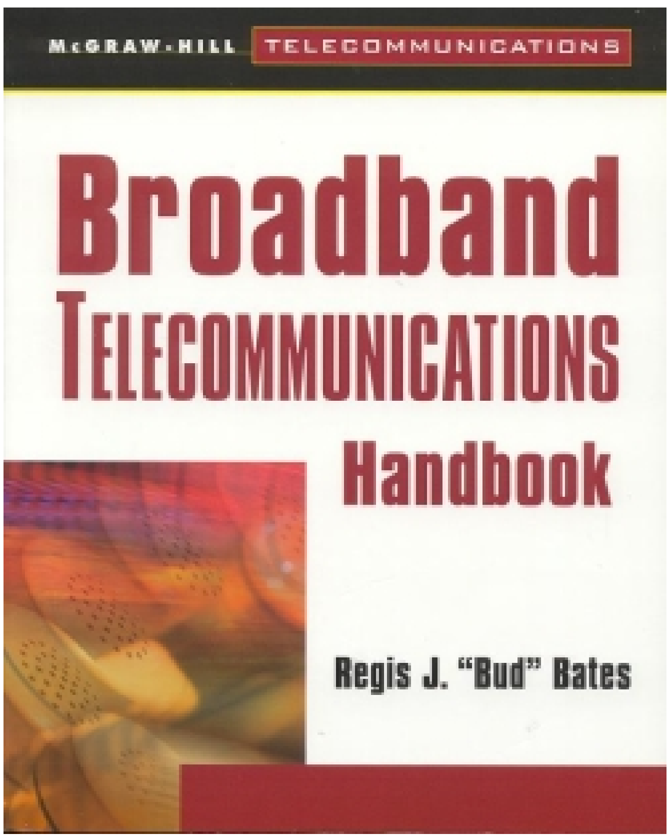
FURTHER READING:

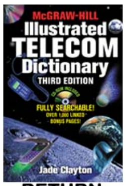
As a preview for further reading, the following reference has been provided from the pages of the book below:

Title: Broadband Telecommunications Handbook
Author: Regis J. “Bud” Bates
Publisher: McGraw-Hill

ISBN: 0071346481



Introduction



RETURN

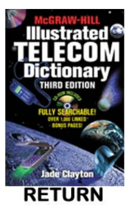
Meeting the high demands for voice, data, and multimedia communications, the larger corporations in the world are moving to the high-speed digital communications of the T Carrier system. This includes many of the copper and optical fiber-based systems. The hottest installation method today is still the T1. Although T1 has been around since 1958 when it was first created, and then rolled out into the carrier communities in 1960, one would think this technique became old. Yet, in 1998 many of the long distance carriers announced that they were experiencing a shortage of T1 ports, and installation times moved from 6 days *after receipt of order* (ARO) to approximately 120–150 days ARO. This supports the statement that T1 is still the hottest method of connecting some of the services discussed in other chapters, such as:

- Frame Relay
- ATM
- Point-to-point data lines
- Internet access
- Signaling systems
- Computer-to-Telephony Integration
- ISDN
- xDSL (HDSL, CDSL)

Why has this become such a popular method? The implementation of digital high-speed communications can be directly linked to the cost and convenience factors. T1 became a very price competitive access method for nearly all the communications services needed.

Yet, there has been other activity in the form of T3 or DS3 services to support the higher demands for the various multimedia and video applications. In both cases, T1 and T3 are now very popular in the commercial industry, as a means of providing the connectivity desired to meet the continuing communications development within organizations. The pricing models now make these services more readily affordable. Moreover, the enhancements to the digital circuits make this a tried and true technology. For example, today's tariffs make both T1 and T3 very price efficient and attractive. Even if a customer does not have enough volume to justify the use of a full T1 or T3, the prices allow the consumer to use portions of the services while holding excess capacity for new applications, without causing undue budgetary hardships. A T3, for example, costs the same as between 5–8 T1s terminated at the same location. This is beyond a financial decision and moves us to think of these services as application supportive.

Once a consumer or organization makes the "leap" into the digital circuit, the availability of bandwidth to host other services that were heretofore unavailable becomes paramount. Newer graphics and video services can be turned on, as needed, if the bandwidth is no longer the constraint.



