T-1/E-1 Technology

Primer

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About This Information

The following topics provide information about this guide:

- Purpose
- Intended Audience
- How to Use This Information
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Purpose

The purpose of this document is to provide fundamental reference information about T-1 and E-1 technology.

Intended Audience

This information is intended for:

- Distributors
- System Integrators
- Toolkit Developers
- Value Added Resellers (VARs)
- Original Equipment Manufacturers (OEMs)

In addition, this information is also intended for Customer Engineering personnel as well as other Dialogic personnel.

How to Use This Information

This information is organized as follows:

- *T-1/E-1 Overview* provides an overview of T-1 and E-1 technology and describes the basics of multiplexing.
- *Pulse Code Modulation* provides detailed information about converting analog voice to digital voice using pulse code modulation.

- *T-1 Technology* describes the fundamentals of T-1 multiplexing, T-1 framing, and T-1 signaling methods .
- *E-1 Technology* describes the fundamentals of E-1 Multiplexing, E-1 framing, and E-1 signaling methods.
- *Interfacing to T-1 or E-1 Service* discusses the aspects of connecting to T-1 or E-1 line.
- *Industry Standards* lists the various standards that apply to T-1 and E-1 networks and provides a brief description of each standard.
- *Intel Terminology* lists Intel terms and definitions as well as the industry terms that apply.
- *Implementation Specifics* provides information about the Intel implementation of T-1 and E-1 technology.

Related Information

For information on Dialogic products, visit our website at *http://www.dialog.com*.

See the following for more information:

- Analog Voice Technology Primer
- IP Technology Primer
- ISDN Technology Primer
- SS7 Technology Primer
- Analog to Digital Voice Solutions Guide

1. T-1/E-1 Overview

The T-1/E-1 overview includes the following major topics:

- Digital Telephony
- Digital Voice
- Basic Time Division Multiplexing
- T-1 and E-1 Applications

1.1. Digital Telephony

T-1 and E-1 are digital telephony schemes provided by communications carriers that multiplex a number of digital voice channels onto a single, higher speed line. T-1 is used primarily in North America and E-1 is used primarily in Europe and Asia. The T-1 or E-1 transmission path is bidirectional and transmits and receives the digital information simultaneously.

Advantages of digital telephony are:

- More efficient because multiple voice channels are multiplexed and transmitted over a common transmission path.
- More economical when compared to the number of equivalent analog lines that would be required.
- More reliable in that repeaters maintain the integrity of the digital signals over long distances.

1.2. Digital Voice

Digital voice is the product of analog to digital conversion. The basic process of converting an analog signal into a digital signal consists of sampling the analog signal at a constant rate, quantizing the sample, and converting each sample into a digital word. Typically, this digital word is an 8-bit byte.

At the receiving end, the process is reversed and an analog signal is recovered based on the digital samples. In essence, the called and calling parties are unable to discern that any manipulations were ever performed on the original analog voice signal.

1.3. Basic Time Division Multiplexing

The process known as time division multiplexing (TDM) is a scheme in which a number of relatively low speed digital signals are scanned and incorporated onto a single higher speed line in a bit or byte-interleaved pattern. The rate of the high speed line is typically the sum of the individual lower speed rates, although it can be faster. The fundamentals of TDM are illustrated in Figure 1, "Basic Time Division Multiplexer".

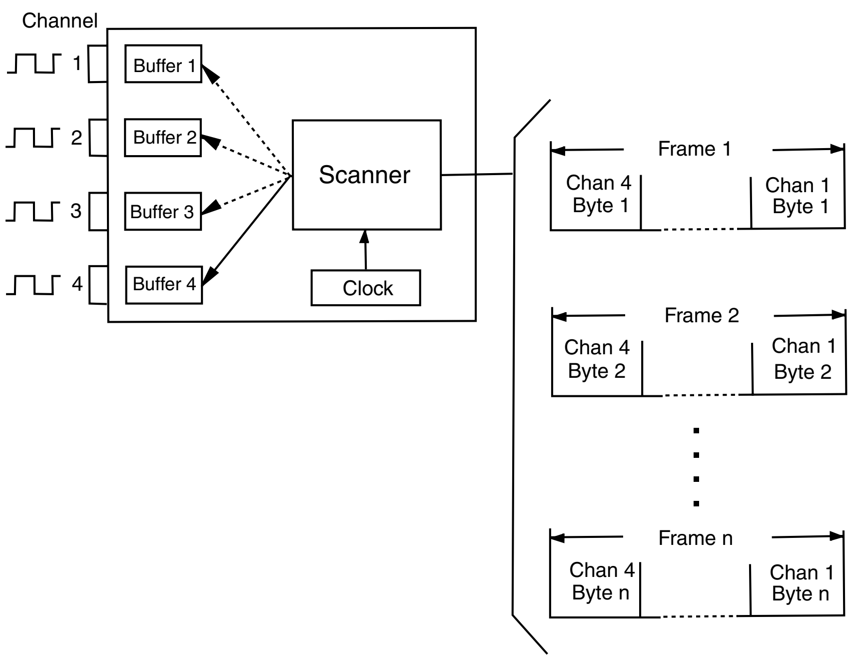

Figure 1. Basic Time Division Multiplexer

In the example, the data from the four low speed channels is applied at the respective inputs to the multiplexer. Each channel's data is buffered to compensate for the difference in speed between the individual channels and the internal scanning rate of the multiplexer, which is determined by the clock frequency. As the multiplexer sequentially scans each buffer, it clocks a byte of data from that buffer and multiplexes the byte onto the high speed output. Thus, the output data from the multiplexer is byte interleaved.

Once the data from the last channel's buffer is read, the scanner immediately returns to the first channel's buffer where it takes the next byte to be multiplexed. One complete cycle of the scanner results in a frame. As a result, the serial data that is output from the multiplexer is a sequence of frames, each frame containing one byte of data from each of the input channels. These data bytes are referred to as time slots within the frames.

At the other end of the high speed line, another multiplexer running at the same clock rate demultiplexes the data from each frame, on a byte-by-byte basis, and directs each byte to the proper channel. In operation, each multiplexer performs both the multiplexing and demultiplexing tasks - multiplexing outgoing data and demultiplexing incoming data.

1.4. T-1 and E-1 Applications

Time division multiplexing is employed in both T-1 and E-1 applications. Also, both schemes multiplex very similar digital voice channels, known as DS-0 channels.

In a T-1 environment, 24 DS-0 digital voice channels are multiplexed onto a high speed line having a data rate of 1.544 megabits per second (Mbps). The T-1 encoding method used to convert analog voice to digital voice is referred to as µ-law. Conveying signaling information such as call set up and dialed digits over the T-1 line is accomplished using a technique known as robbed-bit signaling, where the signaling information associated with each channel is part of that channel's time slot.

In an E-1 environment, 32 DS-0 channels are multiplexed onto a high speed line having a data rate of 2.048 Mbps. The E-1 encoding method used to convert analog voice to digital voice is referred to as A-law. Of the 32 DS-0 channels, 30 are used for digital voice channels and the other two channels are used for signaling and frame synchronization. Unlike T-1, the E-1 line allocates an entire DS-0 channel (one time slot) to conveying signaling information.

2. Pulse Code Modulation

The following topics are discussed in this section:

- Analog to Digital Conversion
- Quantization
- Encoding
- Companding
- Adaptive Differential Pulse Code Modulation

2.1. Analog to Digital Conversion

The process of converting an analog signal to a digital signal is known as A/D (analog to digital) conversion. In digital telephony, the process consists of sampling the continuously varying analog signal at a defined rate and describing the amplitude of the sampled signal using a digital word; in this case, an eight-bit byte. This is known as pulse code modulation (PCM).

Because the amplitude of the voice signal does not change appreciably over a short interval of time between samples, a sample of the signal at any instant is a close representation of the signal. In fact, Nyquist's theorem states that if the analog signal is sampled at a rate of at least twice the highest frequency component of the signal, the samples will include enough information to truly represent the original signal. The frequency range of telephone speech signals varies from approximately 300 to 3300 Hertz. Based on this range, the sampling rate for PCM has been defined as 8000 Hertz (more than twice the bandwidth). This equates to one sample every 125 microseconds. See Figure 2, "Sampling of an Analog Waveform".

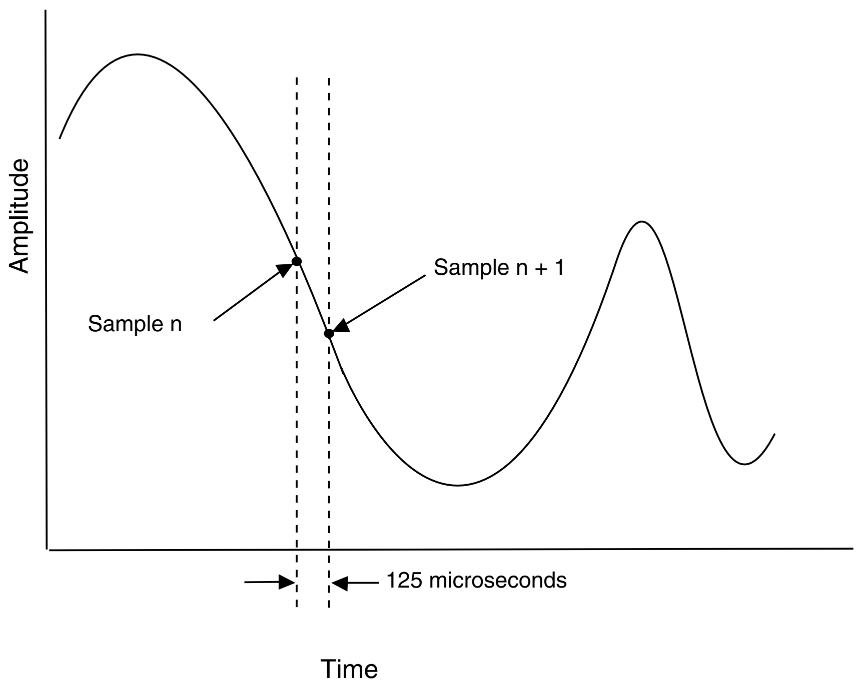

Figure 2. Sampling of an Analog Waveform

Since each analog sampling is described using an eight-bit byte, the resulting digital signal is 64,000 bits per second, or 64 kbps (8 bits x 8000 samples per second).

Because an eight-bit byte is used, the number of discrete digital values used to describe the amplitude of these samples is 2^8, or 256 (0 to 255).

