
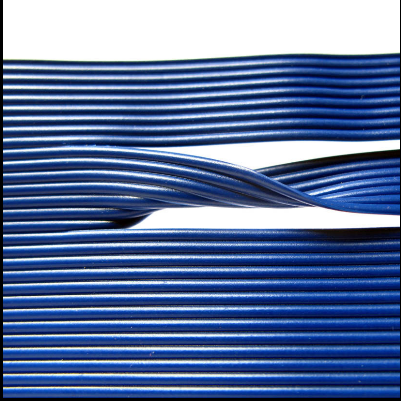
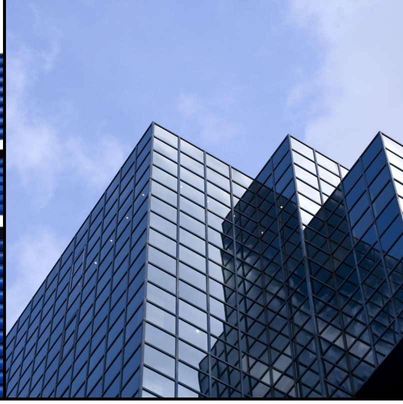

Installation and Maintenance of E1 circuits

ALBEDO AT-2048 is a rugged fully featured battery operated E1/Datacom handheld tester designed in 2010 providing easy navigation and high resolution screen. Low cost, fully featured, it is a truly perfect field tool for installation, acceptance and maintenance of PDH and Datacom links including bi-directional (BER) test functions.

A valuable tool that offers generator, dual analyzer, USB, Ethernet, and RJ45 interfaces. It offers Jitter measurements and pulse mask, therefore it can monitor slots activity, delay and frequency measurements frequency measurements over more than seven hours. Test results can be saved in a Memory stick or transferred to a PC.

ALBEDO Telecom

Installation and Maintenance of E1 circuits

1 An Introduction to Communications Systems

One of the first communications networks known was built by Mediterranean cultures more than 1,000 years ago and consisted of a series of successive towers with a distance of about 5 to 12 km between them. A message could be coded and transmitted from the first tower to the second one by using optical signals, and then be passed on along the line until it reached its final destination.

In this primitive system we can already identify all the elements of a genuine communications network (see Figure 2):

- *Information* consists of the messages interchanged between final users. In order to be introduced into the network, information needs to be coded into signals.
- *Signals* are a physical magnitude, specific for each transmission medium, that change with respect to time.
- The *transmission medium* consists of the links that connect distant nodes.
- *Nodes* are those network elements that receive the signals and retransmit them further along until reaching the final users.

Figure 1 ALBEDO AT.2048, E1, Datacom, Jitter, and Wander tester.

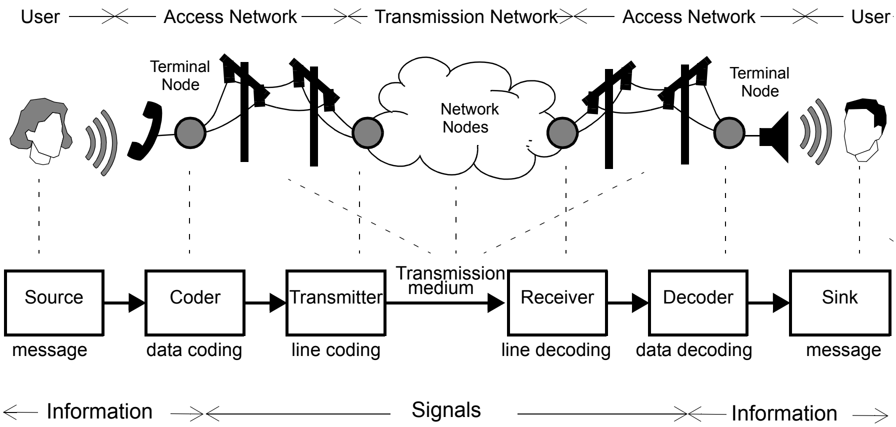

Figure 2 Elements of a telecommunications network.