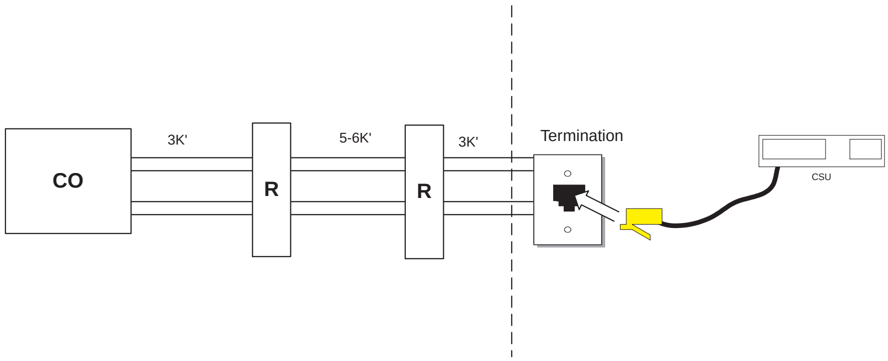
This chapter will discuss the way the multiplexing and access of the digital services is facilitated. There are many other excellent publications that go deeper into the technologies. The reader can choose to do further study if so desired.

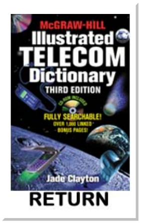
The Difference Between T-x and DS-x

In many cases, the terms become problematic because we tend to intersperse them. Therefore, when discussing the different services, it becomes imperative to discuss the difference between a T-x and a DS-x.

T-x (such as T1 and T3) refers to the services acquired from a carrier or local provider by physical layers one and two. The T carrier is a physical set of wires, repeaters, connectors, plugs and jacks, etc. When referring to a T1, we are actually describing the physical layer interface to the provider networks. A T1, for example, is a four-wire circuit (unless one of the HDSL or SDSL techniques is used with only one pair of wires). The four-wire circuit is installed between the local provider and the customer's premises, as shown in Figure 26-1. In this example, the provider will install the physical wires, or use four wires from a 50 pair bundle on the local loop. From the Central Office, the four wires will be cleaned up to remove splices, bridge taps, and load coils from the wires. At approximately 3,000 feet from the egress point at the CO, the provider will install a digital repeater (or regenerator, as it is also called). Thereafter, every 5,000–6,000 feet, another repeater is required until the last leg of the circuit. At approximately 3,000 feet from the customer's entrance point, the last repeater is installed. The provider then terminates the circuit at the demarcation point in a jack. The customer uses a plug to connect the CPE to the circuit. These are all mechanical and electrical devices allowing the installation of the T carrier.

Figure 26-1
A typical T1 carrier installation





After the circuit is installed and the carrier is terminated, the customer then generates traffic (voice, data, video, etc.) across the circuit. This traffic is carried on the carrier at a digital rate, or what is called the *Digital Signal* level 1 (DS-1). DS-1 operates at a Digital Signaling rate of 1.544 Mbps. The traffic is therefore called a DS-1. (Note in many cases this may be annotated as DS1.)

One can therefore see that there is a difference between the physical carrier and the signaling rate of the traffic carried on the circuit.

DS-1 Framing Review

DS-1 operates in a framed format (some cases exist where a customer may request a nonframed format, but this is rare). The framer establishes the frame in a device called a *Channel Service Unit* (CSU). A frame consists of 193 bits of information created in a 125 µseconds. In a DS-1 there are 24 time slots associated with the signal. Each time slot carries 8 bits of user information. When the CSU sets up the frame and has 24 slots of 8 bits each, it then adds one overhead bit for framing delimitation. This creates the 193 bits referred to above.