2.2. Quantization

Because the digital signal representing each sample is a discrete value and the signal being sampled is continuously varying, the digital signal is usually a very close approximation of the actual amplitude of the analog signal. Quantization is the method used for assigning a value to a particular sample. In this method, threshold levels are established and values are assigned to the samples based on where the amplitude of the analog signal falls within each of these threshold bands. Therefore, the quantized value of each sample is the closest value to the true value of the analog signal. Figure 3, "Quantization" illustrates the fundamental concept of quantization.

In the figure, the first sampled signal falls into threshold Band 1 even though the actual sample is less than the true value of 1 by an amount, D, the quantization error. This quantization error is not a constant value; it is the difference between the quantized value assigned and the true value. As seen in the figure, the quantized value can be less than or greater than the true value to which it is quantized. This difference adds noise to the signal, called quantization noise. As can be seen in the figure, the noise represents a greater percentage of the signal at lower signal levels (1 and 2) than it does at the higher levels (9 and 10).

2.3. Encoding

After the analog signal sample has been quantized, the next step is to convert the quantized value into a digital value; in this case, an eight-bit byte. This process is known as encoding. Basic encoding produces an output which is linear with respect to the analog input.

In pulse code modulation, it is necessary to quantize both positive and negative signals. Because of this, one of the bits in the eight-bit code must be used to identify the polarity of the signal. This leaves seven bits to represent the value. As a result, the number of quantizing levels is reduced by a power of two, from 256 levels to 128 positive levels and 128 negative levels.

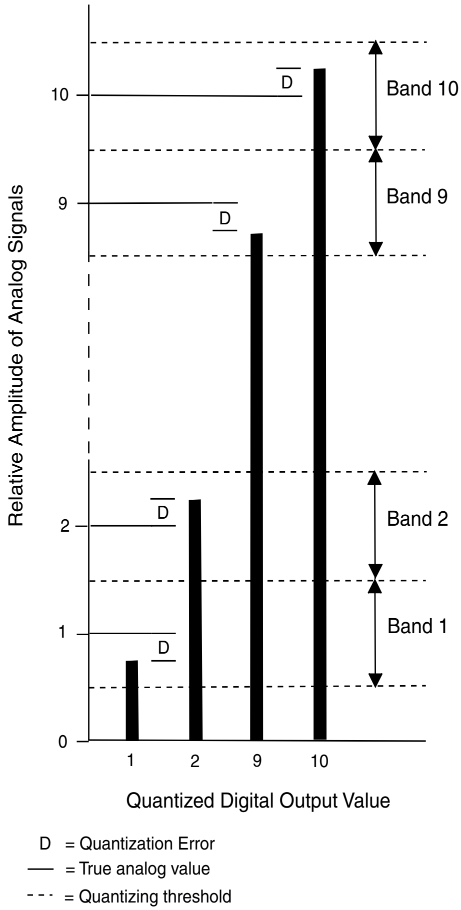

Figure 3. Quantization

2.4. Companding

The problem with linear encoding is that a given amount of quantizing noise represents a greater percentage of noise at low signal levels than at high signal levels. This is expressed as the signal to quantizing noise ratio, or SQR. This condition is undesirable, as small signals are more likely to occur than high ones. To reduce the effects of quantizing noise at low signal levels, the quantizing intervals are adjusted to be smaller (more samples are taken) at low signal levels and larger (fewer samples are taken) at high signal levels. This results in an encoder that produces a nonlinear output which is compressed in relation to the input. At the receiving end, the decoder uses a complimentary expanding characteristic to restore linearity to the signal. The combination of these two characteristics is referred to as a compander and the overall transmission/receiving process of the coder/decoder is called companding.

For T-1 applications, the logarithmic curve is defined by a companding algorithm know as μ-Law. For E-1 applications, the logarithmic curve is defined by a companding algorithm know as A-Law. The two curves are very similar and both reduce the effects of quantization noise by a similar amount.

2.5. Adaptive Differential Pulse Code Modulation

Differential pulse code modulation is a modification of pulse code modulation (PCM). Where PCM uses an eight-bit byte to describe the total amplitude of the sampled analog signal, differential PCM only requires a four-bit byte to describe the *difference* between the actual value of the analog signal and the predicted value of the analog signal. Since the sampling rate is still 8000 Hertz, the bandwidth of the differential PCM signal is 32 Kbps (4 bits x 8000 samples per second). This is possible because voice signals usually vary relatively slowly so that the change in amplitude between any two samples is small.

On those occasions, however, when the change between two successive samples is large, the differential technique is not as precise, and the discrepancy introduces large amounts of quantizing noise and distortion. To address this problem, a technology known as adaptive differential pulse code modulation (ADPCM) is used. This technology employs a sophisticated algorithm that assigns different meanings to the four-bit signal. In essence, the algorithm dynamically increases or decreases the volume range represented by the four-bit signal. If the difference is large, the volume range represented by the four-bit signal is increased; if the difference is small, the range is decreased. Thus, the encoding *adapts* to the changing analog voice signal.

3. T-1 Technology

The following topics provide information about the major aspects of T-1 technology:

- T-1 Multiplexing
- T-1 Framing
- T-1 Signaling
- T-1 Coding Types
- T-1 Alarms
- T-1 Protocol/Switch Types

3.1. T-1 Multiplexing

T-1 is a multiplexing scheme used primarily in North America that allows 24 individual voice channels to be carried on a common transmission medium. This technology was developed by AT&T in 1962 to provide a more efficient means of transporting telephone transmissions over long distances. This section discusses the following topics:

- T-1 Channel Bank
- T-1 Data Rate
- T-1 Transmission
- T-1 Ones Density

3.1.1. T-1 Channel Bank

By converting analog voice signals to digital signals using pulse coded modulation (PCM) and then multiplexing these signals onto a high speed digital line, 24 separate phone calls can be transmitted simultaneously over a single transmission path. Today, T-1 technology is also used for connecting central offices and providing a more efficient service to private enterprise customers who have a high volume of telephone traffic.

The multiplexing device used by the public switched telephone network (PSTN) to provide this service is a channel bank. This device embodies the fundamental principles of T-1 multiplexing as shown in Figure 4, “T-1 Channel Bank”.

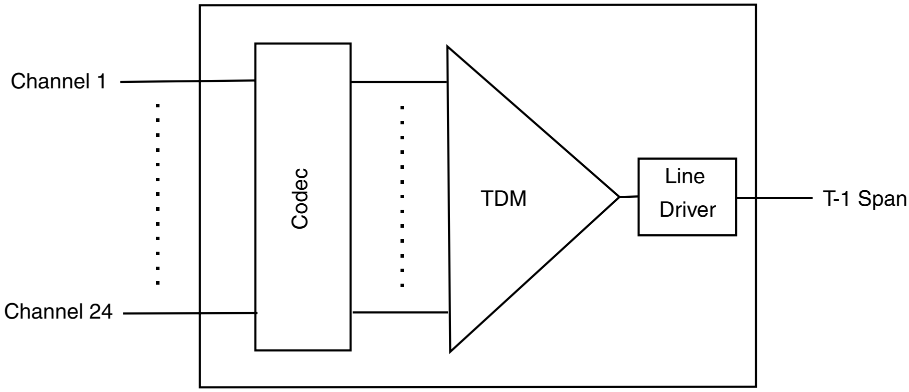

Figure 4. T-1 Channel Bank

In a T-1 channel bank, 24 analog voice channels are converted to digital voice channels (DS-0 channels) by the codec (coder/decoder) using pulse code modulation. These channels then become 64 kbps digital voice channels. The 24 digital voice channels are then multiplexed by the time division multiplexer (TDM) onto a single high speed line, the T-1 span, which is referred to as a DS-1 line.

In each scan cycle, the multiplexer sequentially takes a byte of information from each of the 24 channels and outputs this byte-interleaved information as a serial bit stream in the T-1 span. In the bit stream, the eight bits of information from each channel is called a time slot. For each scan cycle of the TDM, an 8-bit byte of information from each of the 24 channels is placed in that channel’s time slot. The 24 time slots represent one frame of data on the T-1 line. See Figure 5, “T-1 Data Stream”.

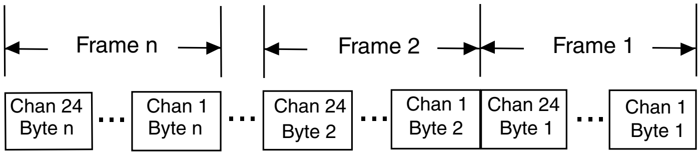

Figure 5. T-1 Data Stream

3.1.2. T-1 Data Rate

The data rate of the T-1 line is the product of multiplying the speed of an individual DS-0 channel by the number of channels being multiplexed (24).

```
64,000 bps X 24 = 1,536,000 bps
```

In addition to the 24 channels, an extra 8000 bits is required for control signals and synchronizing the digital data transmitted and received over the T-1 span.

```
1,536,000 bps + 8000 bps = 1,544,000 bps
```

3.1.3. T-1 Transmission

In T-1 transmission, data is transmitted over one signal pair and simultaneously received on another signal pair. This is known as full duplex transmission.

Before the data is output to the T-1 line it must be conditioned by the line driver to meet the electrical characteristics of the T-1 span. Such things as pulse width, pulse height, and pulse voltages must conform to the T-1 transmission requirements.

In addition, the line driver converts the unipolar signal output from the multiplexer into a bipolar signal which is also required by the T-1 line. In bipolar signaling, each successive digital 1 has the opposite polarity of the previous one. See Figure 6, “Bipolar Signal”. The type of bipolar signaling normally used in T-1 transmission is known as bipolar return to zero (BRZ).

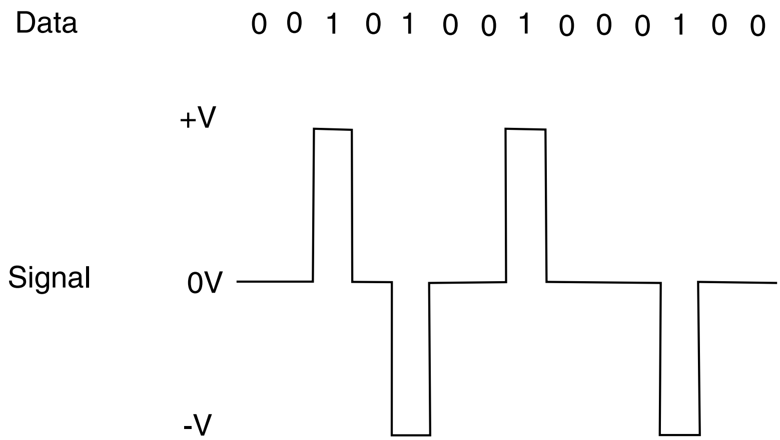

Figure 6. Bipolar Signal

3.1.4. T-1 Ones Density

To maintain proper synchronization between two multiplexers, the T-1 line must contain frequent one's. There is no separate clock signal, so the timing information is derived from the received data signal. If a long string of zero's occur in the data, the potential exists for the T-1 line to lose synchronization. A T-1 carrier cannot have more than 15 consecutive zero's and there must be approximately three one's in every 24 consecutive bits.