In other words, in a telecommunications network, user information is distributed as signals from one point to another through the transmission medium that connects the nodes in the system.

1.1 Signals and Information

The messages to be transmitted are meaningful for the users and are structured hierarchically in lexical, syntactic, and semantic layers, in line with the grammar of the natural language used, whereas signals, by comparison, are only meaningful inside the telecommunications network. The signals used in telecommunications systems can be of two types (see Table 1):

1. *Analog or continuous*: They can take any of an unlimited number of values within a given range.
2. *Digital or discrete*: They can only take a limited number of values. In a binary system, the only valid values are 0 and 1.

Table 1
Combinations of signals and information.

Signal	*Analog Information*	*Digital Information*
Analog	Modulation (e.g., AM/FM radio and TV)	Digital modulation (e.g., ADSL)
Digital	Digitalization (e.g., audio CD, GSM)	Coding (e.g., frame relay)

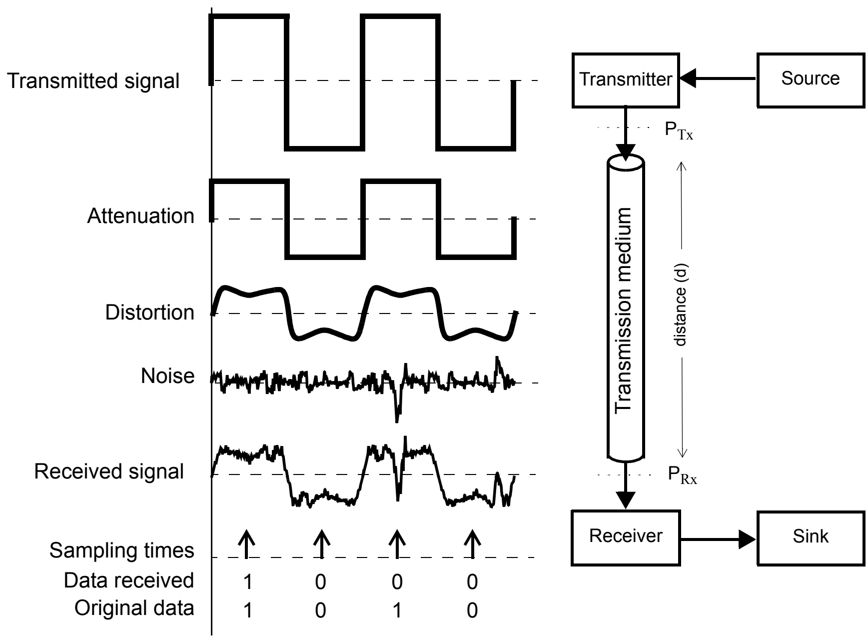

Figure 3 Effects of attenuation, distortion, and noise on transmission.

1.2 Transmission Medium

The transmission medium can be defined as the environment where a signal is transmitted, be it material (electrical wires, optical fiber, open air, etc.) and nonmaterial, or vacuum, through which only electromagnetic waves are propagated.

The material transmission medium can be divided into two main groups:

1. A *conductive medium*, in which the information is transmitted in the form of electrical impulses. Typical examples of this medium are twisted-pair and coaxial cables.
2. A *dielectric medium*, in which the information is transmitted in the form of radioelectrical or optical signals; for example, the atmosphere and optical fiber.

The propagation of signals over one of these media is what we call transmission. The success of transmission of information in telecommunications networks depends basically on two factors: the quality of the signal transmitted, and the quality of the transmission medium used. In addition, there are natural forces that can resist transmission and modify the original characteristics of the signals, which may end up being degraded by the time they reach their destination.

The most significant impairments are attenuation, noise, and distortion. We look at these below in respect to a communications channel, which is defined as a means of unidirectional transmission of signals between two points.