24 channels @ 8 bits per channel + 1 framing bit = 1 frame

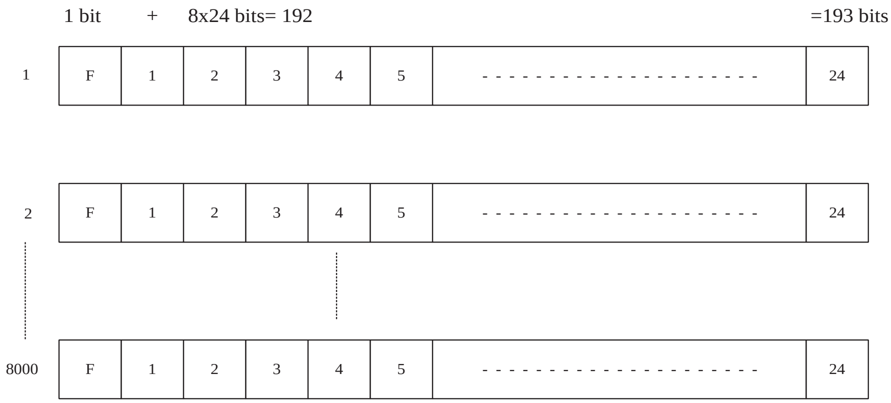
24 × 8 + 1 = 193 bits/frame

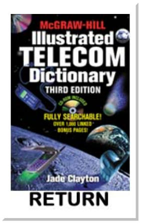
Continuing the math, the 193 bit frames are generated 8,000 times per second:

193 bits/frame × 8000 frames/sec. = 1.544 Mbps

This framing and formatting is shown in Figure 26-2.

Figure 26-2
A frame of information occurs 8,000 times/second



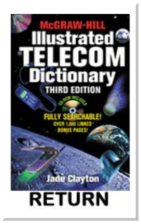


The frames of information are serially transmitted across a four-wire circuit, and operate full duplex. This gives the end user a 1.544 Mbps in each direction simultaneously.

Pulse Coded Modulation (PCM)

Before going any further into the DS-1, it may be appropriate to review the modulation technique used to create the digital signal. When DS-1 was first created, it was designed around converting analog voice communications into digital voice communications. To do that, voice characteristics were analyzed. What the developers learned was that voice operates in a telephony world in a band-limited channel operation. The normal voice will produce both amplitude and a frequency change ranging from 100 to approximately 5,000 times a second. These amplitude and frequency shifts address normal voices. However, the telephone companies decided, long ago, that carrying true voice would be too expensive and not provide any real added value to the conversation. They then determined that the normal conversation from a human actually carries the bulk of the information when the frequency and amplitude shifts vary between 100 and 3,300 times per second. Armed with this information, the developer determined that reasonable and understandable voice could be handled when carried across a band-limited channel operating at 3,000 cycles of information per second (what was termed as a 3 kHz channel). Taking all the electromagnetic spectrum available to them, the developers then channelized the frequency spectrum (in *Radio Frequencies* [RF] and in electrical spectrum available on the cabling systems they had) to smaller capacities. The norm was set at 4 kHz channels. This is the foundation of the voice telephone network.

From there, the providers of the infrastructure (the telephone companies) placed bandpass filters on the facilities to limit the amount of electrical information that could pass across their wires (or any other communications facility). Using a standard 4 kHz channel, they limited the bandpass to no more than 3 kHz (see Figure 26-3).



Following upon this thought, the developers then wanted to convert the analog communications to a digital format. Here they studied the way the user modulates voice conversation. Now they had something to work with. The voice modulated at a normal rate operates in the 3 kHz range, but now must be converted to a digital signal consisting of 1s and 0s. Using the Nyquist theorem from 1934, the developers used a formula to convert the continuously changing amplitude and frequency shifts to a discreet set of values represented by the 1s and 0s. Using a three-step process, as shown in Table 30-1, they determined that they could carry digital voice. The table represents the three steps followed to do this conversion.

Once the values were determined, the developers used another process. Setting the values in place, they had to determine where along the wave the sample fell. They created a value system to show the 256 levels by using the table shown in Table 30-2. First, the digital PCM signal is representative of both positive and negative values. To reflect where along the wave the sample fell, the 8th bit in each sample is used as a sign (+/-) value. The other seven bits therefore represent the actual sample value. There are 128 points on the positive side of the wave and 128 points on the negative side of the wave. In this table, we look at the major stepping points. Two different formats are shown in the table, the PCM used in North America (and Japan) is Mu-Law, whereas the rest of the world uses an A-Law method. These are different as shown.