To maintain ones density when digital data is being transmitted (as opposed to digital voice), a method called Pulse Stuffing is used. This scheme sets the eighth bit in every byte to a value of 1. Because of this, only the first seven bits can be used for data, resulting in 56 kbps channels.

To maintain one's density in digital voice channels, a coding sequence is inserted to replace a long string of zeros. The inserted bits are coded such that the bipolar signal is violated because successive ones have the same polarity. These violations are recognized at the receiving end of the T-1 line and replaced by a string of zeros, restoring the data to the original state. For T-1 lines, the method used is known as binary eight zero substitution (B8ZS).

3.2. T-1 Framing

Two formats are used in T-1 framing to support robbed bit signaling, D4 Superframe and Extended Superframe. This section discusses the following:

- Basic Framing Format
- D4 Superframe
- Extended Superframe

3.2.1. Basic Framing Format

The basic T-1 frame consists of 24 time slots, each time slot carrying eight bits of data that represent one pulse code modulation (PCM) encoded voice signal digitized at 64 kbps. To denote the beginning of each sequence of 24 digitized DS-0 channels, a special framing bit is inserted at the beginning of each multiplexing cycle. See Figure 7, "Basic T-1 Framing Format". This pattern of 24 DS-0 channel time slots and a framing bit is known as the DS-1 frame.

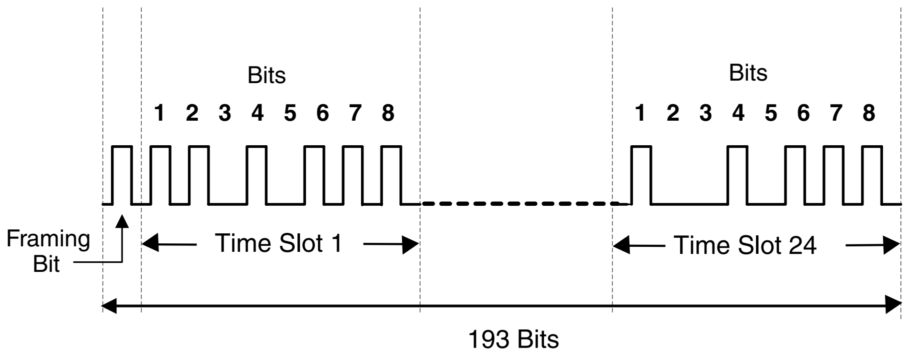

Figure 7. Basic T-1 Framing Format

In order to convey signaling for each of the 24 DS-0 digitized voice channels, a method known as robbed-bit signaling is used. This signaling method requires that individual frames be identified, as bit robbing is only performed on certain frames.

3.2.2. D4 Superframe

The D4 Superframe is based on the standard D4 format used in AT&T D4 channel banks and consists of 12 consecutive T-1 frames. The framing bits in this group of 12 T-1 frames form a repetitive pattern that identifies the group as a superframe. See Figure 8, "D4 Superframe Format" for the structure of a D4 Superframe.

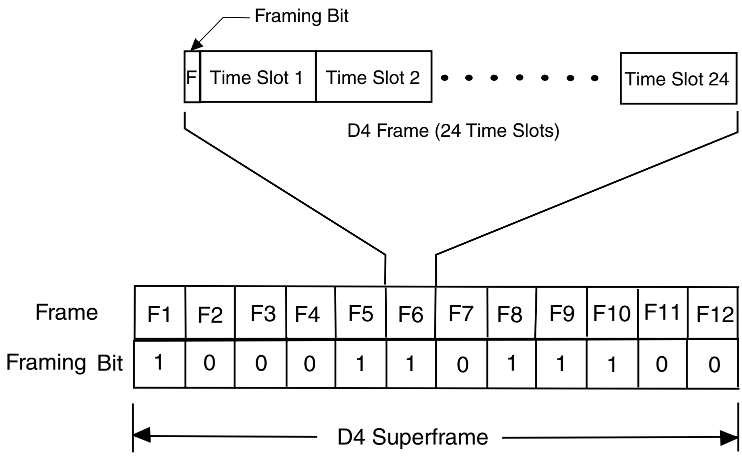

Figure 8. D4 Superframe Format

By providing a repetitive pattern in the framing bit, the receiving end is able to identify each frame transmitted in the superframe. This is important in robbed-bit signaling as bit robbing is performed on the DS-0 channels in Frame 6 and Frame 12 in the D4 Superframe.

3.2.3. Extended Superframe

The Extended Superframe (ESF) extends the number of frames in the framing bit (F-bit) pattern from 12 to 24. See Figure 9, "Extended Superframe Format". Unlike D4 framing in which the framing bits form a specific repeating pattern, the ESF bit pattern can vary.

The ESF framing bits are used for the following purposes:

- The odd numbered bits, known as 'd' bits, are used by the carrier to provide network monitoring, alarm generation, and reconfiguration information.
- Frame bits 2, 6, 10, 14, 18, and 22 are used as a 6-bit cyclic redundancy check sum known as CRC-6. These bits are used by the receiving end to measure the bit error rate.
- Frame bits 4, 8, 12, 16, 20, and 24 are used to generate the ESF framing pattern of 001011.

Similar to D4 framing, the ESF framing pattern allows the receiving end to identify each frame transmitted in the extended superframe for the purpose of robbed-bit signaling. In this case, bit robbing is performed on each DS-0 channel in Frame 6, 12, 18, and 24.

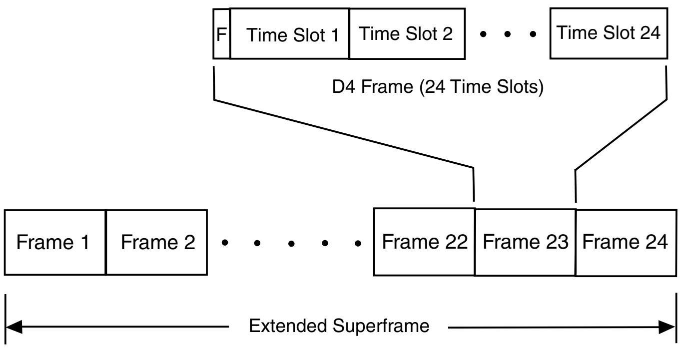

Figure 9. Extended Superframe Format

3.3. T-1 Signaling

In addition to carrying digital voice signals, the T-1 line must also convey signaling information for each of the DS-0 channels. Two different methods are used to transmit this information:

- Channel Associated Signaling (CAS)
- Common Channel Signaling (CCS)

3.3.1. Channel Associated Signaling (CAS)

This section discusses the following topics related to CAS signaling:

- Robbed-Bit Signaling
- E&M Signaling
- Loop Start Signaling
- Ground Start Signaling

Robbed-Bit Signaling

In channel associated signaling (CAS), the signaling information is directly associated with each respective digital voice channel. The method for providing this type of signaling on T-1 lines is referred to as robbed-bit signaling. This type of signaling requires that specific frames in the T-1 transmission be identified because the least significant bit is "robbed" from each channel in specific frames.

In the case of the D4 Superframe, this operation is performed in frames 6 and 12; in the Extended Superframe (ESF), it is performed in frames 6, 12, 18, and 24. By using only the least significant from each channels digital voice date in every 6th frame for signaling purposes, no discernible change can be detected in the voice signal by the called party.

For example, the following 8-bit byte:

```
10101101
```

has a decimal equivalent of 173. If we were to remove the least significant bit, the decimal equivalent would be 172 - a negligible change when conveying digital voice information. If this were digital data other than voice, however, the change would be significant, and could not be tolerated.

The bit that is used from frame 6 is called the "A bit" and the bit that is used from frame 12 is called the "B bit". This is also referred to as AB signaling. In ESF, since bits are robbed from two additional frames, this is referred to as ABCD signaling. In some cases for ESF, however, the C and D bits are used to repeat the A bit and B bit information respectively. D4 Superframe is limited to AB signaling which means that only a limited number of signal conditions can be conveyed (2^2). These conditions include off hook, on hook, busy, and dial pulses.

In the case of a fully implemented ESF facility, 16 different signal conditions can by conveyed (2^4), including those supported by AB signaling.

E&M Signaling

E&M signaling uses both two-wire and four-wire circuits to connect telephone company switches and PBXs. In E&M signaling, the local end signals the remote end by applying -48 VDC to its M lead to seize control of the circuit. This signal is received by the remote end E lead which then toggles its M lead to signal the local end to proceed with the call. The call is terminated when either end drops its M lead.

Loop Start Signaling

Loop start is a two-wire (tip and ring) signaling method used between a telephone and a PBX or between a telephone and central office switch. In loop start signaling, going off hook closes a relay at the PBX or central office that causes current to flow in a circuit (loop) between the telephone and PBX or central office.

Once off hook is detected by the PBX or central office, a dial tone is returned to the telephone. The dialed digits are then processed by the PBX or central office and the call is established. Once the call is completed, going on hook opens the relay and causes current to stop flowing in the loop.

Ground Start Signaling

Ground start signaling is similar to loop start in that both use two-wire circuits and the current flows in a loop. Ground start signaling is normally used between a PBX and central office switch, but it can be used in other applications where the ability to seize the line from either end is desired. In ground start signaling, seizure of the line is accomplished by a momentary grounding of one of the circuit wires, usually the ring of the tip and ring circuit.

3.3.2. Common Channel Signaling (CCS)

Common channel signaling (CCS) is a method in which a separate channel is used to carry the signaling information for a group of other channels. This common channel can be inband where it is a part of the same trunk as the group of channels for which it is providing the signaling, or it can be out-of-band where the signaling information travels over a different path from the other channels. CCS types include:

- Primary Rate ISDN
- Signaling System 7

Primary Rate ISDN

Primary rate Integrated Services Digital Network (ISDN) is an example of inband CCS. In primary rate ISDN, DS-0 channel 24 carries the signaling information for DS-0 channels 1 through 23. This channel is referred to as the D channel.

