses of the same pulse duration were transmitted in the empty spaces of a low duty-cycle signal, then the transmission channel utilization would be increased for that sample interval. The utilization increase would occur without a significant increase needed in the bandwidth required for distortionless transmission of a low duty-cycle pulse signal. The two adjacent sample intervals of Figure 6 represent the composite signal presented to the transmission channel when sampled signals from three additional data sources, $(f_2(t) - f_4(t))$, are multiplexed with the top sampled signal. The interleaving. on a time basis, of signals from many sources into a composite signal for the purpose of increasing the utilization of a transmission medium without significantly increasing the band-
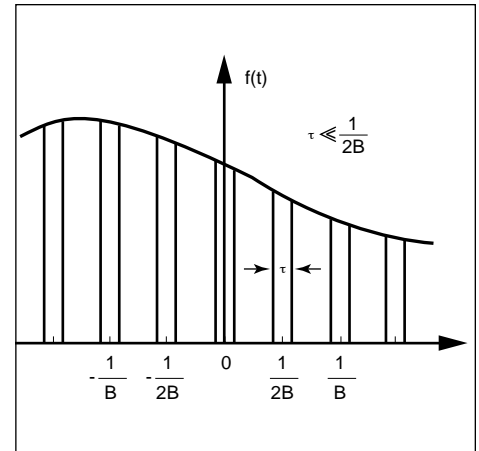
width requirements forms the basis for the use of TDM in telecommunications.

Three of the consequences of allowing the pulse duration to decrease are: (1) Wider transmission bandwidths; (2) Due to transmission bandwidths not being infinitely wide, pulse spreading occurs, which requires that new pulses be regenerated regularly; (3) Pulses must stay within certain time windows, which requires additional circuitry for synchronization purposes. The data source signals of Figure 6 represent sampled signals which, for illustrative purposes, have been sampled at the same rate, and the sampling circuitry for each signal is synchronized to a master data-source clock. The data-source commutator of Figure 6, which is an electronic switch that does the multiplexing, is also synchronized for $f_2(t)$ from data-source two, to arrive at data-sink two. In the simplest case, the master data-source clock is transmitted to the data-sink area through a separate, dedicated channel. The more common and more devastating failure modes in TDM systems are related to the loss of synchronization between transmitting and receiving ends.

PULSE CODE MODULATION

Pulse code modulation consists of three processes, the first of which is the sampling of a bandlimited information signal to create a PAM signal. Voice signals are generally band-
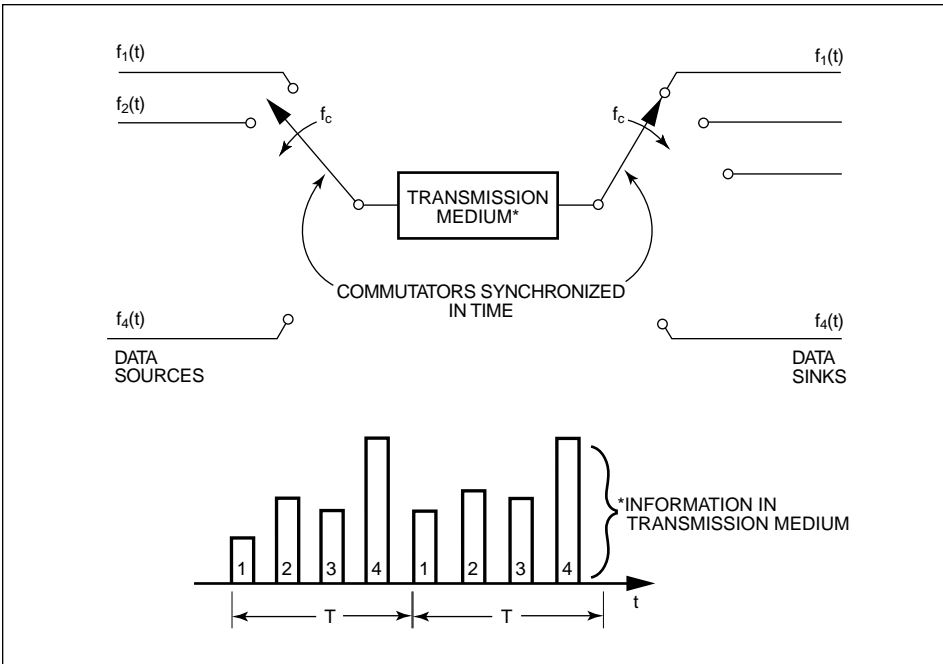
Figure 6. Time division multiplexing between four data sources and four data sinks.
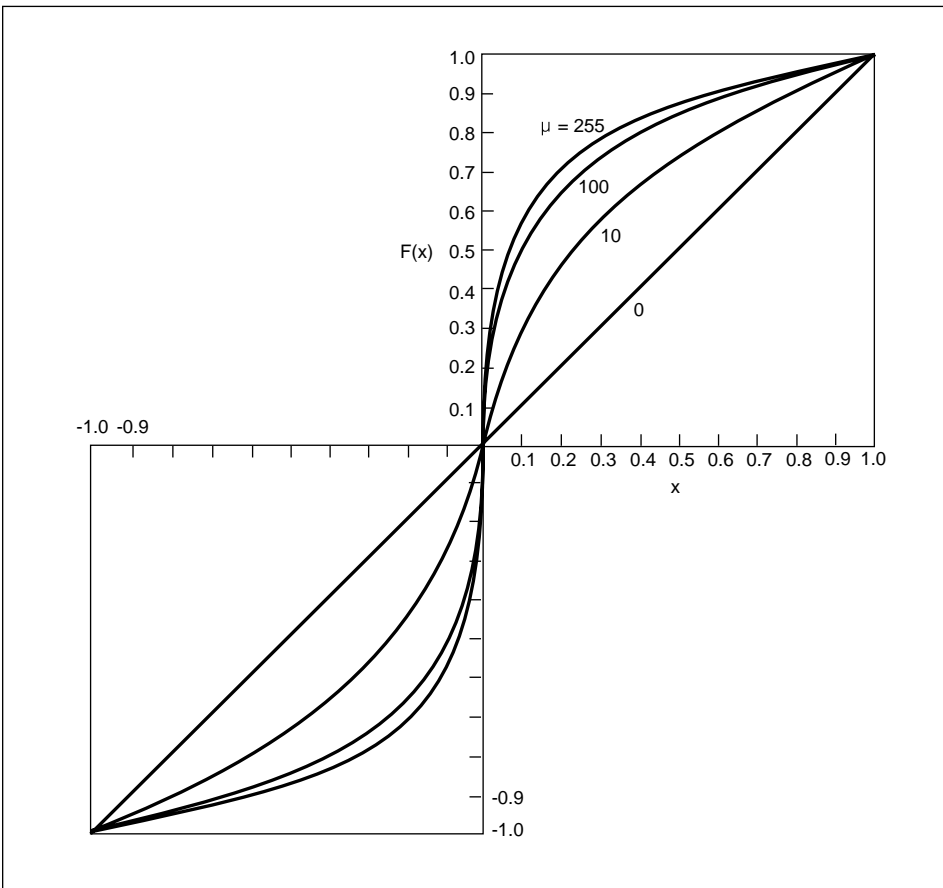