Figure 26-3 Band-limited channel limits the amount of information carried to 3 kHz

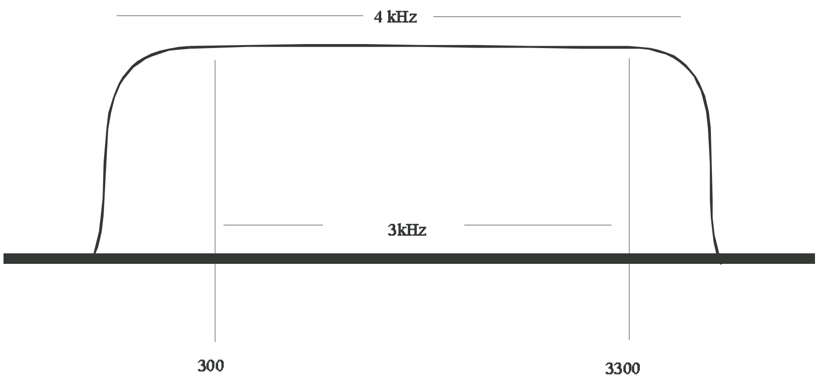


Table 26-1

The three-step process used to create the digital signal

Process	Result
1. Sample the analog wave at twice the highest range of frequencies that can be carried across the line. Using a 4 kHz channel capacity, the highest range of frequencies allowable is 4,000.	A sampling rate of 8,000 times per second: **4,000 × 2 = 8,000**
2. Quantify the values using a logical pattern of 1s and 0s to represent the height of the signal at any point in time. This deals with the amplitude shifts only.	Using an 8-bit sequence, the result is a total of 256 combinations of amplitude that can be represented. Although there may be more amplitude heights, (values) the 256 quantities were determined to be sufficient. **2^8 = 256 possible combinations**
3. After the values are determined from the samples, the final step is to encode the signal into a digital format and transmit information in its digital format onto the wires.	A sample value of 5 on the positive side of the wave will then be represented as an 8-bit data stream. **Binary 5 = 00000101**

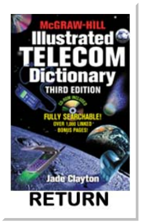
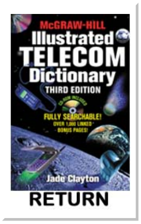


Table 26-2

Summary of values for PCM in Mu-Law and A-Law formats

Coded Numerical Value	Bit Number		Comments
	Mu-Law	A-Law	The left-most bit (bit number 1) is transmitted first. It is the most significant bit. This bit is used as the sign bit; it is a 1 for positive values, and it is a 0 for negative values.
	12345678	12345678	
+127	10000000	11111111	
+ 96	10011111	11100000	
+ 64	10111111	11000000	
+ 32	11011111	10100000	
0	11111111	10000000	
0	01111111	00000000	
- 32	01011111	00100000	Note that 0 has two different values. Bits 2–8 are inverted between A-Law PCM and Mu-Law PCM. In A-Law all even bits are inverted prior to transmission.
- 64	00111111	01000000	
- 96	00011111	01100000	
- 126	00000001	01111110	
- 127	00000000	01111111	

The E-1 Pattern



Because the A-Law was shown in the table for comparative purposes, it is also wise to introduce the concept of the E-1. In North America, the DS-1 is the way we use our digital services. However, most of the world adopted a different standard called the E series. The E-1 is the European equivalent of a DS-1. There are many differences between the two that make them incompatible. The first is the companding method of creating the digital values as shown in the previous table. Above and beyond the companding values (using A-Law), there are differences between the formats used. An E-1 is a digital transmission rate that operates at 2.048 Mbps compared to the DS-1, which operates at 1.544 Mbps. Moreover, the E-1 carries 32 channels of 64 Kbps, whereas the DS-1 only carries 24 channels of 64 Kbps. Other multiplexing forms create higher speed E series just as we do with the DS signals. The following table summarizes the DS-n series compared to the E-n series. See Table 30-3 below for the channel speeds of the digital transmission systems.