Signaling System 7

Signaling System 7 (SS7) is an example of out-of-band CCS. In this system, SS7 signaling messages are exchanged between network elements over 56 or 64 kbps bidirectional channels. One application of SS7 signaling is in calling a toll-free number. For this type of call, the numbers are not directly associated with another telephone exchange. Instead, the local service provider needs to query a database to determine where the number is physically located.

3.4. T-1 Coding Types

For T-1 lines, the coding types are:

- Alternate Mark Inversion
- Binary Eight Zero Substitution

3.4.1. Alternate Mark Inversion

Alternate mark inversion (AMI) is a form of bipolar signaling in which each successive mark (digital 1) is of the opposite polarity and spaces (digital 0's) have zero amplitude. In AMI, the transmitted power is concentrated in the middle of the transmission bandwidth. This minimizes signal distortion while eliminating DC voltage buildup on the line.

AMI line encoding does not, however, provide a method for maintaining ones density. To ensure adequate line synchronization, pulse stuffing may be required, restricting the useful bandwidth of each DS0 channel to 56 kbps. While satisfactory for digital data, this will not support 64 kbps digital voice.

3.4.2. Binary Eight Zero Substitution

Advantages of bipolar signaling are the absence of a DC voltage component in the signal and the ability to recover clocking from an all-one's condition. A disadvantage, however, is that during a period of zero voltage (no signal), the T-1 line may not be able to recover clocking and some type of coding may be required to maintain synchronization.

In binary eight zero substitution (B8ZS), each string of eight consecutive zero's in a byte are replaced by the B8ZS code. If the pulse preceding an all-zero byte is positive, the inserted code is:

```
000+-0-+
```

If the pulse preceding an all-zero byte is negative, the inserted code is:

```
000-+0+-
```

Either of these codes results in bipolar violations occurring in the fourth and seventh bit positions. Both ends of the T-1 line must recognize these codes and replace a byte containing the coded bipolar violations with the original eight zeros.

B8ZS encoding, therefore, provides a method of line coding that permits the full 64 kbps bandwidth of each DS0 channel to be utilized.

3.5. T-1 Alarms

Alarm and error conditions are monitored and reported on T-1 lines. Principal T-1 alarms include:

- Red Alarm
- Yellow Alarm

3.5.1. Red Alarm

A red alarm is generated by the device at the receiving end of a T-1 line to report a loss of signal or frame alignment (synchronization) in the signal being received (incoming data). This alarm is declared after the condition has existed for a specific time period, which is typically defined as 2.5 seconds.

A red alarm condition will exist until frame alignment is recovered and remains recovered for a specific time period, which is typically defined as 12 seconds.

3.5.2. Yellow Alarm

A yellow alarm is generated by the device at the receiving end of a T-1 line and sent to the device at the transmitting (remote) end to signify that a red alarm condition exists at the receiving (local) end.

The yellow alarm is sent to the transmitting device as long as the red alarm condition exists at the receiving device.

3.6. T-1 Protocol/Switch Types

T-1 protocols are primarily determined by the type of switch that the T-1 line is associated with. A number of different switch types are used in T-1 networks. Each switch type uses a unique protocol that must be supported by any device that interfaces to that switch. T-1 protocol/switch types include:

- D4 (Robbed Bit Signaling)
- 4ESS
- 5ESS

- DMS
- QSIG
- NTT
- NI2

3.6.1. D4 (Robbed Bit Signaling)

The D4 switch is a fourth generation AT&T channel bank that provides D4 framing (D4 Superframe). D4 framing supports robbed-bit signaling, but does not support any ISDN protocols. For more information about D4 framing, see Section 3.2.2, "D4 Superframe", on page 16.

3.6.2. 4ESS

The 4ESS switch is an AT&T toll telephone switch that can handle more than 100,000 T-1 trunks and over 500,000 call attempts per hour. The 4ESS switch is primarily used for switching digital voice channels, but it also supports primary rate ISDN.

3.6.3. 5ESS

The 5ESS switch is an AT&T multi-service modular switch that uses distributed intelligence. The switch can support up to 250,000 subscriber lines and over 100,000 T-1 trunks. The 5ESS switch can handle both digital voice channels as well as data, and supports both basic rate and primary rate ISDN.

3.6.4. DMS

The DMS protocol applies to the DMS-100 switch provided by Northern Telecom (Nortel) for primary rate ISDN applications.

3.6.5. QSIG

QSIG is a primary rate ISDN standard that is used globally by private ISDN exchanges. The signaling protocol for this standard is defined by Q.931.

3.6.6. NTT

The NTT protocol applies to the INS-Net 1500 switch that is used by Nippon Telephone and Telegraph (NTT) for primary rate ISDN.

3.6.7. NI2

NI2 (National ISDN-2) is a U.S. ISDN standard software interface that can be installed on most switch types, providing maximum inter operability with ISDN lines.

4. E-1 Technology

The following topics provide information about the major aspects of E-1 technology:

- E-1 Multiplexing
- E-1 Framing
- E-1 Signaling
- E-1 Coding Types
- E-1 Alarms
- E-1 Protocol/Switch Types

4.1. E-1 Multiplexing

E-1 is a multiplexing scheme used primarily in Europe that allows 30 individual voice channels to be carried on a common transmission medium. E-1 is also referred to as a CEPT PCM-30 system. In this system, 30 DS-0 channels are used to carry digital voice channels and 2 channels are used for signaling and timing information. This section includes the following topics regarding E-1 multiplexing:

- Basic E-1 Data Stream
- E-1 Data Rate
- E-1 Transmission
- E-1 Ones Density

4.1.1. Basic E-1 Data Stream

E-1 multiplexing is based on the Conference of European Postal & Telecommunications (CEPT) standards and provides a more efficient means of transporting telephone transmissions over long distances. By converting analog voice signals to digital signals using pulse coded modulation (PCM) and then multiplexing these signals onto a high speed digital line, 30 separate phone calls can be transmitted simultaneously over a single transmission path. E-1 technology is also used for connecting central offices and providing a more efficient service to private enterprise customers who have a high volume of telephone traffic.

In each scan cycle, the E-1 multiplexer sequentially takes a byte of information from each of the 30 channels and outputs this byte-interleaved information as a serial bit stream in the E-1 span. In the bit stream, the eight bits of information from each channel is called a time slot. For each scan cycle of the TDM, an 8-bit byte of information from each of the 30 channels is placed in that channel's time slot. The 30 voice time slots (time slots 1-15 and 17-31) plus a time slot for channel signaling (time slot 16) and a time slot for frame synchronization (time slot 0) represent one frame of data on the E-1 line. See Figure 10, "E-1 Data Stream".

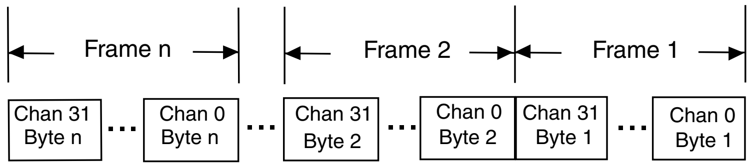

Figure 10. E-1 Data Stream

4.1.2. E-1 Data Rate

The data rate of the E-1 line is the product of multiplying the speed of an individual DS-0 channel by the number of channels being multiplexed (32).

```
64,000 bps X 32 = 2,048,000 bps
```

4.1.3. E-1 Transmission

In E-1 transmission, data is sent over one signal pair and simultaneously received on another signal pair. This is known as full duplex transmission.

Before the data is output to the E-1 line it must be conditioned by the line driver to meet the electrical characteristics of the E-1 span. Such things as pulse width, pulse height, and pulse voltages must conform to the E-1 transmission requirements.

In addition, the line driver converts the unipolar signal output from the multiplexer into a bipolar signal which is also required by the E-1 line. In bipolar signaling, each successive digital 1 has the opposite polarity of the previous one. See Figure 6, "Bipolar Signal". The type of bipolar signaling normally used in E-1 transmission is known as bipolar return to zero (BRZ).

4.1.4. E-1 Ones Density

To maintain proper synchronization between two multiplexers, the E-1 line must contain frequent one's. If a string of zero's occur in the data, the potential exists for the E-1 line to lose synchronization. An E-1 line is monitored for any group of four consecutive zeros.

To maintain ones density, a method employing bipolar violations is used. This method inserts a coding sequence to replace a string of zeros in the data stream. The inserted bits are coded such that the bipolar signal is violated because successive ones have the same polarity. These violations are recognized at the receiving end of the E-1 line and replaced by a string of zeroes, restoring the data to its original state. For E-1 lines, the most common method used is HDB3 (high density bipolar 3).

4.2. E-1 Framing

This section also includes information on the following topics:

- Basic Framing Format
- E-1 Frame Synchronization
- E-1 Multiframe

4.2.1. Basic Framing Format

The basic E-1 frame consists of 32 time slots. Time slots 1 through 15 and 17 through 31 each carry eight bits of data that represent one pulse code modulation (PCM) encoded voice signal digitized at 64 kbps. Time slot 0 is used for synchronization of the E-1 line and time slot 16 is used to convey signaling for all 30 voice channels on the E-1 line. See Figure 11, "E-1 Framing Format".

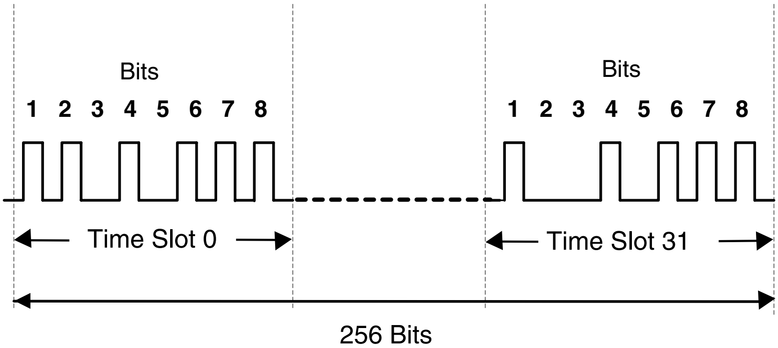

Figure 11. E-1 Framing Format

4.2.2. E-1 Frame Synchronization

So that the receiver is able to identify each DS-0 channel in the E-1 frame, a frame alignment signal (FAS) is transmitted in time slot 0 of every even frame using bits 2 through 8. Bit 1 is used for the international bit.