Figure 7. μ-law compander characteristic. $f(x) = \log(1 + \mu x) / \log(1 + \mu)$, μ = 100 for early Bell Telephone system PCM encoders.

limited to 3300 Hz, and are sampled at a rate of 8000 samples per second. The result of the first process is a PAM signal with a sampling interval of 125 microseconds. The second process transforms the analog PAM signal into a digital PAM signal by quantizing the amplitude samples. The third process is a coding process. The quantized sample is transformed into a digital word, which is a word in a finite group of words. There will be 2^n quantized values in an n-bit word system; that is, 128 values for a 7-bit word and 256 values for an 8-bit word.

Quantization can introduce an error into the process of up to half a quantum step between the actual signal and the quantized value. Since the receive terminal of a PCM system can only interpret the received binary word as the corresponding quantized value, a discrepancy will occur between the original magnitude of the sample and its reconstructed value.

These discrepancies appear as noise which is superimposed on the received signal. This noise is known as quantizing distortion, and is a significant source of distortion in PCM. The maximum possible error will be the same for all sample magnitudes in a system that utilizes equally spaced quantization levels. The signal-to-quantizing distortion level will be worse for low-level signals than for high-level signals. If the spacing of the quantization levels is made to vary with the signal level; that is, small spacing, for low-level signals and larger spacing for high-level signals, then the ratio of signal-to-distortion can be made to approach a constant for all signals within the input range. Voice signals have a dynamic range of 40 dB, and it is important to have a constant signal-to-distortion ratio for low-level signals as well as high-level signals. A logarithmic coding law is used in the majority of systems; its main attribute is that the ratio of signal-to-quantizing noise is constant. Figure 7 features an example of the logarithmic law used in early Bell Telephone System PCM coders. The logarithmic-based coder has the effect of compressing the signal at the

coder input and expanding that signal by the same amount in the decoding process. This compression-expansion process is known as companding, and has been used in voice transmission systems to improve the signal-to-noise ratio of low-level voice signals. The early PCM systems used an instantaneous compander followed by a linear coder to implement a continuous function corresponding to a logarithmic law. Due to implementation problems, the majority of new PCM systems use nonlinear coding, which gives a segmented approximation to a logarithmic coding law.

PCM CHANNEL BANKS

The first commercial system for digital telecommunications was composed of an equipment assembly called the D1 channel bank, which terminated 24 audio channels and transmitted a digital signal onto the T1 repeater transmission line. The D1 channel bank and the T1 repeater transmission line were introduced by the Bell System in 1962.

Channel banks primarily perform a multiplexing function. Channel banks used for FDM purposes take unmultiplexed signals and multiplex them, on a frequency basis, into a composite FDM signal. Channel banks used for PCM take voice-frequency signals and create a time division multiplexed digital signal. The D1 channel bank samples 24 voice frequency channels and, using the $\mu = 100$ coding law, generates a 7-bit digital word for each sample. The composite signal is constructed from the 24 digital words which have been interleaved on a word-by-word basis.