1.2.1 Attenuation

Attenuation weakens the power of the signal proportionally to the transmission medium length. It is expressed in decibels (A_{dB}) through the logarithmic ratio of the transmitted power (P_{Tx}) and received power (P_{Rx}), measured at both ends of the distance (d) being examined (see Figure 3). Transmission media can usually be characterized by their attenuation per unit of length ($A_{dB / Km}$):

$$10log(P_{Tx}/P_{Rx}) = d \cdot A_{dB/Km}$$

$$A_{dB} = d \cdot A_{dB/Km}$$

Example: Thus for a transmission medium with A=0.2 dB/Km, after 15 Km, the attenuation is A_{dB}=3 dB. If the transmitted power is P_{Tx}=1W. After 10 Km received power is P_{Rx}= 0.5W, because 10 log (1/P_{Rx}) = 3 dB (see Figure 4).

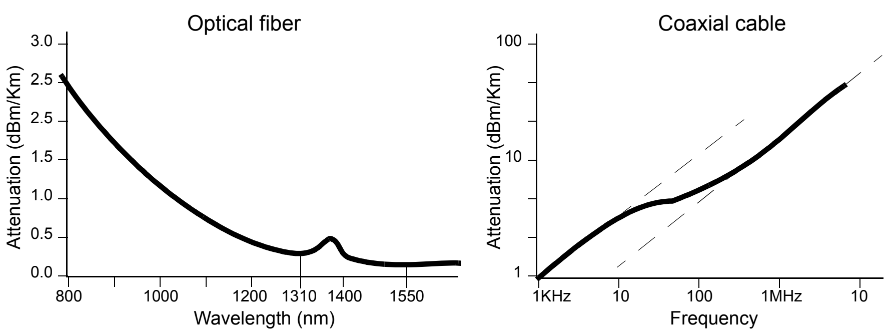

Figure 4 Typical attenuation values for single mode optical fiber and coaxial cable.

At the far end the received signal must have enough power (P_{Rx}) to be interpreted, otherwise amplifiers (also known as repeaters or regenerators in digital transmission) must be inserted along the transmission medium to improve the power of the received signal.

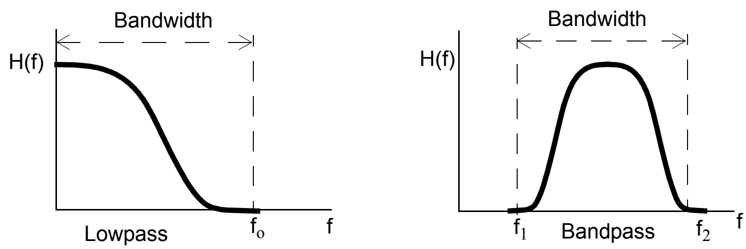

Figure 5 The two basic transmission channels. In the frequency domain the channel transfer function H(f) determines the attenuation of each frequency and consequently the amplitude distortion.

1.2.2 Distortion

Distortion produces a change in the original shape of the signal at the receiver end. There are two types: amplitude distortion and delay distortion.

- When the impairments affect the amplitudes of the frequency components of the signal differently, this is said to produce *amplitude distortion* (sometimes called absorption). Amplitude distortion is caused because the transmission channel is limited to certain frequencies (see Figure 5). To overcome this problem amplifiers must equalize the signal, separately amplifying each band of frequencies.[1]
- When the velocity of propagation of a signal varies with the frequency, there is said to be *delay distortion* (sometimes called dispersion). Delay distortion is particularly disturbing in the digital transmission producing *intersymbol interference* (ISI), where a component of the signal of one bit is misplaced in the time slot reserved for another bit. ISI limits the capacity to extract digital information from the received signal.

Harry Nyquist showed that the maximum transmission capacity (C) is limited by ISI and depends on the channel bandwidth (B) and the number of signal elements (M) coding the information.

$$C_{bps} = 2B\log_2 M$$

1. Note that attenuation is a specific case of amplitude distortion that equally affects all frequencies of the signal.

Example: For a modem using 16 signal elements and a channel bandwidth (B) of 4,000 hertz (Hz), the maximum data transfer rate (C) is 32,000 bits per second (bit/s).