The odd frames are used to carry national and international signaling as well as an alarm indication for loss of frame alignment. Bit 2 in the odd frame is set to a 1 to prevent the bit pattern in the odd frame from duplicating the FAS in the even frames. See Figure 12, “E-1 Frame Synchronization”.

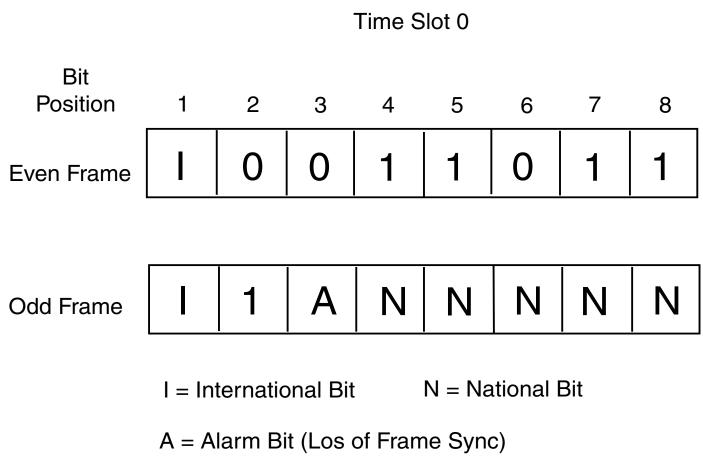

Figure 12. E-1 Frame Synchronization

4.2.3. E-1 Multiframe

The E-1 multiframe consists of 16 consecutive E-1 frames. Because frames 1 through 15 in the multiframe carry the signaling for specific DS-0 channels in time slot 16, it is necessary to identify one frame from another in the multiframe. Frame 0 contains additional synchronization information that identifies the beginning of a multiframe. This is signified in time slot 16 by a pattern of four zeros in bit positions 1 through 4 and is known as the multiframe alignment signal (MAS). Bits 5,7, and 8 of this time slot are used for multiframe synchronization and bit 6 is used to indicate the loss of multiframe alignment. See Figure 13, “Multiframe Alignment” and Figure 14, “E-1 Multiframe Format”.

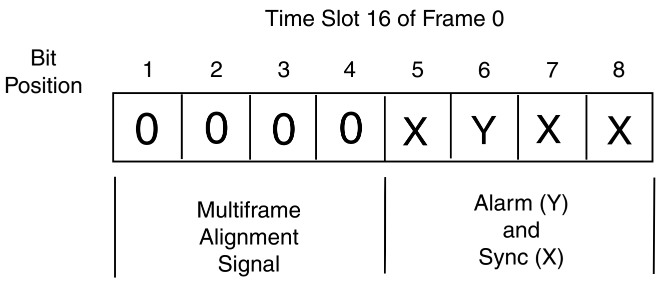

Figure 13. Multiframe Alignment

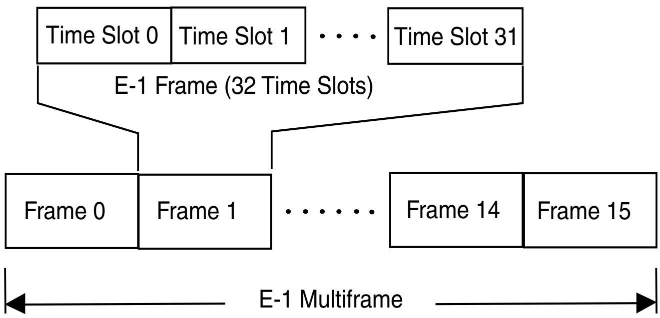

Figure 14. E-1 Multiframe Format

4.3. E-1 Signaling

In addition to carrying digital voice signals, the E-1 line must also convey signaling information for each of the DS-0 channels. The signaling includes such information as off hook, on hook, busy, and dial pulses. The E-1 signaling types include:

- Basic ABCD Signaling
- Common Channel Signaling

- Clear Channel

4.3.1. Basic ABCD Signaling

Time slot 16 of frames 1 through 15 in an E-1 multiframe is used for channel signaling to convey such conditions as on-hook, off-hook, dialed digits, and call progress. Each frame carries the ABCD signaling for two of the 30 channels. In a given frame, the first four bits of time slot 16 are used to convey the ABCD signaling for a specific DS-0 channel and the other four bits in the time slot convey the ABCD signaling for another specific DS-0 channel. See Figure 15, "E-1 Channel Signaling". Frame 1 in the multiframe carries the signaling for DS-0 channels 1 and 16, Frame 2 carries the signaling for DS-0 channels 2 and 17. This pattern continues through to frame 15 which carries the signaling for DS-0 channels 15 and 30. Because a single channel (DS-0 channel 16) carries the signaling for all 30 DS-0 digital voice channels, this is referred to as common channel signaling (CCS).

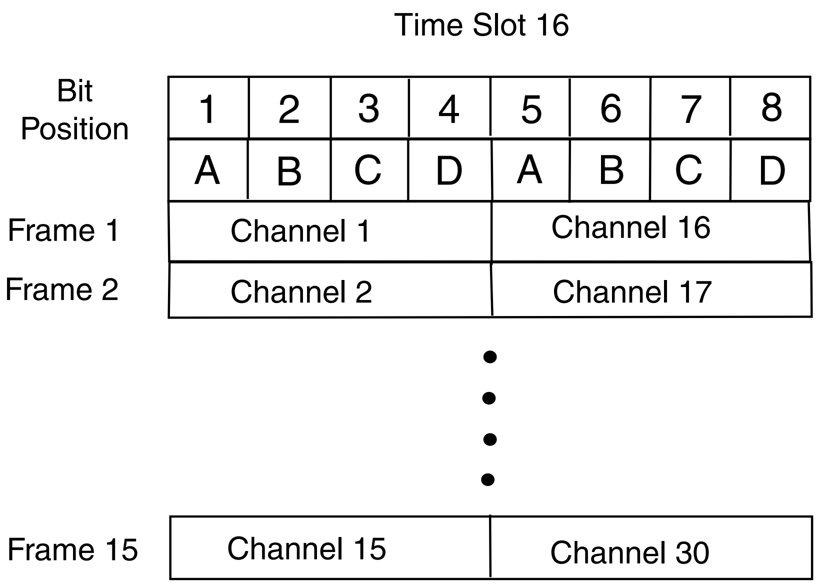

Figure 15. E-1 Channel Signaling

4.3.2. Common Channel Signaling

In addition to the standard ABCD signaling scheme, other common channel signaling (CCS) types used with E-1 transmission facilities include:

- Primary Rate ISDN
- Digital R2MF
- Signaling System 7

Primary Rate ISDN

In E-1 primary rate Integrated Services Digital Network (ISDN) applications, DS-0 channels 1 through 15 and 17 through 31 carry data, digital voice, and compressed video. These channels are referred to as bearer (B) channels. DS-0 channel 16 carries the ISDN signaling for all 30 B channels and is referred to as the D channel. This is a type of inband common channel signaling.

Digital R2MF

Digital R2MF signaling is a form of inband common channel signaling that uses DS-0 channel 16 to convey the signaling for the 30 DS-0 voice channels. In R2MF signaling, two different sets of six frequencies in combination are used to convey signaling information. One set is used for the forward signals and the other set is used for the backward signals.

For each set of six frequencies, each signal is composed of two out of six fundamental frequencies to create a multi frequency code. This produces 15 multi frequency signals for each direction. The forward signals (signals sent from the central office to the customer premise equipment) are classified as Group I and Group II signals. The backward signals (signals sent from the customer premise equipment to the central office) are classified as Group A and Group B signals.

Group I forward signals and Group A backward signals are used for setting up calls and transferring address information. Group II forward signals provide the calling party's category information and Group B backward signals provide the condition of the called subscriber's line.

Signaling System 7

In Signaling System 7 (SS7), signaling messages are exchanged between network elements over 56 or 64 kbps bidirectional channels. This is known as out-of-band common channel signaling.

One application of SS7 signaling is in calling a toll-free number. For this type of call, the numbers are not directly associated with another telephone exchange. Instead, the local service provider needs to query a database to determine where the number is physically located.

4.3.3. Clear Channel

Clear channel uses none of the E-1 line's bandwidth for signaling. All of the DS0 channels are available to carry data. An example of a clear channel application is TS16, where time slot 16 is used to carry data instead of signaling information.

4.4. E-1 Coding Types

For E-1 lines, the coding types are:

- Alternate Mark Inversion
- High Density Bipolar Three Coding

4.4.1. Alternate Mark Inversion

Alternate mark inversion (AMI) is a form of bipolar signaling in which each successive mark is of the opposite polarity and spaces have zero amplitude. In AMI, the transmitted power is concentrated in the middle of the transmission bandwidth. This minimizes signal distortion while eliminating DC voltage buildup on the line.

4.4.2. High Density Bipolar Three Coding

Advantages of bipolar signaling are the absence of a DC voltage component in the signal and the ability to recover clocking from an all-one's condition. A disadvantage, however, is that during a period of zero voltage (no signal), the E-1 line may not be able to recover clocking and some type of coding may be required to maintain synchronization.

In high density bipolar three (HDB3) coding, each string of four consecutive zeros in a byte are replaced by the HDB3 code. Two different HDB3 codes are used to insure that the bipolar violations from adjacent four-zero groups are of the opposite polarity. The choice of which of the two codes to use depends on whether there was an odd or even number of ones since the previous bipolar violation occurred.

If an odd number of ones occurred since the previous bipolar violation, the following coding method is used:

```
000V
```

where V is a bipolar violation.

If an even number of ones occurred since the previous bipolar violation, the following coding method is used:

```
P00V
```

where P is a bit having the opposite polarity of the immediately preceding bit and V is a bipolar violation.

4.5. E-1 Alarms

Alarm and error conditions are reported on E-1 lines. Principal E-1 alarms include:

- Red Alarm
- Yellow Alarm
- Multiframe Red Alarm
- Multiframe Yellow Alarm

4.5.1. Red Alarm

A red alarm is generated by the device at the receiving end of an E-1 line to report a loss of signal or frame alignment (synchronization) in the signal being received (incoming data). The red alarm is indicated by the alarm indication signal, bit 5 in time slot 0 of the odd frames. When a red alarm has been declared, this bit is set to a 1. See Figure 12, “E-1 Frame Synchronization”.

This alarm is declared after the condition has existed for a specific time period, which is typically defined as 2.5 seconds. A red alarm condition will exist until frame alignment is recovered and remains recovered for a specific time period, which is typically defined as 12 seconds.