The Western Electric Company (WECO), which was the manufacturing arm of the Bell Telephone System, put into production at least three primary channel banks. *Primary* indicates that the channel banks interface to audio channels at one end and generate digital signals at the other end. These digital signals are at the primary level of the hierarchy that their encoding law belongs to. The three WECO channel banks are the D1, D2 and D3 channel banks. The main differences
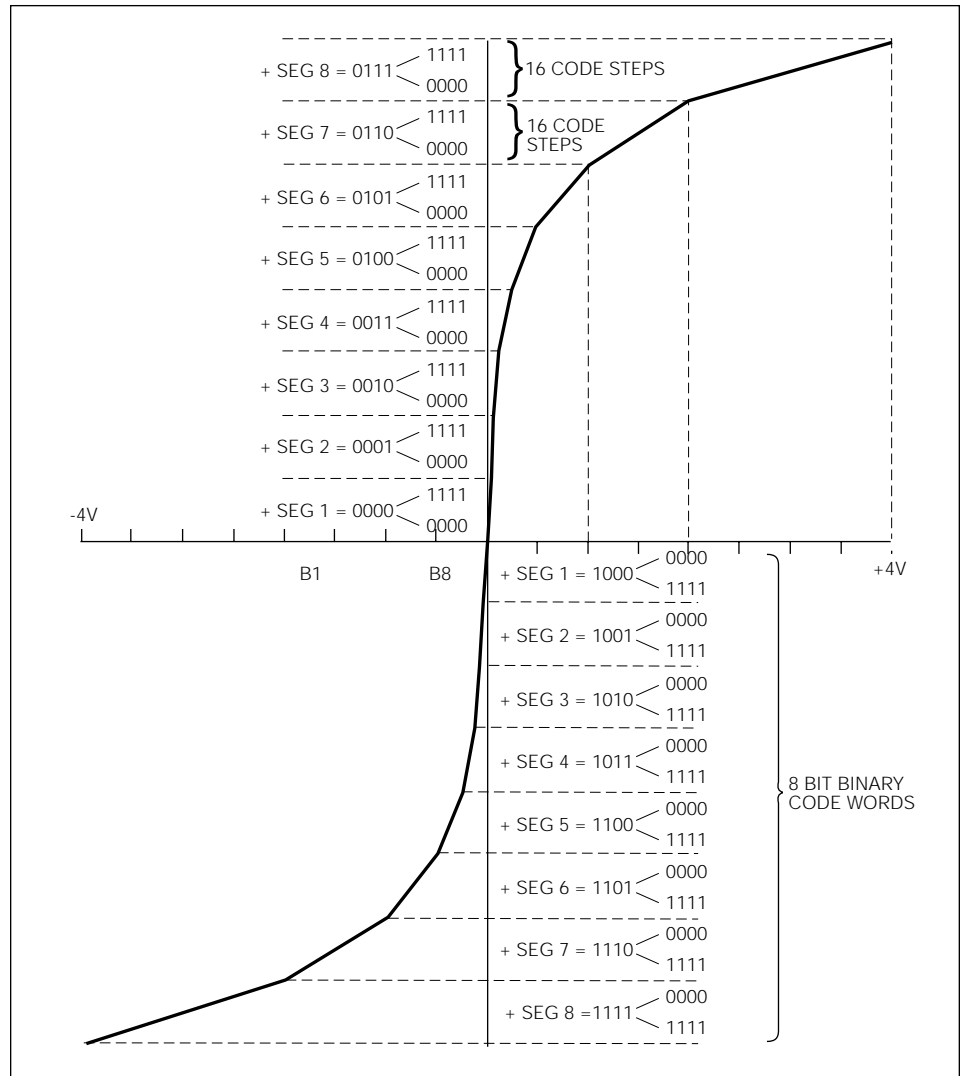

Figure 8. μ-law compander characteristics. A segmented approximation to f(x) = log (1 + μx) / log (1 + μ), μ = 255.

between channel banks are the encoding law used or the number of channels handled, or both.

All three of the WECO channel banks use the encoding law, which is known as the μ-law. D1 uses a value of 100 for μ, while D2 and D3 use a value of 255 for μ. The encoding law for μ = 100 is illustrated in Figure 7 and the μ = 255 law is depicted in Figure 8. The D1 channel bank accepts 24 audio channels and one digital channel. The D2 channel bank handles 96 (4 times 24) audio channels and generates four digital channels. The D3 channel bank has the same number of channels as the D1 channel bank (24 audio channels and one digital channel).

The digital signal assembled and disassembled by the WECO channel banks is called a DS1 signal. This signal is specified by its bit rate and by its nature as a bipolar signal. The multiplexed data stream is modified into a bipolar signal for two reasons: First, to eliminate DC components from the spectrum so that the T1-type line can carry DC power for powering the regenerative repeaters; second, to increase the clock information that is present in the transitions of the marks. *Mark* refers to a signal which represents a logic one and which is about 3 volts in WECO channel banks, and space refers to a logic zero. The bipolar pulse stream uses pulses of

alternating polarity for marks, which increases the number of transitions present in a long succession of marks. Increasing the number of transitions helps the repeaters to recover the clock information so that they can be synchronized to the data stream. Figure 9 shows three waveforms: the top waveform is how the original data stream might appear, and the middle and bottom waveforms show the representation of the data stream as bipolar waveforms with a 100% duty cycle and a 50% duty cycle, respectively. A 100% duty cycle has a mark duration equal to the original mark duration. A 50% duty cycle has a mark duration equal to 50% of the original mark duration. A duty cycle of 50% is used in WECO channel banks because it has the property of centering the power spectrum of the transmitted signal at a frequency equivalent to half the bit rate. Again, in bipolar transmission, the spaces are coded as absence of pulses and the marks are alternately coded as positive and negative pulses, with the alternation taking place at every occurrence of a mark. This mode of transmission is also called alternate mark inversion (AMI).