Table 26-3

Comparison of North American and European digital services

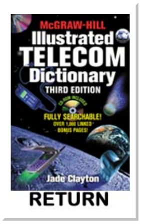
DS-n	Channels	Speed	E-n	Channels	Speed
DS-0	1	64 Kbps	E-0	1	64 Kbps
DS-1	24	1.544 Mbps	E-1	32	2.048 Mbps
DS-2	96	6.312 Mbps	E-2	128	8.448 Mbps
DS-3	672	44.736 Mbps	E-3	512	34.368 Mbps

The Framing Protocols: D4 Framing

Before transmitting the frames across the circuit, the CSU device uses protocols that describe other activities. One protocol is a framing convention to preserve timing, signaling, and check for errors. The older version of this protocol is called D4 framing. D4 is comprised of 12 samples from each of the 24 inputs. The samples are held for a short time in a buffer by the CSU.

When all 12 passes are completed, the CSU then places the framing bit on each of the frames and transmits the information across the link.

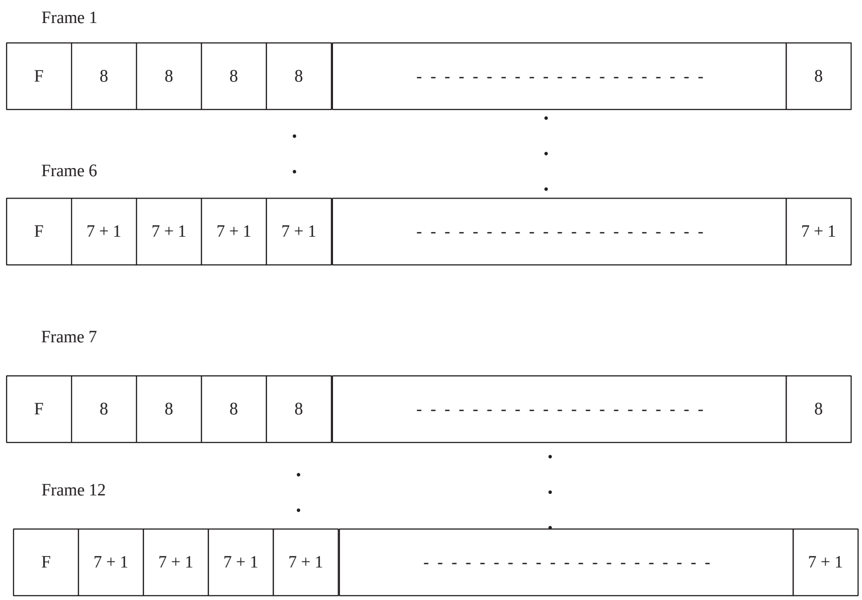
When preparing the 12 frames, however, the CSU will also do a few other things. In the 6th and 12th frames, the CSU will steal (rob) one bit from each of the 24 inputs and use this bit for signaling. This means that the CSU steals the least significant bit from the sample and reassigns it. The unfortunate result of this framing convention is that it relegates the end user to only getting 7 data bits and 1 signaling bit every sixth frame. The 7 bits are trusted, whereas the eighth bit is suspect for use as data (because it is signaling). This results in a data carrying capacity of 56 Kbps (7 bits times 8,000 frames/sec). This is shown in Figure 26-4, using the robbed bit signaling approach.



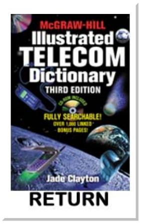
Contrasting the E-1 and DS-1 Frame

Once again differences exist in the format of the framing patterns for the E-1 and the DS-1. Whereas the DS-1 is shown with 24 channels operating at 64 Kbps, plus a framing bit (1 bit added to the 24 byte samples) to create a frame of 193 bits, the E-1 uses 32 channels of 64 Kbps. However, the first channel of an E-1 frame (time slot 0) is used for synchronization, alarms

Figure 26-4
Robbed bit signaling in the D4 framing format



and international carrier use combined. This means that only 31 channels are available to carry voice or data traffic. This frame format is shown in Figure 26-5. The use of the pattern shown in this frame format uses Timeslot 0 (TS0) for framing and synchronization. Then TS16 is used for signaling for the various channels in the payload, whereas the DS-1 uses robbed bit signaling as discussed previously or common channel signaling on TS24. These differences are significant enough to prevent an even mapping from a DS-1 to an E-1.