4.5.2. Yellow Alarm

A yellow alarm is generated by the device at the receiving end of an E-1 line and sent to the device at the transmitting (remote) end to signify that a red alarm condition exists at the receiving (local) end. The yellow alarm is sent to the transmitting device as long as the red alarm condition exists at the receiving device.

The yellow alarm is indicated by the alarm indication signal, bit 5 in time slot 0 of the odd frames. When a yellow alarm has been declared, this bit is set to a 1. See Figure 12, “E-1 Frame Synchronization”.

4.5.3. Multiframe Red Alarm

A multiframe red alarm is generated by the device at the receiving end of an E-1 line to report a loss of multiframe alignment (synchronization) in the signal being received (incoming data). A receiver loses multiframe alignment because of either the occurrence of two consecutive errors in the multiframe alignment signal (MAS), or because time slot 16 contains all zeros for at least one multiframe.

The multiframe red alarm is indicated by the alarm indication signal, bit 2 in time slot 16 of frame 0. When a red alarm has been declared, this bit is set to a 1. See Figure 13, “Multiframe Alignment”.

This alarm is declared after the condition has existed for a specific time period, which is typically defined as 2.5 seconds. A red alarm condition will exist until frame alignment is recovered and remains recovered for a specific time period, which is typically defined as 12 seconds.

4.5.4. Multiframe Yellow Alarm

A multiframe yellow alarm is generated by the device at the receiving end of an E-1 line and sent to the device at the transmitting (remote) end to signify that a multiframe red alarm condition exists at the receiving (local) end. The multiframe yellow alarm is sent to the transmitting device as long as the multiframe red alarm condition exists at the receiving device.

The multiframe yellow alarm is indicated by the alarm indication signal, bit 2 in time slot 16 of frame 0. When a yellow alarm has been declared, this bit is set to a 1. See Figure 13, “Multiframe Alignment”.

4.6. E-1 Protocol/Switch Types

E-1 protocols are primarily determined by the type of switch that the E-1 line is associated with. A number of different switch types are used. Each switch type uses a unique protocol that must be supported by any device that interfaces to that switch. E-1 protocol/switch types include:

- Net5
- QSIG
- R2MF
- TS16

4.6.1. Net5

Net5 is a European ISDN primary rate switch.

4.6.2. QSIG

QSIG is a primary rate ISDN standard that is used globally by private ISDN exchanges. The signaling protocol for this standard is defined by Q.931.

4.6.3. R2MF

The digital version of R2MF is an international signaling system that is used mostly in Europe and Asia in non-ISDN applications to permit the transmission of numerical and other information relating to the called and calling subscriber lines. R2MF uses a multi frequency code based on six fundamental frequencies in the forward direction and a different set of six frequencies in the backward direction.

4.6.4. TS16

TS16 is a form of clear channel operation which allows time slot 16 to be used for data instead of signaling.

5. Interfacing to T-1 or E-1 Service

This section includes information about:

- Interfacing to a T-1 Line
- Interfacing to an E-1 Line

5.1. Interfacing to a T-1 Line

Connecting to a T-1 line requires specific termination equipment that conforms to the interface specifications of the provider. When connecting to a T-1 service, certain conditions must be met. The following topics discuss those conditions:

- Customer Premises Equipment
- T-1 Signal Characteristics
- Connecting to the T-1 Line

5.1.1. Customer Premises Equipment

T-1 signals are transmitted primarily over standard twisted-pair copper wire. One twisted pair is used for the transmit signal and a second twisted pair is used for the receive signal. The T-1 line requires a repeater every 6000 feet to maintain the proper signal conditions. The last repeater in the span before connecting to customer premises equipment (CPE) should be no more than 3000 feet from the interface. This repeater also includes the network interface unit (NIU). See Figure 16, “Interfacing to the T-1 Line”.

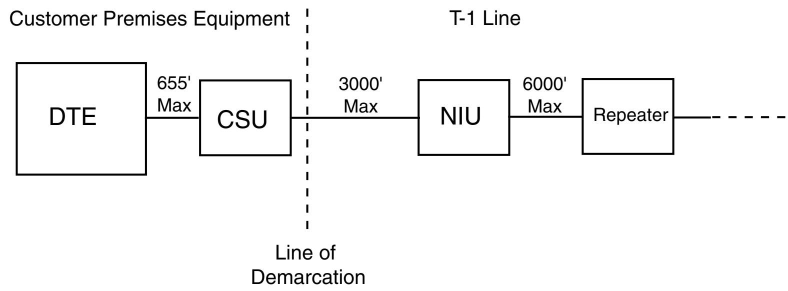

Figure 16. Interfacing to the T-1 Line

The NIU connects to the CPE via a channel service unit (CSU). The CSU is a type of data communications equipment (DCE) and can be thought of as a type of digital modem. The CPE is a form of data terminal equipment (DTE) and includes:

- Multiplexers
- PBXs
- Channel Banks
- Front-End Processors
- Computers

The CSU provides a number of termination and interface functions that include:

- Electrical interface
- Surge and lightning protection
- Signal regeneration
- Pulse density
- Keep Alive signal
- Yellow Alarm signal
- Loopback capability

5.1.2. T-1 Signal Characteristics

The T-1 signal received from the T-1 line and transmitted to the T-1 line is a bipolar pulse train. Logical zeros are zero voltage, while logical ones alternate between plus and minus 3 volts. When a pulse is present, the pulse width will be one half of the time slot bit duration. See Figure 17, "T-1 Signal".

According to AT&T Publication 62411, the positive voltage is 3.0 volts ±0.3 volts, while the negative voltage is its absolute value (without sign) and must be within 0.20 volts of the positive voltage, but no less than 2.7 volts or greater than 3.3 volts.

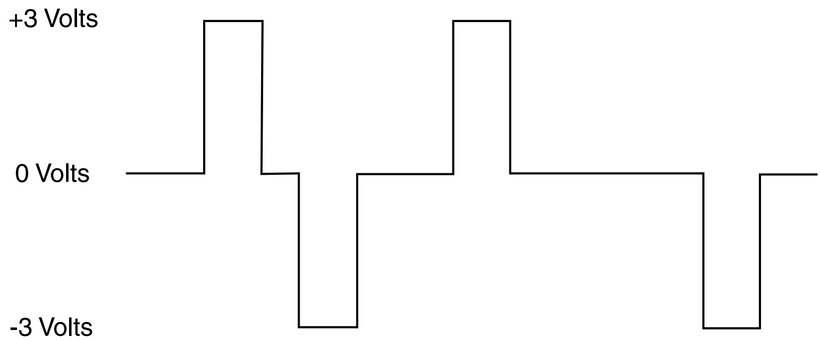

Figure 17. T-1 Signal

5.1.3. Connecting to the T-1 Line

Typically, a T-1 line is terminated at the CSU by an RJ-48C connector. This is an 8-pin connector and has the following pin assignments:

- Pin 1: Receive signal from the network (ring)
- Pin 2: Receive signal from the network (tip)
- Pin 3: No connection

- Pin 4: Transmit signal to the network (ring)
- Pin 5: Transmit signal to the network (tip)
- Pin 6: No connection
- Pin 7: Optional shield
- Pin 8: Optional shield

5.2. Interfacing to an E-1 Line

Connecting to an E-1 line requires specific termination equipment that conforms to the interface specifications of the provider. When connecting to E-1 service, certain conditions must be met. The following topics discuss those conditions:

- Customer Premises Equipment
- E-1 Signal Characteristics
- Connecting to the E-1 Line

5.2.1. Customer Premises Equipment

E-1 signals are transmitted primarily over standard twisted-pair copper wire. One twisted pair is used for the transmit signal and a second twisted pair is used for the receive signal. The E-1 line requires a repeater every 6000 feet to maintain the proper signal conditions. The last repeater in the span before connecting to customer premises equipment (CPE) should be no more than 3000 feet from the interface. This repeater also includes the network interface unit (NIU). See Figure 18, “Interfacing to the E-1 Line”.

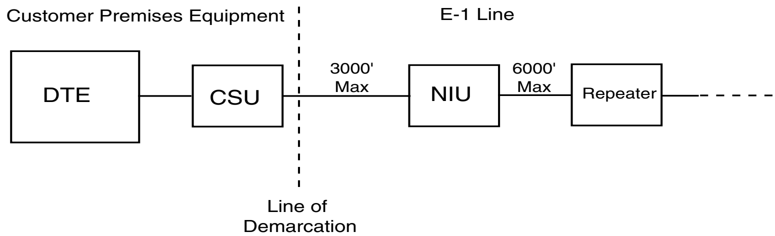

Figure 18. Interfacing to the E-1 Line

The NIU connects to the CPE via a channel service unit (CSU). The CSU is a type of data communications equipment (DCE) and can be thought of as a type of digital modem. The CPE is a form of data terminal equipment (DTE) and includes:

- Multiplexers
- PBXs
- T1 Channel Banks
- Front-End Processors
- Computers

The CSU provides a number of termination and interface functions that include:

- Electrical interface
- Surge and lightning protection
- Signal regeneration
- Pulse density
- Keep Alive signal
- Yellow Alarm signal
- Loopback capability

5.2.2. E-1 Signal Characteristics

The E-1 signal received from the E-1 line and transmitted to the E-1 line is a bipolar pulse train. Logical zeros are zero voltage, while logical ones alternate between plus and minus 3 volts. When a pulse is present, the pulse width will be one half of the time slot bit duration. See Figure 19, “E-1 Signal”.

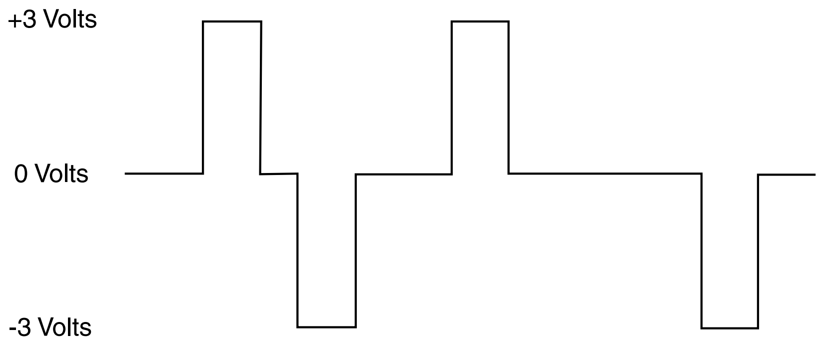

Figure 19. E-1 Signal

5.2.3. Connecting to the E-1 Line

An E-1 line is typically terminated at the CSU by either an RJ-48C connector (120 ohm balanced interface) or a pair of BNC (coaxial) connectors (75 ohm unbalanced interface).