CCITT

The D3 format used in the WECO D2 and D3 channel banks is one of the two primary PCM multiplex standards adopted in 1972 by the CCITT. CCITT stands for International Telegraph and Telephone Consultative Committee. The CCITT and the International Radio Consultative Committee (CCIR) are two member bodies of the International Telecommunications Union, an agency of the United Nations. **[The CCITT is now known as the International Telecommunication Union or ITU. -Ed.]** These two bodies are charged with recommending regulations for world use in the areas of telecommunications. The standard which utilizes the D3 format is known as the CCITT (-law, and is covered by CCITT Recommendation G.733.
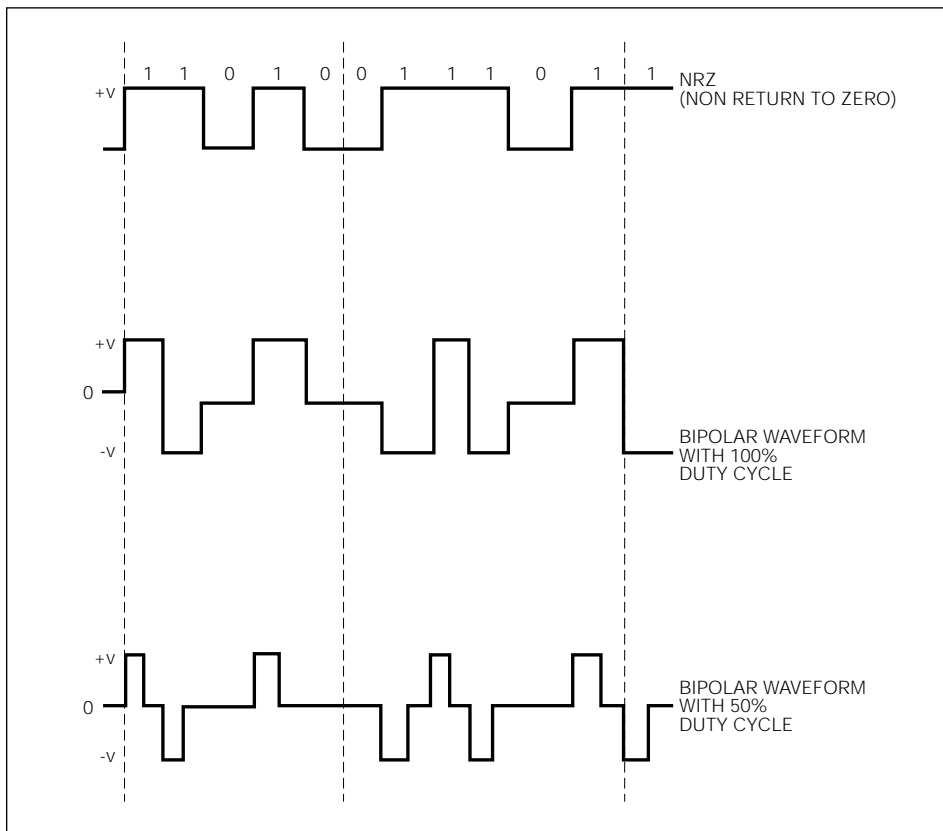

Figure 9. Bipolar waveforms. The NRZ data stream can be represented by bipolar data streams with 100% or 50% duty cycle.

The other primary PCM multiplex standard was the product of the Conference of European Postal and Telecommunications Administrations (CEPT); it is known as the CCITT A-law, and is covered by CCITT Recommendation G.732. These recommendations and the companion recommendation that deals specifically with the coding characteristics, G.711, are in Volume III of the CCITT books of the Sixth Plenary Assembly (1976).

CCITT µ-LAW

The CCITT µ-law has as its basic element a 193-bit frame with a frame duration of 125 microseconds. Figure 10 shows that a frame has 24 channels. It additionally shows that each channel has 8 bits. The number of bits for 24 channels is then 192 (24 x 8); Figure 10 indicates that an additional bit, which is used for frame and multiframe alignment, is inserted before the 192 channel bits for a frame length of 193 bits. The bit rate is referred to as the line rate and is the same - for this frame - as the DS1 signal of the WECO D3 channel bank. The line rate is calculated to be 1,544 kbits/sec by multiplying 193 bits per frame by 1 frame per 125 microseconds. This is also known as the T1 rate.

Each frame carries bits which represent data and bits which are used for signalling information. Each frame has a bit at the beginning which is used for frame synchronization in the odd frames and multiframe alignment in the even frames. Even and odd frames refer to a multiframe structure which arises out of the use of the group of overhead bits known as the A and B signalling bits.