1.2.3 Noise

Noise refers to any undesired and spurious signal that is added to an information signal. It is usually divided into five categories:

1. *Thermal noise:* This is caused by the agitation of electrons in any conductor in a temperature different than absolute zero. The noise (N) is independent of the frequency and proportional to the bandwidth (B) and the temperature (T) in degrees Kelvin:

   $$N = k \cdot T \cdot B$$

   (k is the Boltzmann's constant in joules/kelvin, k = 1.3803 x 10^{-23})
2. *Intermodulation noise:* This is caused when two or more signals of frequencies f_1 and f_2, transmitted in the same medium, produce a spurious signal at frequencies that are a linear combination of the previous ones.
3. *Atmospheric noise:* This is caused by the static discharge of clouds, or ionized gas from the sun, or high frequency signals radiated by the stars.
4. *Impulse noise:* Of short duration but high amplitude, these energy bursts are caused by sources such as electrical machinery, a drop in voltage, atmospheric interference, and so on. These do not tend to be a problem for analog signals, but are a prime cause of errors in digital transmission.
5. *Crosstalk:* Whenever a current flows through a conductor a magnetic field is set up around it that can induct a current into a second conductor collocated in a short distance.

Noise is always present in transmission channels, even when no signal is being transmitted. A key parameter at the receiver end to distinguish between information and spurious power is the signal-to-noise ratio (S/N):

$$(S/N)_{dB} = 10log(Power_{Signal}/Power_{Noise})$$

Claude Shannon proved that the signal-to-noise ratio (S/N) determines the theoretical maximum transmission capacity (C) in bits per second of channel with a limited bandwidth (B):

$$C_{bps} = Blog_2(1 + S/N)$$

Example: A typical value of S/N for a voice grade line is 30 dB (equivalent to a power ratio of 1,000:1). Thus for a bandwidth of 3,100 Hz the maximum data transfer rate (C) should be 30,894 bit/s.

If we pay attention only to the Nyquist formula (see Section 1.2.2) we could inaccurately conclude that for a given bandwidth (B) the data rate can be increased endlessly, by increasing the number of signal elements. However in reality, the signal-to-noise ratio sets up the theoretical limit of the channel capacity.

The Shannon theorem makes no statement as to how the channel capacity is achieved. In fact, channels only approach this limit. The task of providing high channel efficiency is the goal of coding techniques.

1.2.4 The transmission channel

A digital channel is a communication subsystem with capacity to send and receive information between two points: a source and a sink. Related concepts are:

- *Bandwidth*, expressed in hertz (Hz). This is the difference between the highest and the lowest frequency that can be transmitted across a line or a network.
- *Data rate*, expressed in bits per second (bit/s). This is a measure of the speed with which information is transferred. It depends on the bandwidth, transmission medium impairments, and the technological capacity to efficiently use the available bandwidth.
- *Performance*, expressed in bit error rate (BER). This is the probability of a single bit being corrupted in a defined interval. Performance is on indication of the quality of the channel.

Channel capacity is the data rate that can be transmitted over a communication path under specific conditions.When two channels define a two-way communication, it is more usual to talk about a circuit.

1.3 Channel Coding

Channel coding is the process that transforms binary data bits into signal elements that can cross the transmission medium. In the simplest case, in a metallic wire a binary 0 is represented by a lower voltage, and a binary 1 by a higher voltage. However, before selecting a coding scheme it is necessary to identify some of the strengths and weaknesses of line codes:

- *High-frequency* components are not desirable because they require more channel bandwidth, suffer more attenuation, and generate crosstalk in electrical links.

- *Direct current* (dc) components should be avoided because they require physical coupling of transmission elements. Since the earth/ground potential usually varies between remote communication ends, dc provokes unwanted earth-return loops.
- The use of *alternating current* (ac) signals permits a desirable physical isolation using condensers and transformers.
- *Timing control* permits the receiver to correctly identify each bit in the transmitted message. In synchronous transmission, the timing is referenced to the transmitter clock, which can be sent as a separate clock signal, or embedded into the line code. If the second option is used, then the receiver can extract its clock from the incoming data stream thereby avoiding the installation of an additional line.