For the 120 ohm balanced interface, this is an 8-pin RJ-48C connector that has the following pin assignments:

- Pin 1: Receive signal from the network (ring)
- Pin 2: Receive signal from the network (tip)
- Pin 3: No connection
- Pin 4: Transmit signal to the network (ring)

- Pin 5: Transmit signal to the network (tip)
- Pin 6: No connection
- Pin 7: Optional shield
- Pin 8: Optional shield

For the 75 ohm unbalanced interface, a pair of BNC connectors is used to connect to the E-1 line. One BNC connector accepts the receive signal coax cable and a second BNC connector accepts the transmit signal coax cable.

6. Industry Standards

This section discusses:

- Industry Standards for T-1 Networking
- Industry Standards for E-1 Networking

6.1. Industry Standards for T-1 Networking

Standards that support T-1 networking include:

- AT&T Publication 43801
- AT&T Publication 54016
- AT&T Publication 62411
- ANSI T1.403-1989
- Bellcore TR-TSY-000194

6.1.1. AT&T Publication 43801

AT&T Publication 43801 provides information regarding digital channel bank requirements and objectives.

6.1.2. AT&T Publication 54016

AT&T Publication 54016 provides requirements for interfacing data terminal equipment to services employing the extended superframe format.

6.1.3. AT&T Publication 62411

AT&T Publication 62411 addresses the specifications for a high capacity digital service channel interface.

6.1.4. ANSI T1.403-1989

ANSI T1.403 is the American National Standard for a DS1 metallic interface with respect to carrier-to-customer installations.

6.1.5. Bellcore TR-TSY-000194

Bellcore TR-TSY-000194 describes the Bellcore extended superframe (ESF) format interface specification as it applies to their high capacity digital special access services.

6.2. Industry Standards for E-1 Networking

Standards that support E-1 networking include:

- ITU-T Recommendation G.703
- ITU-T Recommendation G.704
- ITU-T Recommendation G.706
- ITU-T Recommendation G.711
- ITU-T Recommendation G.732
- ITU-T Recommendation G.823
- ITU-T Recommendation I.431

6.2.1. ITU-T Recommendation G.703

ITU-T Recommendation G.703 addressed both the physical and electrical characteristics of hierarchical digital interfaces.

6.2.2. ITU-T Recommendation G.704

ITU-T Recommendation G.704 addresses synchronous frame structures used at 1.544, 2.048, 6.312, 8.488, and 44.736 Mbps. This recommendation refers to a framing format without CRC-4 capabilities.

6.2.3. ITU-T Recommendation G.706

ITU-T Recommendation G.706 addresses frame alignment and cyclic redundancy check (CRC) procedures relating to the basic frame structures defined in ITU-T Recommendation G.704. This recommendation refers to a multiframe format with CRC-4 capabilities.

6.2.4. ITU-T Recommendation G.711

ITU-T Recommendation addresses the pulse code modulation (PCM) of analog voice frequencies.

6.2.5. ITU-T Recommendation G.732

ITU-T Recommendation G.732 addresses the characteristics of primary pulse code modulation (PCM) multiplexing equipment operating at 2.048 Mbps.

6.2.6. ITU-T Recommendation G.823

ITU-T Recommendation G.823 addresses the control of jitter and wander within digital networks which are based on the 2.048 Mbps hierarchy.

6.2.7. ITU-T Recommendation I.431

ITU-T Recommendation I.431 addresses primary rate user-network interface relative to the ISO Layer 1 specification.

7. Intel Terminology

Following is a list of terms used by Intel to identify certain aspects of computer telephony. These terms are listed here because they differ from those used by the industry in general. A definition is provided for each term as well as the equivalent industry term.

baseboard: A printed circuit board without any daughter boards attached used in computer telephony applications. The industry refers to this as a *board*.

device channel: A voice data path that processes one incoming or outgoing call at a time. This would be referred to by the industry as a *channel* or *time slot*.

E1 CAS: Signaling used in E-1 transmission where time slot 16 is used to convey ABCD signaling for each of the 30 voice channels. In frames 1 through 15 of each multiframe, the first four bits of time slot 16 carries the signaling for a specific voice channel and the second 4 bits of the time slot carries the signaling for another specific voice channel. At the end of one multiframe, all 30 voice channels will have been accommodated. The industry refers to this as *basic E-1 ABCD signaling*.

echo-producing circuit: Typically, the interface between a 2-wire (analog) lineside circuit and 4-wire (digital) trunkside circuit, which, due to impedance mismatches, reflects part of the receive signal into the transmit signal. The industry refers to this as a *hybrid*.

in-band signaling: signaling transmitted within an 8-bit voice signal or time slot, as in T-1 “robbed-bit” signaling. The industry refers to this as *channel associated signaling (CAS)*.

NOTE: Both CAS and common channel signaling (CCS) are considered to be “in band” if the signaling is within the bandwidth of the T-1 or E-1 line.

span: The portion of a high speed digital system that connects two switching centers (for example, central offices or PBXs). The industry term for this is a *trunk*.

8. Intel Implementation Specifics

This section discusses representative examples of Intel boards that implement both T-1 and E-1 technology and includes information about the:

- DualSpan Series
- DTI/SC Series
- QuadSpan Voice Series

8.1. DualSpan Series

This section discusses the DualSpan Series and includes the following topics:

- Introduction
- T-1 DualSpan Boards
- E-1 DualSpan Boards

8.1.1. Introduction

The DualSpan Series of digital voice and network interface boards provides two T-1 or E-1 Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) lines of service termination and call processing. The T-1 boards provide service termination and call processing for 24 or 48 voice channels, while the E-1 boards provide service termination and call processing for 30 or 60 voice channels.

These boards handle all telephony signaling and perform DTMF, MF, and audio/voice signal processing tasks. Onboard digital signal processors (DSPs) provide variable voice coding at 24 and 32 kbps adaptive differential pulse code modulation (ADPCM) and 48 and 64 kbps μ-law and A-law pulse code modulation (PCM).

Applications for the DualSpan Series of T-1 and E-1 boards include:

- Call center
- Telemarketing
- Operator services (auto attendant)

- Interactive voice response
- Voice messaging
- Online data entry and query

8.1.2. T-1 DualSpan Boards

Each T-1 interface on the DualSpan board can be connected directly to a channel service unit (CSU) that supports D4 superframe format or extended superframe format (ESF), a digital service unit (DSU), or to other network terminating equipment. The DualSpan Series provides most of the functions that are performed by a DSU, including unipolar to bipolar conversion and T-1 framing. The boards process digital signaling information and voice signals from the T-1 telephone network.

The T-1 DualSpan board supports both robbed-bit signaling and T-1 ISDN signaling protocols. In real time, the board can connect to 24 (D/240SC-2T1) or 48 (D/480SC-2T1) channels, detect touch tones, play voice messages to the caller, compress and record voice signals, and place outbound calls.

8.1.3. E-1 DualSpan Boards

Each E-1 interface on the DualSpan board can be connected directly to a channel service unit (CSU), a digital service unit (DSU), or to other network terminating equipment. The DualSpan Series provides most of the functions that are performed by a DSU, including unipolar to bipolar conversion and E-1 framing. The boards process digital signaling information and voice signals from the E-1 telephone network. E-1 DualSpan boards also support CRC4 cyclic redundancy check.

The E-1 DualSpan board supports both R2MF signaling and E-1 ISDN signaling protocols. In real time, the board can connect to 30 (D/300SC-2E1) or 60 (D/600SC-2E1) channels, detect touch tones, play voice messages to the caller, compress and record voice signals, and place outbound calls.

8.2. DTI/SC Series

This section discusses the DTI/SC Series and includes the following topics:

- Introduction
- T-1 Interface Boards
- E-1 Interface Boards

8.2.1. Introduction

The DTI/SC Series of network interface boards provide digital telephone network service termination for computer telephony systems based on the SCbus. These boards terminate T-1/DSX-1 or E-1/DSX facilities in single or dual-span configurations. For T-1, 24 or 48 ports of voice and data information are passed in real time between the digital telephone network and the SCbus, while for E-1, 30 or 60 ports are supported.

Applications for the DTI/SC Series of T-1 and E-1 network interface boards include:

- Call center
- Telemarketing
- Operator services (auto attendant)
- Predictive dialing
- Voice/audio response systems
- Automatic call rerouting

8.2.2. T-1 Interface Boards

The T-1 boards support robbed-bit signaling and are fully compatible with resource devices that use 1.544 MHz clocking and µ-law pulse code modulation (PCM). All channel associated signaling (CAS) functions are supported, including pulse dialing and detection, winking, and flash. The DTI/SC T-1 boards also support T-1 primary rate Integrated Digital Network Services (ISDN).

The T-1 interface can be connected directly to a channel service unit (CSU) that supports D4 superframe format, a digital service unit (DSU), or to other network terminating equipment. The DTI/SC Series provides most of the functions that are performed by a DSU, including unipolar to bipolar conversion and T-1 framing. The boards process digital signaling information and voice signals from the T-1 telephone network.

8.2.3. E-1 Interface Boards

The E-1 boards support CEPT channel associated signaling and are fully compatible with resource devices that use 2.048 MHz clocking and A-law PCM. R2/MF Compelled Protocol is also supported. All CAS functions are supported, including pulse dialing and detection, winking, and flash. The DTI/SC E-1 boards also support E-1 primary rate ISDN.

The E-1 interface can be connected directly to a channel service unit (CSU), a digital service unit (DSU), or to other network terminating equipment. The DTI/SC Series provides most of the functions that are performed by a DSU, including unipolar to bipolar conversion and E-1 framing. The boards process digital signaling information and voice signals from the E-1 telephone network. E-1 DTI/SC boards also support CRC4 cyclic redundancy check.

8.3. QuadSpan Voice Series

This section discusses the QuadSpan Voice Series and includes the following topics:

- Introduction
- T-1 QuadSpan Boards
- E-1 QuadSpan Boards

8.3.1. Introduction

The QuadSpan Voice Series boards provide advanced voice processing and telephony networking features based on the CT Bus. QuadSpan boards terminate up to four T-1 or E-1 digital network interfaces. The QuadSpan digital signal processors (DSPs) provide voice processing features that include voice compression, recording and playback, telephony tone signaling, DTMF detection using local echo cancellation, and automated out-bound call progress analysis with positive voice detection and positive answering machine detection.