Voice frequency information must be accompanied by signaling information for toll purposes and, to that end, certain frames have been designated as signaling frames. Two signaling bits, the A and B bits, are needed for each channel. The transmission of the signaling bits with the information bits is known as in-band signaling, and it must be implemented in such a way as to minimize its effect on
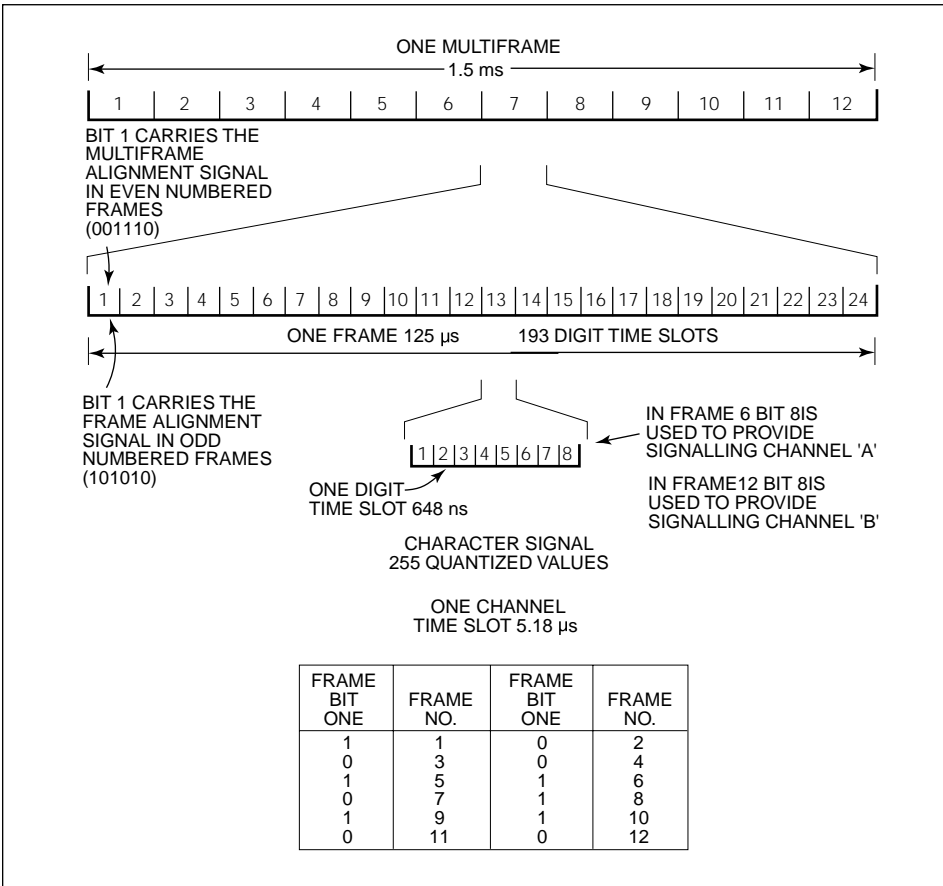
Figure 10. CCITT μ-law multiframe, frame, and channel structures.

the quality of the transmitted data signals. To minimize the impact, a bit is stolen from each channel in only the sixth and twelfth frames of a 12-frame master frame. Figure 10 shows, in the make up of a channel time slot, that there are 8 digits which are numbered from 1 to 8. The most significant bit is 1 and the least significant bit is 8. The stolen bit is the least significant bit which, according to Figure 10, is used for signalling bit A in frame 6 and signalling bit B in frame 12. The encoded law normally used generates 8 bits of data per channel in non-signalling frames 1 to 5 and 7 to 11. Signalling frames 6 and 12 use a different encoding law than that used in non-signalling frames. They utilize a μ = 255 code, which yields 7 bits of resolution instead of 8 bits. Encoding of each of the 24 channels present in frame 6 and in frame 12 puts 7 bits of data into each channel's digits (1 to 7). A voice channel in a 6-frame group has 8 bits encoding for 5 frames and it has 7 bits encoding for 1 frame, which implies that the average coding of a voice channel is 7 and 5/6 bits (8 bits times 5 frames +7 bits times 1 frame, the sum divided by 6 frames).

The μ-law hierarchy has 4 tiers which are specified in terms of voice-frequency channel capacity. The first, or bottom tier, is the primary multiplex, which has a 24-channel capacity. The second-order multiplex carries 96 channels. The third-order multiplex carries 672 channels and the fourth-order multiplex carries 4,032 channels. The 96 channels of the second order are 4 primary multiplex signals (WECO DS1 signals) which have been multiplexed together on a bit-by-bit basis and have a line rate of 6,312 Kbits/s (WECO DS2 signal). The output line rate is slightly more than the combined input line rates because of the need to add housekeeping and synchro-

nization bits. The third-order multiplex is the combination of 7 DS2 signals. The third-order multiplex has a line rate of 44,736 Kbits (WECO DS3 signal). The fourth-order multiplex (WECO D54 signal) has a line rate of 274,176 Kbit/s.

CCITT A-LAW

The A-law arose out of the CEPT; its encoding characteristics can be seen in Figure 11. The reason that there are two incompatible primary PCM multiplex standards is that there were tremendous quantities of equipment employing both of the standards at the time of the formal adoption of the standards, thus hindering the drive toward one world standard.

The CCITT adopted a uniform sampling rate for voice of 8,000 samples per second. Since a frame is defined as the time from one sampling instant to the next, then the frame has the same duration (125 microseconds) as the CCITT μ-law. The frame and multiframe structure for CCITT A-law are shown in Figure 12. The frame is defined as the duration from one sampling instant to the next, and the multi-frame is defined - as is also true for CCITT μ-law - as the time from one signaling bit to the next. The CCITT A-law frame seen in Figure 12 has 32 channels numbered from 0 to 31.