Extended Superframe Format (ESF)

As a means of overcoming the limitations of the D4 framing protocols, the industry developed a newer protocol called ESF. This is an extension of the 12-frame format in order to double the size in the buffers at the CSU. When

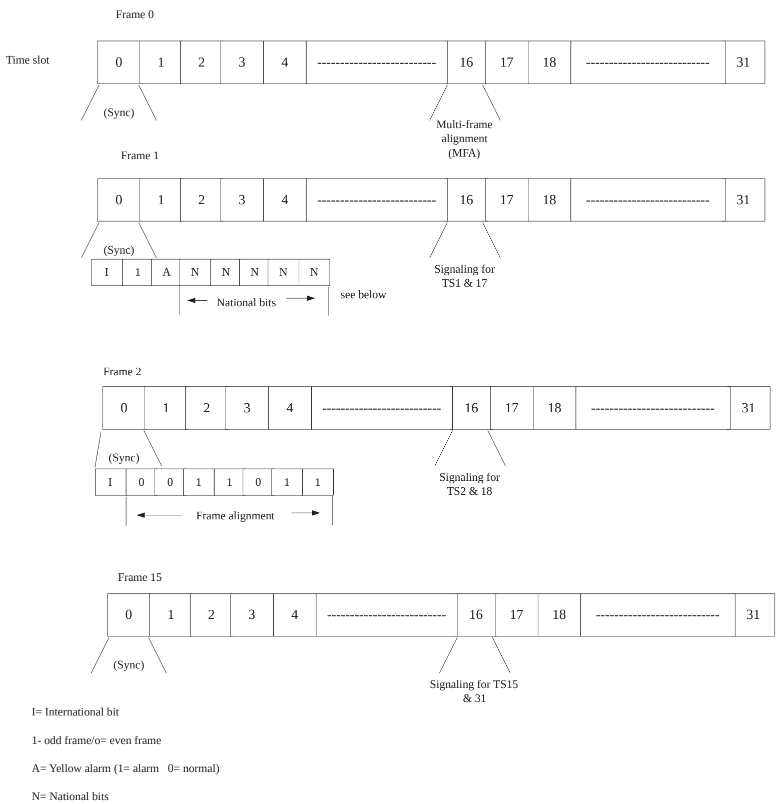
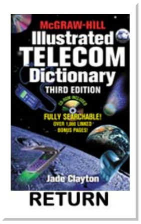


Figure 26-5
An E-1 frame format

the CSU has 24 frames of information, the CSU will use the framing bit for more than just a locator bit. It will define where the signaling is handled, and use this framing bit for other things such as the following: (Note: Remember there are 8,000 framing bits per second)

- 2 K bits are used for framing
- 2 K bits are used for error checking (CRC−6)
- 4 K bits are used for a maintenance and diagnostic capability to troubleshoot the circuit and improve availability.



The framing format of the ESF is shown in Figure 26-6.

After the framing bits had been addressed, another choice was used to handle the T1 circuit. As described previously, the robbed bit signaling gets in the way of data transmission. It is not a problem for voice communications, but definitely impacts the data side of the business. To solve this problem, a technique called common channel signaling was introduced. The use of channel number 24 was assigned for strictly signaling (call setup and teardown) for the other 23 channels. By using a dedicated out-of-band signaling channel, the other 23 channels can carry all 64 Kbps for data, yielding a higher throughput per channel, but with a penalty of losing one channel from the T1. The format of the common channel signaling arrangement is shown in Figure 26-7. This is a choice a user has to make. If all the user wants is voice communications, the robbed bit signaling will suffice. If, however, the user wants to transmit data, then the common channel signaling will be a potential benefit to the data transmission.