In order to meet these requirements, line coding is needed before the signal is transmitted, along with the corresponding decoding process at the receiving end. There are a number of different line codes that apply to digital transmission, the most widely used ones are alternate mark inversion (AMI), high-density bipolar three zeros (HDB3), and coded mark inverted (CMI).

1.3.1 Non-return to zero

Non-return to zero (NRZ) is a simple method consisting of assigning the bit "1" to the positive value of the signal amplitude (voltage), and the bit "0" to the negative value (see Figure 6). There are two serious disadvantages to this:

1. No timing information is included in the signal, which means that synchronism can easily be lost if, for instance, a long sequence of zeros is being received.
2. The spectrum of the signal includes a dc component.

1.3.2 Alternate mark inversion

Alternate mark inversion (AMI) is a transmission code, also known as pseudoternary, in which a "0" bit is transmitted as a null voltage and the "1" bits are represented alternately as positive and negative voltage. The digital signal coded in AMI is characterized as follows (see Figure 6):

- The dc component of its spectrum is null.
- It does not solve the problem of loss of synchronization with long sequences of zeros.

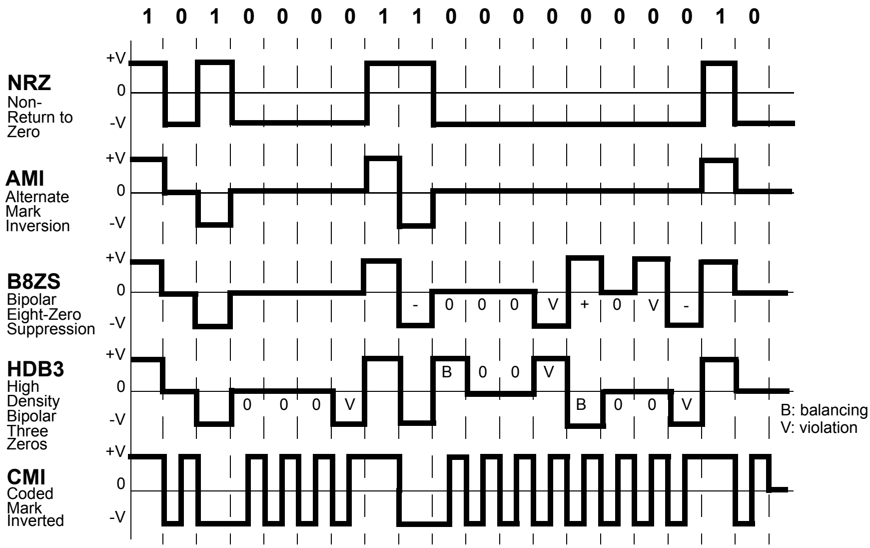

Figure 6 Line encoding technologies. AMI and HDB3 are usual in electrical signals, while CMI is often used in optical signals.

1.3.3 Bit eight-zero suppression

Bit eight-zero suppression (B8ZS) is a line code in which bipolar violations are deliberately inserted if the user data contains a string of eight or more consecutive zeros. The objective is to ensure a sufficient number of transitions to maintain the synchronization when the user data stream contains a large number of consecutive zeros (see Figure 1.5 and Figure 1.6).

The coding has the following characteristics:

- The timing information is preserved by embedding it in the line signal, even when long sequences of zeros are transmitted, which allows the clock to be recovered properly on reception
- The dc component of a signal that is coded in B8Z3 is null.

1.3.4 High-density bipolar three zeroes

*High-density bipolar three zeroe*s (HDB3) is similar to B8ZS, but limits the maximum number of transmitted consecutive zeros to three (see Figure 6). The basic idea consists of replacing a series of four bits that are equal to “0” with a code word

B8ZS

		Substitution
Last pulse polarity	+	$000V^+{-}0V_-{+}$
	–	$000V_-{+}0V^+{-}$

HDB3

		Number of ones	
		Odd	Even
Last ‘1’ polarity	+	B_-00V_-	$000V^+$
	–	$000V_-$	B^+00V^+

Figure 7 B8ZS and HDB3 coding. Bipolar violations are: V^+ a positive level and V_- negative.