Figure 12 further shows that channel 0 is the frame alignment signal and that channel 16 carries the signaling information. CCITT A-law differs from p-law in that μ-law uses all channels for data and steals bits for signaling information, while A-law uses 30 channels for data (1 to 15 and 17 to 31) and 2 separate dedicated channels (0 and 16) for overhead and signaling. A 16 frame multiframe is defined by Figure 12 because channel 16 is used for multiframe alignment in the first frame (0) in the multiframe, and then it is used for signaling information in frames 1 to 15. Note 2 in Figure 12 points out that the signaling information has 4 bits dedicated to signaling; net number signaling information
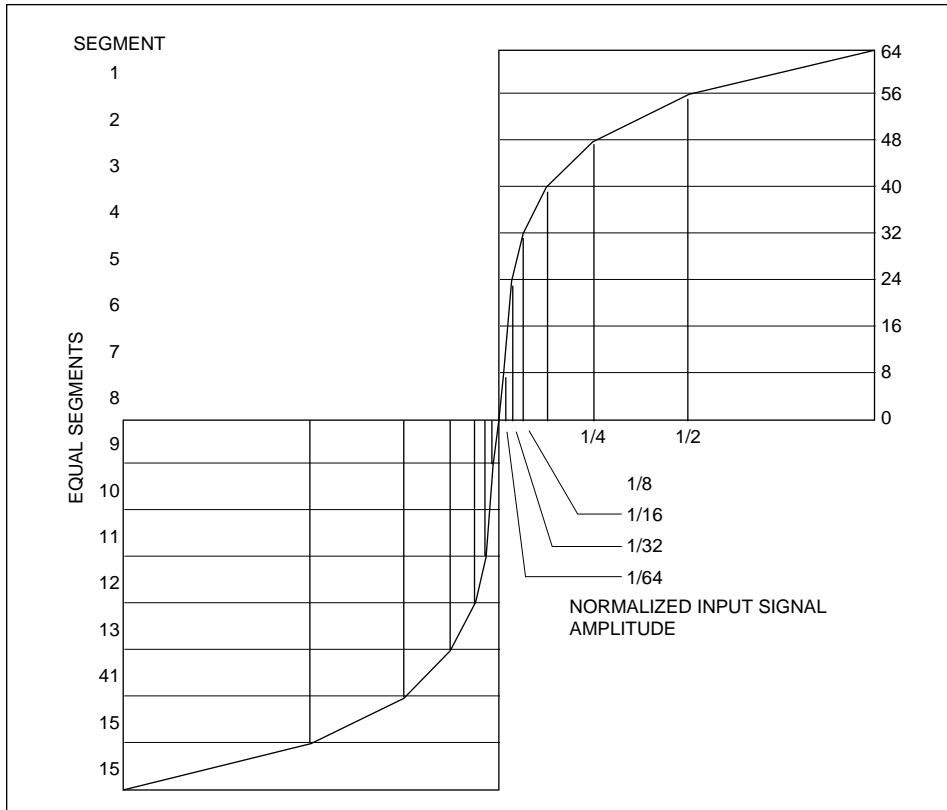
Figure 11. CCITT A-law. A = 87.6, f(x) = 1 + log Ax / 1 + log Ax, 1/A ≤ x ≤ 1. fx = Ax / 1 + log A, 0 ≤ x ≤ 1/A.

will be tied to the frame number. Likewise, channel signaling for channels 16-30 is tied to frames 1 to 15. For example, frame 3 carries the signalling information for both channel 3 and channel 18.

The CCITT A-law has as its basic element a 256-bit frame with a frame duration of 125 microseconds. The line rate is calculated to be 2048 kbit/s, which is 256 bits/frame (32 channels per frame times 8 bits per channel) times 1 frame/125 microseconds. Channel timeslot numbering is shown in Figure 12. Voice channels are numbered 1 to 30. Voice channels which are numbered 1 to 15 occupy channel time slots 1 to 15, and voice channels which are numbered 16 to 30 occupy channel time slots 17 to 31. Channel time slots 0 to 16, as already illustrated, carry signaling and synchronization information.

The tier above the primary PCM multiplex in the A-law hierarchy is the second order multiplex at 8,448 kbit/s; this carries 120 voice frequency channels. The 120 channels are four A-law primary multiplex signals that have been multiplexed on a bit-by-bit basis. Third-order multiplex has a line rate of 34,368 kbit/s and carries 480 channels. Fourth-order multiplex carries 1920 channels and the rate is 139,264 kbit/s.

HIGHER-ORDER MULTIPLEXERS

Higher-order multiplexers are different from primary multiplexers in that they interleave data streams on a bit-by-bit basis, as opposed to the word-by-word basis, and they must take into account bit synchronization. The primary data streams, derived from independent clocks, are not synchronous upon arrival at a higher-order multiplexer. The primary data streams are called

plesiochronous signals because they each have approximately the same bit rate, but they are not derived from the same clock. Since a time-division multiplexer normally scans all of its inputs at the same rate, small clock differences can cause traffic outages. The prevalent method used for synchronization of plesiochronous signals is pulse justification, or "pulse stuffing." Pulse justification allows independent primary and secondary multiplex clocks to exist without causing frame synchronization outages.

Pulse stuffing multiplexers can use positive justification, negative justification, or a combination of both methods. Positive pulse stuffing requires that the outgoing line rate of the multiplexer be higher than the sum of all incoming rates by stuffing in additional pulses. When a pulse is stuffed, that is, added to the data stream, an additional communication channel is used to inform the receiving terminal of the location of the added pulse. Digital switching of TDM signals is not easy to do in the presence of a pulse justification synchronization scheme and, where implemented, is done by breaking the voice frequency channels back down to audio information.

PCM TRANSMISSION

The digital transmission of PCM signals is not as bandwidth-efficient, due to pulse transmission requirements, as is the dominant analog technique, FDM-FM. The digital transmission of PCM signals has the advantage that if a 20 dB signal-to-noise ratio is maintained, the system operates nearly error free.