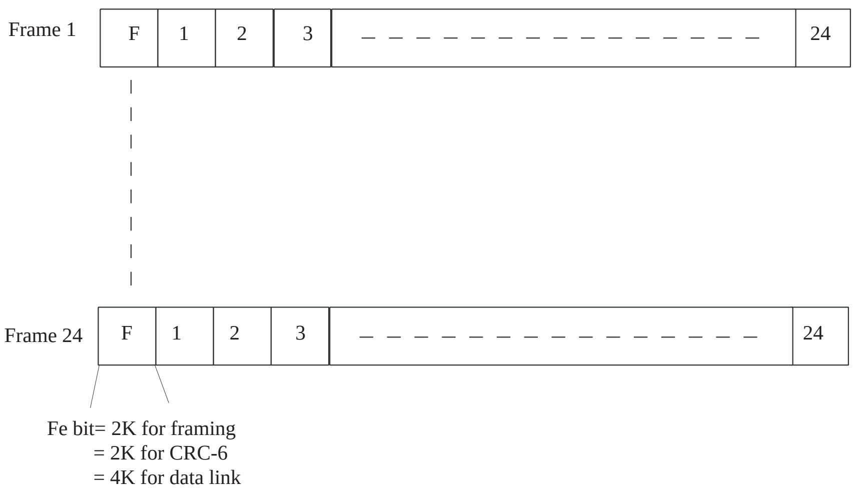
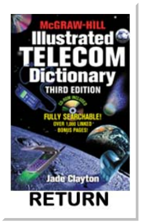


Figure 26-6
Extended Superframe Format improves uptime and error checking

Figure 26-7
Common channel signaling uses channel number 24

	TS1	2	3	4		23	Comm Channel
F	8	8	8	8	-------------------------------	8	SIG



Other Restrictions

Another problem surfaced when T1 was introduced for data transmission. In order to synchronize the carrier services, T1 uses a byte synchronization plan. In every eight bits there must be at least one pulse (represented as a digital one) for the transmitters to derive their timing on the line. If voice is the application, there are minimal problems with this constraint. As long as a party on the conversation is talking, there are sufficient numbers of ones pulses transmitted to keep the line synchronized. Yet, when data is transmitted, there may be strings of zeros transmitted continuously. This string of zeros will cause the transmitters to drift and lose timing.

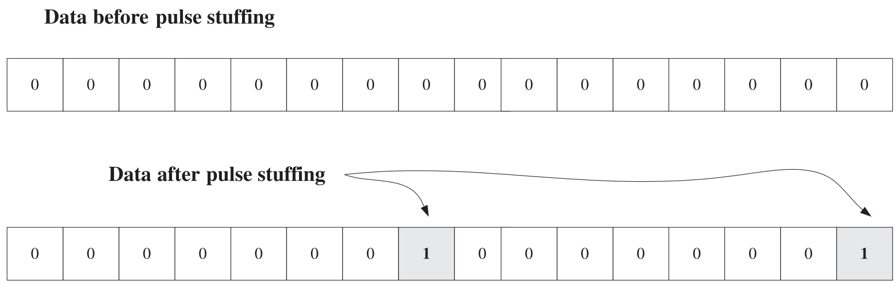
To solve this problem, the Bell system introduced a "ones density requirement," which states that in every eight bits transmitted there must be at least one digital one. Further, no more than 15 consecutive zeros may be transmitted in a row; otherwise the timing will be lost. The solution to meeting this rule is handled in the CSU. If the CSU receives 8 zeros in a row, it will strip off the least significant bit and substitute a one in its place. This will meet the requirements of the ones density.

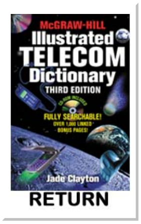
However, when the zero is stripped off and a one substituted in its place (called pulse stuffing), there is no way to know when this is done or when it is not. Consequently, this solution meets the timing and synchronization demands of the network, but leaves the user with the risk that the 8th bit is wrong. This leads the user to only trusting the first 7 bits, but ignoring the 8th bit. Ultimately, the result is 7 usable bits instead of 8 or a 56 Kbps throughput on the line instead of 64 Kbps. (7 bits times 8,000 samples = 56 Kbps). This results in a lot of wasted bandwidth and a limited throughput for data transmission. The pulse stuffing mechanism is shown in Figure 26-8.