“000V” or “B00V,” where “V” is a pulse that violates the AMI law of alternate polarity, and B it is for balancing the polarity.

- “B00V” is used when, until the previous pulse, the coded signal presents a dc component that is not null (the number of positive pulses is not compensated by the number of negative pulses).
- “000V” is used under the same conditions as above, when, until the previous pulse, the dc component is null (see Figure 7).
- The pulse “B” (for balancing), which respects the AMI alternation rule and has positive or negative polarity, ensuring that two consecutive “V” pulses will have different polarity.

1.3.5 Coded mark inverted

The *coded mark inverted* (CMI) code, also based on AMI, is used instead of HDB3 at high transmission rates, because of the greater simplicity of CMI coding and decoding circuits compared to the HDB3 for these rates. In this case, a “1” is transmitted according to the AMI rule of alternate polarity, with a negative level of voltage during the first half of the period of the pulse, and a positive level in the second half. The CMI code has the following characteristics (see Figure 6):

- The spectrum of a CMI signal cancels out the components at very low frequencies.
- It allows for the clock to be recovered properly, like the HDB3 code.
- The bandwidth is greater than that of the spectrum of the same signal coded in AMI.

1.4 Multiplexing and Multiple Access

Multiplexing is defined as the process by which several signals from different channels share a channel with greater capacity (see Figure 8). Basically, a number of channels share a common transmission medium with the aim of reducing costs and complexity in the network. When the sharing is carried out with respect to a remote resource, such as a satellite, this is referred to as multiple access rather than multiplexing.

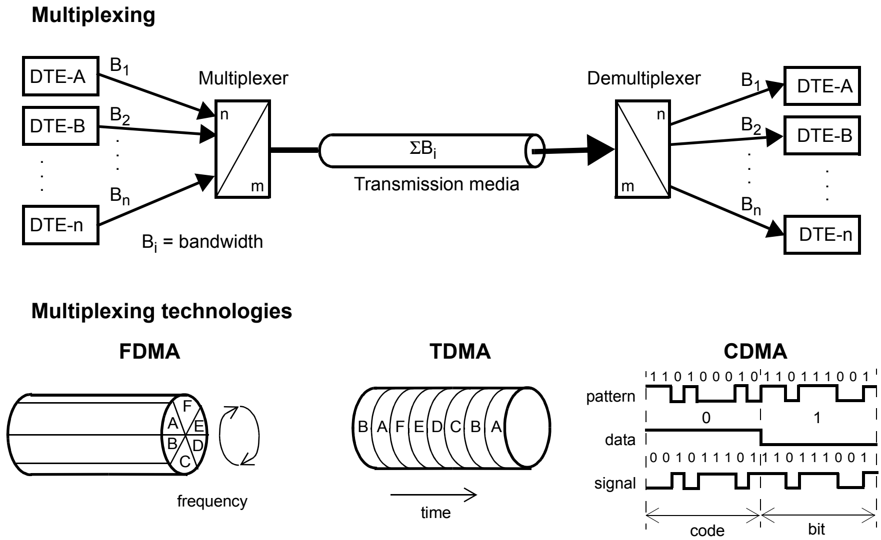

Figure 8 Multiplexing consolidates lower capacity channels into a higher capacity channel. Frequency division multiplexing access (FMDA) is used by radio, TV, and global system mobile (GSM). Time division multiplexing access (TDMA) is used by the integrated services digital network (ISDN), frame relay (FRL), and GSM. Code division multiplexing access (CDMA) is used by the third generation networks (3G) of mobiles.