When PCM signals are digitally transmitted, they are either transmitted over voice frequency (VF) cable pairs or they are transmitted, using modulation techniques, at microwave frequencies. A bipolar pulse stream format (Figure 9) is used with PCM signals transmitted via VF cables to eliminate any DC component from the spectrum of the transmitted data stream and to enhance the operation of the regenerative repeaters. The finite nature of the electromagnetic spectrum requires that
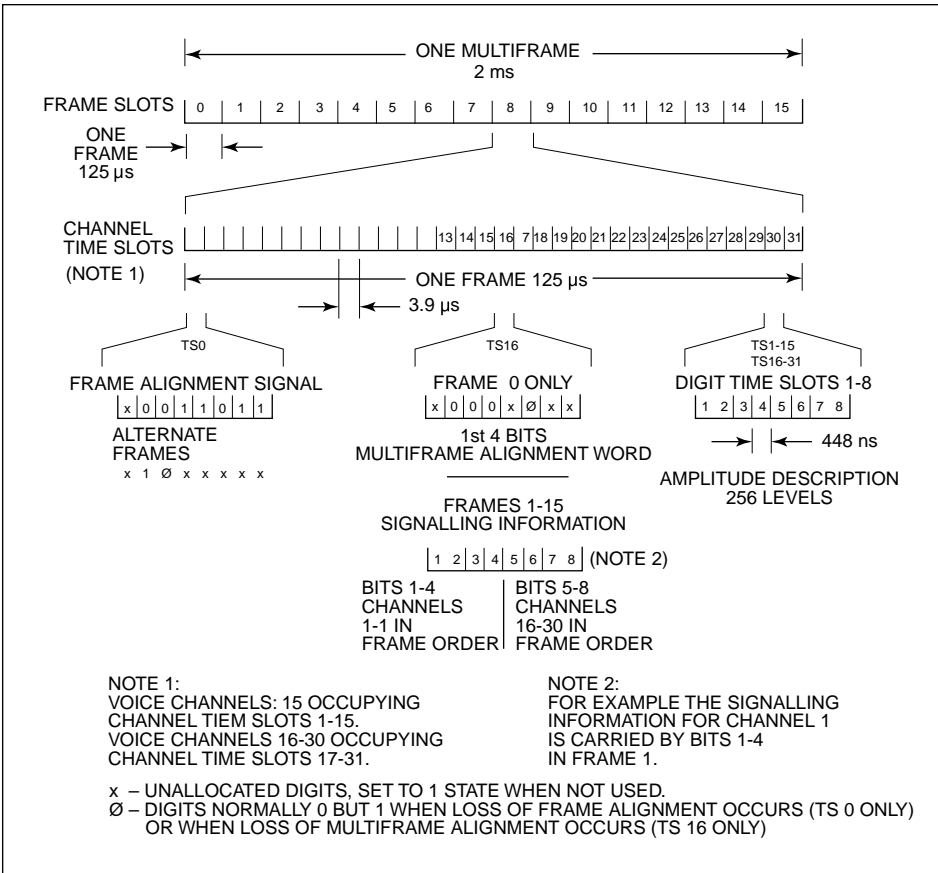
Figure 12. CCITT A-law multiframe, frame, and channel structure

pulses, then the modulation is equivalent to 4-level PSK. If each baseband signal is coded through partial response coding to form a three-level signal (controlled intersymbol interference), then the modulation format is known as quadrature partial response signalling, and the performance is comparable to 8-level PSK.

SUMMARY

The intent of this article has been to sketch out, for those unfamiliar with digital telecommunications, some of the foundational concepts and practices underlying digital telecommunications. Background material included the concept of sampling and the concept of time division multiplex. A good grasp of these concepts enables one to better understand both digital modulation techniques and digital telecommunication hierarchies. The glossary, that follows, contains terms common to PCM hierarchies, and is meant to aid in comprehension of the PCM hierarchies in particular, and digital telecommunications in general.

PCM signals transmitted through the atmosphere be subject to regulations which are aimed at limiting the large amount of transmission bandwidth normally required. The requirement to use digital modulation techniques that increase spectral efficiency are derived from regulations like Docket No. 19311 of the Federal Communications Commission (FCC), which set operating parameters for digital microwave systems. The FCC established a floor of one bit per second per Hertz of authorized RF bandwidth for microwave transmitters operating below 15 GHz. Other countries have similar regulations; one system which has been developed in response, and employed in Canada, achieves an efficiency of 2.25 bits per second per Hertz.

BIT-EFFICIENT MODULATION

The main digital modulation techniques are often used in combination. One of the modulation techniques involves controlled intersymbol interference to achieve a high spectral efficiency, while utilizing realizable filters; this technique is called Partial Response Signalling (PRS). Another technique is Quadrature Amplitude Modulation (QAM). A method that combines both PRS and QAM is Quadrature Partial Response Signalling (QPRS), and is employed by Bell Canada to achieve an efficiency of 2.25 bit per second per Hertz.

The principle of QAM is to independently modulate two carriers of the same frequency whose phase differ by 90μ. The two carriers are known as the I (in-phase) and Q (quadrature) carriers. The original carriers are suppressed in the process, allowing all the necessary information to be carried in the sidebands. If the two baseband signals are discrete two-level

GLOSSARY

Adaptive Delta Modulation This is a method for digitally encoding an analog waveform. A problem with delta modulation is that if the analog signal changes magnitude rapidly, the encoded signal magnitude lags behind the actual signal (slope overload distortion). One solution is to increase the sampling rate, but this faster bit stream is more costly to transmit. Adaptive delta modulation uses a slower sample rate, but when it encounters a rapidly changing analog signal, it increases the size of the quantizing step.