QuadSpan voice boards provide the following functionality in real time on all 96 (T-1) or 120 (E-1) channels:

- Automatically answer calls using virtually any international telephony signaling protocol
- Detect DTMF (touchtone)
- Digitally compress and record voice signals
- Play voice messages to a caller
- Place outbound calls and automatically track call progress

Applications for the QuadSpan Voice Series of T-1 and E-1 boards include:

- Unified messaging
- Voice portal
- Network call center/contact center
- Conferencing
- Prepaid/debit card
- International callback
- Gateway switch

8.3.2. T-1 QuadSpan Boards

Each T-1 interface on a QuadSpan board supports all robbed-bit signaling protocols and is compatible with all interface devices that use 1.544 MHz clocking and μ-law pulse code modulation (PCM). All basic channel associated signaling (CAS) functions such as pulse dialing and detection, winking, and switch hook flash are supported. The T-1 QuadSpan board also supports ISDN primary rate.

The T-1 QuadSpan board connects up to 96 telephone channels via DSX-1 termination. With GlobalCall software, the QuadSpan Series can provide a common signaling interface for network-enabled applications, regardless of the T-1 signaling protocol used to connect to the local telephone network.

8.3.3. E-1 QuadSpan Boards

Each E-1 QuadSpan board supports all CEPT channel associated signaling (CAS) functions and is fully compatible with all interface devices that use 2.048 MHz clocking and A-law pulse coded modulation (PCM).

The E-1 QuadSpan board connects up to 120 telephone channels via DSX-1 termination. With GlobalCall software, the QuadSpan Series can provide a common signaling interface for network-enabled applications, regardless of the E-1 signaling protocol used to connect to the local telephone network.

Glossary

This glossary contains terms specific to T-1 and E-1 technology.

µ-law: An algorithm used in companding a pulse code modulated voice signal. The µ-law algorithm is primarily used in North American T-1 telephone networks.

adaptive differential pulse code modulation: American National Standards Institute (ANSI) standard for digitizing voice signals. By using a modulation method in which only the difference between successive signals is coded, ADPCM requires only a 4-bit byte to describe the analog signal. This results in a 32 kbps digital voice which is half the bandwidth of a standard PCM digital voice channel.

ADPCM: *See* adaptive differential pulse code modulation.

A-bit: In robbed-bit signaling, the A bit is the signaling bit that is robbed from from each DS-0 channel in frame 6.

A-law: An algorithm used in companding a pulse code modulated voice signal. The A-law algorithm is primarily used in European E-1 telephone networks.

alternate mark inversion: A form of bipolar signaling used in T-1 and E-1 transmission in which each successive mark (digital 1) is of the opposite polarity and spaces (digital 0's) have zero amplitude.

AMI: *See* alternate mark inversion

B-bit: In robbed-bit signaling, the B bit is the signaling bit that is robbed from from each DS-0 channel in frame 12.

B8ZS: *See* binary eight zero substitution

binary eight zero substitution: A modification of the AMI encoding scheme used to avoid the possible loss of synchronization by the receiver due to lack of signal transitions when a long sequence of zeros occur. A string of eight consecutive zeros in a transmitted byte is replaced by either of two B8Zs codes, depending on whether the pulse preceding an all-zeros byte is positive or negative.

The inserted code causes a bipolar violation and, as a result, is replaced by all zeros at the receiving end, thereby restoring the digital data to its original value.

bipolar return to zero: Each successive digital 1 has the opposite polarity of the previous one. This type of signal is required for transmission on both T-1 and E-1 lines.

bipolar violation:

BRZ: *See* bipolar return to zero

CAS: *See* channel associated signaling.

CCS: *See* common channel signaling

central office: A local telephone company facility where subscriber lines are linked to switching equipment that connects subscriber lines to each other, locally, or to long distance line.

channel associated signaling: A form of signaling in which the signaling information is associated with the specific circuit. In T-1 transmission, robbed-bit signaling is a form of channel associated signaling.

channel bank: A type of time division multiplexer used in T-1 and E-1 multiplexing applications.

channel service unit: A device that terminates a T-1 or E-1 line at a customer's premise. The channel service unit performs certain line coding, line conditioning, and equalization functions. This device also responds to loopback commands sent from the central office.

CO: *See* central office.

common channel signaling: A form of signaling in which the signaling for all the affected circuits is carried over a separate common circuit. An example of common channel signaling in both T-1 and E-1 transmission is the D channel used in primary ISDN signaling.

CPE: *See* customer premises equipment.

CSU: *See* channel service unit.

customer premises equipment: Terminal equipment such as computers, PBXs, and multiplexers connected to the telephone network and residing on the customer's premises.

D4: A term associated with a fourth generation T-1 channel bank. A D4 channel bank provides D4 framing which is a sequence of 12 frames known as a superframe. D4 channel banks support robbed-bit signaling.

D4 channel bank: *See* D4.

data terminal equipment: Communications devices that transmit and receive data traffic in a communications system. They may also provide interfaces to users. Examples of data terminal equipment are computers and multiplexers.

DS-0: Digital signal level 0. Standard term for a 64 kbps digital telecommunications signal or channel.

DS-1: Digital signal level 1. A standard term describing the 1.544 Mbps digital signal carried on a T-1 facility or the 2.048 Mbps digital signal carried on an E-1 facility.

DSX-1: Digital signal cross-connect level 1. Standard term for the set of parameters used where DS1 digital signal paths are cross connected.

DTE: *See* data terminal equipment.

E-1: A digital transmission facility used in Europe to primarily transmit 30 digital voice channels. The digital voice channels are time-division multiplexed into a single serial data stream of 2.048 Mbps. E-1 facilities are also used to transport a mix of digital data, compressed video, and digital voice in ISDN applications.

ESF: *See* extended superframe.

extended superframe: A sequence of 24 T-1 frames grouped together that supports ABCD robbed-bit signaling. Bit robbing is performed on frames 6, 12, 18, and 24 as opposed to superframe robbed bit signaling in which bits are only robbed from frames 6 and 12. The extended superframe format provides for more extensive signaling than the standard superframe.

FAS: See frame alignment signal.

frame: In T-1 and E-1 multiplexing, a frame is a term used to define the results of one complete multiplexer scan cycle in which a byte of data from each channel has been placed in its respective time slot for transmission over the T-1 or E-1 line. A T-1 frame consists of 24 time slots and an E-1 frame consists of 32 time slots. The multiplexed data is transmitted and received frame by frame.

frame alignment signal: The frame alignment signal is used to establish and maintain frame synchronization of the E-1 frames. This signal is comprised of bits 2 through 8 in time slot 0 of every even frame.

HDB3: *See* high density bipolar 3.

high density bipolar 3: A form of coding used in E-1 transmission to maintain ones density. In high density bipolar 3 coding, each string of four consecutive zeros in a byte is replaced by a code that causes a bipolar violation. Either of two codes is used, depending on whether an odd or even number of ones occurred since the last bipolar violation. Because the inserted code causes a bipolar violation, it is replaced by all zeros at the receiving end, thereby restoring the data to its original value.

Integrated Services Digital Network: A digital service offered by both T-1 and E-1 providers for the transport of digitized voice, digital data, and compressed video. When offered at the full T-1 or E-1 data rate, this is known as primary rate ISDN.

ISDN: See Integrated Services Digital Network.

kbps: Kilobits per second. One kilobit is equal to one thousand bits.

network interface unit: This device serves as the point of contact for customer premise equipment and identifies where network equipment begins. The network interface unit allows the carrier to conduct automated loopback tests.

NIU: *See* network interface unit.

MAS: *See* multiframe alignment signal

Mbps: Megabits per second. One megabit is equal to one million bits.

multiframe: In E-1 transmission, a multiframe is comprised of 16 consecutive E-1 frames. The E-1 multiframe provides for ABCD signaling in that time slot 16 of frames 1 through 15 conveys the signaling for two specific DS-0 channels in each frame.

multiframe alignment signal: In frame 0 of an E-1 multiframe, time slot 16 contains a pattern of four zeros in bit positions 1 through 4 that are used to identify the beginning of a multiframe. These four bits are known as the multiframe alignment signal.

PBX: *See* private branch exchange.

PCM: *see* pulse code modulation.

private branch exchange: A telephone exchange located at a user's premise that provides switching services for local extensions as well as access to the public telephone network.

pulse code modulation: A method for converting analog voice to digital voice. The analog voice signal is sampled at a rate of 8000 times per second and each sample is described digitally using an 8-bit byte. This results in a 64 kbps digital voice signal.

R2MF: An E-1 signaling system used in Europe and Asia that uses compelled handshaking on each multifrequency signaling digit.

robbed-bit signaling: An in-band signaling method used in T-1 digital voice transmission where the least significant bit is "robbed' from the 8-bit byte of each DS-0 channel's time slot in frames 6 and 12 of a superframe; and frames 6, 12, 18, and 24 of an extended superframe. The robbed bits are used to convey signaling information for each respective DS-0 channel. This is a form of channel associated signaling.

Signaling System 7: A global standard for telecommunications defined by the International Telecommunications Union (ITU). This standard defines the procedures and protocol by which network elements in the public switched telephone network (PSTN) exchange information over a digital signaling network to effect wireless (cellular) and wireline call setup, routing and control.

SS7: *See* Signaling System 7.

superframe: A sequence of 12 T-1 frames grouped together that supports AB robbed-bit signaling. Bit robbing is performed on frames 6 and 12.

T-1: A digital transmission facility used in North America and Japan to primarily transmit 24 digital voice channels. The digital voice channels are time-division multiplexed into a single serial data stream of 1.544 Mbps. T-1 facilities are also used to transport a mix of digital data, compressed video, and digital voice in ISDN applications.

TDM: *See* time division multiplexing.

time division multiplexing: A method in which digital information from lower data rate channels is bit or byte interleaved into time slots on a higher data rate stream for efficient transmission. The high speed data rate is typically the aggregate of the data rate of all the lower speed channels. In time division multiplexing, the data received on the high speed line is de-multiplexed from the individual time slots and directed to each of the lower data rate channels respectively. Time division multiplexing is used in T-1 and E-1 transmission.

time slot: In T-1 and E-1 multiplexing, a time slot is the amount of bandwidth allocated on the high speed line for an 8-bit byte of digital information from each DS-0 channel being multiplexed. That channel's time slot appears in the same sequence in each T-1 or E-1 frame.

trunk: A dedicated aggregate telephone circuit that connects two switching systems such as central offices and PBXs.

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