B8ZS

It did not take long before the user began to demand better efficiency of the T1 line. Therefore, a newer technique is used called *Bipolar* (or binary) *8 Zero Substitution*. With B8ZS, the CSU is responsible for substituting a

Figure 26-8 Pulse stuffing allows the timing to be preserved





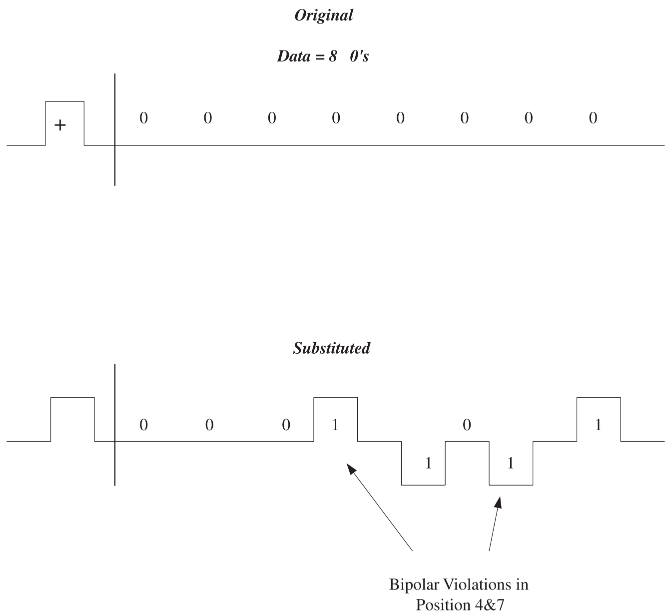
long string of zeros, eight at a time, with a fictitious 8 bits to meet the demands of the line synchronization, while at the same time allowing the receiving device to recognize that 8 zeros were originally sent. The B8ZS inserts a bit pattern of 0001 1011, easily satisfying the demands of the synchronization plan. By inserting these bits, the 4th and the 7th bits will be set as violations to the bipolar rule (*alternate mark inversion* [AMI]). Receiving a bit pattern of 0001 1011 with a bipolar violation in positions 4 and 7 are recognized as a flag that 8 zeros were intended. Therefore, the receiving CSU will strip off the fictitious word and reinsert all zeros to the receiving device. This allows for clear-channel 64 Kbps data transmission. The B8ZS is shown in Figure 26-9.

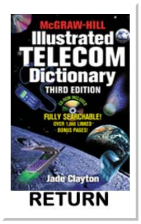
With all these problems and solutions, one can only wonder why everyone uses the T1. The primary reason is the 1.544 Mbps throughput (or slightly less based on the restrictions) for voice and data transmission. The second reason is the cost/benefits ratio associated with bundling the services onto a single, four-wire circuit. Third, is the benefit of all digital circuitry to the door. It is because of these three reasons that T1 is the most widely used digital transmission service in North America.

T-2 Transmission (or DS-2)

For all its work, the T1 was a beginning step to the North American Digital Hierarchy. From there, other multiplexing steps were instituted to get higher speed communications services. A second means of achieving the high-speed communications is to multiplex four DS-1s into a DS-2. The DS-2 (physically it is a T-2) is a data channel carrying 96 channels of 64 Kbps each, plus overhead yielding a nominal 6.312 Mbps data transmission speed. To create the DS-2, four DS-1 signals are combined. This creates a

Figure 26-9 B8ZS inserts an 8-bit pattern that is recognizable by the receiver as substituted





DS-2 frame as shown in Figure 26-10. The DS-2 frame is often referred to as the DS-2 M frame and consists of four subframes. Each of the subframes are labeled: M1 though M4. A subframe consists of six blocks of information; each block contains 49 bits.

The first bit in each block is an overhead bit. A DS-2 frame has 24 overhead bits. The remaining 48 bits in each block are DS-1 information bits. Carrying this out then there is:

48 DS1 bits/block X 6 blocks/subframe X 4 subframes/ DS-2 frame = 1,152 DS-1 information bits

The four subframes do not represent each of the separate DS-1 signals; instead a bit interleaving of the four DS-1 signals forms the DS-2 frame.

The overhead bits precede the data bits in each of the blocks. The data bits are interleaved, where 0i designates the time slot devoted to DS-1 input i. After 48 information bits, 12 from each of the DS-1s, a new DS-2 overhead bit is inserted. The total number of DS-1 bits transmitted per second in a DS-2 frame is therefore:

DS-1 rate	×	**4 DS-1 signals/DS2**	**= information bits**
1,544 Mbps	×	**4 DS-1 signals**	**= 6.176 Mbps**

The rate chosen for the DS-2 is 6.312 Mbps. This allows extra overhead for bit stuffing to synchronize the signals.