Bipolar Code The bipolar code defines a pulse stream that can be used to transmit digital signals. A binary 1 is represented by a pulse and a binary 0 is represented by the absence of a pulse. In addition, pulses must be of alternating polarity.

Bipolar Violation In a bipolar pulse stream, the pulses are of alternating polarity. A bipolar violation (error) occurs if, after transmission, two pulses of the same polarity are received.

B3ZS (Binary 3 Zero Substitution) A method of achieving zero code suppression in digital signals carried by bipolar pulse streams. Whenever three zeros occur they are replaced by a code that introduces a bipolar violation so that it may be recognized, and removed, after transmission. B6ZS is a similar method of replacing 6 zeros.

Codec An assembly comprising an encoder and a decoder in the same equipment.

Companding The term companding is a contraction of the words COMpressing and exPANDING. Companding was first used in analog transmission systems to reduce the effect of noise. Before input to the transmission system, the dynamic range of an analog signal is compressed. That is, the low-level signals are increased and the high-level signals are reduced.

After transmission, during which noise is accumulated, the signal (plus noise) is expanded back to its original dynamic range.

The signal-to-noise ratio will have been reduced for the lower signal levels and will have increased for the higher signal levels. However, the human ear is more sensitive to noise at low signal levels than at high signal levels. Thus, the companding will have improved the transmission as perceived by the listener.

The companding effect is achieved in PCM systems by effectively using more quantizing levels for the smaller analog signal samples and less for the larger samples. The effect of companding is to reduce the noise perceived by the listener, although in the PCM case, it is quantizing noise and not transmission noise that is present.

Companding Law Many different algorithms (companding laws) could be used to obtain companding effects. Two companding laws in widespread use in PCM systems are the CCITT p-law and the CCITT A-law.

Delta Modulation An encoding scheme that looks at the difference in magnitude between successive analog signal samples. Only one quantizing level is available, so only one bit is needed to record whether the signal went up or down. This system is simple to implement, but requires a faster sampling rate than the Nyquist frequency.

Encoding Law The law defining the relative value of the quantum steps used in quantizing and encoding.

Frame A set of consecutive digit time slots in which the position of each digit time slot can be identified by reference to a frame alignment signal.

Justification A process of changing the rate of a digital signal to change the digit rate from its existing nominal value to a higher predetermined nominal value.

Nyquist Frequency To adequately reproduce a bandwidth-limited analog waveform, a sampling frequency of at least twice the maximum analog waveform frequency must be used. The minimum acceptable sampling frequency (i.e., twice highest waveform frequency) is termed the Nyquist frequency.

PCM Multiplex Equipment Equipment for deriving a single digital signal at a defined digit rate from two or more analog channels by a combination of pulse code modulation and time division multiplexing (multiplexer), and also for carrying out the inverse function (demultiplexer).

Pulse Code Modulation (PCM) A process in which the signal is sampled and the magnitude of each sample with respect to a fixed reference is quantized and converted by coding to a digital signal.

Regeneration. The process of recognizing and reconstructing a digital signal so that the amplitude, waveform and time are constrained within stated limits.

Sampling The process of determining the magnitude and polarity of an analog waveform at a number of points in time. Normally, sampling is done at equally spaced intervals of time.

Segmented Encoding Law An encoding law in which an approximation to smooth law is obtained by a number of linear segments.

Signalling The exchange of electrical information (other than by speech) specifically concerned with the establishment and control of connections, and management, in a communications network.

Zero code Suppression Digital transmission facilities using bipolar code cannot tolerate strings of zeros. Zero's are represented by absence of pulses, and the resulting lack of pulses can cause the transmission equipment to lose its internal synchronization. Zero code suppression limits the number of zeros transmitted.

REFERENCES

1. Bell-Northern Research Limited Systems Education, Course Notes – "Digital Technology In Telecommunication," Bell-Northern Research Limited, Ottawa, Canada, 1977.

2. Bennet, G.H., "PCM and Digital Transmission," Marconi Instruments Limited, White Cresent Press Limited, Luton, England, April1978.

3. Bylanski, P., and D.G.W. Ingram, "Digital Transmission Systems," Institution of Electrical Engineers, Peter Peregrinus Limited, Southgate House, Stevenage, Herts SGI 1HQ England, 1979.

4. CCITT Orange Book, Volume III (3 parts), "Line Transmission," published by International Telecommunications Union, 1977.

5. Freeman, Roger L., "Telecommunication System Engineering," John Wiley & Sons, New York, 1977.

6. George Washington University Continuing Engineering Education Course Notes – "Digital Transmission System Engineering," George Washington University, Washington, D.C., 1980.

7. Gregg, W. David, "Analog and Digital Communication," John Wiley & Sons, New York, 1977.

8. Panther, Philip F., PhD., "Modulation, Noise and Spectrall Analysis," McGraw-Hill Book Company, New York, 1970.

9. Schwartz, Mischa, "Information Transmission, Modulation, and Noise," McGraw-Hill Book Company, New York, 1970.

10. Taub, Herbert and Donald L. Schilling, "Principles of Communication Systems," McGrawHill Book Company, New York, 1971.