Some of the most common multiplexing technologies are:

1. *Frequency division multiplexing/frequency division multiple access* (FDM/FDMA): Assigns a portion of the total bandwidth to each of the channels.
2. *Time-division multiplexing/time division multiple access* (TDM/TDMA): Assigns all the transport capacity sequentially to each of the channels.
3. *Code-division multiplexing access* (CDMA): In certain circumstances, it is possible to transmit multiple signals in the same frequency, with the receiver being responsible for separating them. This technique has been used for years in military technology, and is based on artificially increasing the bandwidth of the signal according to a predefined pattern.
4. *Polarization division multiple access* (PDMA): Given that polarization can be maintained, the polarization direction can be used as a multiple access technique, although when there are many obstacles, noise can make it unsuitable,

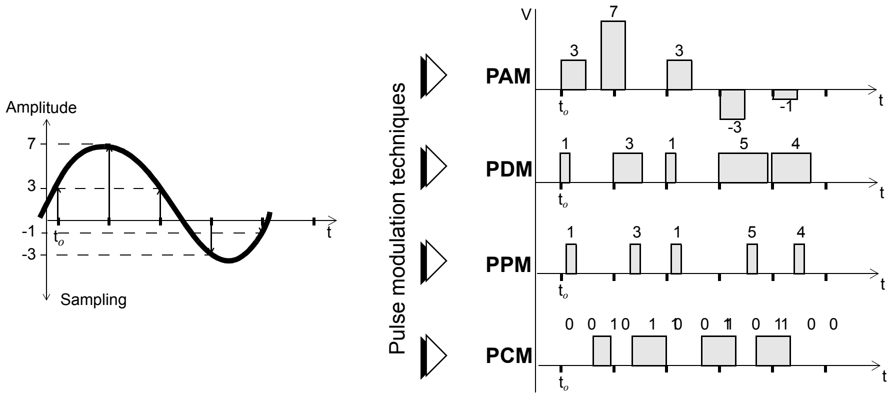

Figure 9 Pulse code modulation (PCM) was the technology selected to digitalize the voice in telephone networks. Other pulse techniques are pulse amplitude modulation (PAM), pulse duration modulation (PDM), and pulse position modulation (PPM).

which is why it is not generally used in indoor installations. Outside, however, it is widely exploited to increase transmission rates in installations that use microwaves.

5. *Space division multiple access* (SDMA): With directional antennas, the same frequency can be reused, provided the antennas are correctly adjusted. There is a great deal of interference, but this system lets frequencies obtain a high degree of reusability.

2 Pulse Code Modulation

The *pulse code modulation* (PCM) technology (see Figure 9) was patented and developed in France in 1938, but could not be used because suitable technology was not available until World War II. This came about with the arrival of digital systems in the 1960s, when improving the performance of communications networks became a real possibility. However, this technology was not completely adopted until the mid-1970s, due to the large amount of analog systems already in place and the high cost of digital systems, as semiconductors were very expensive. PCM's initial goal was that of converting an analog voice telephone channel into a digital one based on the sampling theorem (see Figure 10):

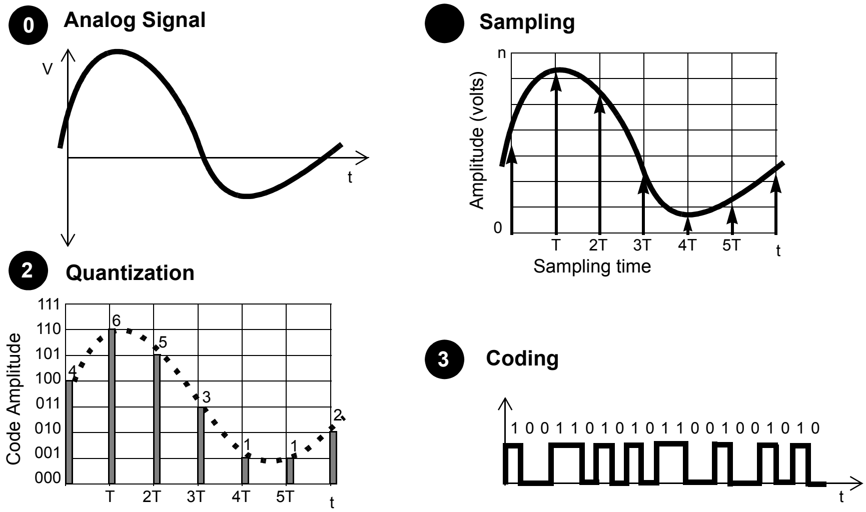

Figure 10 The three steps of digitalization of a signal: sampling of the signal, quantization of the amplitude, and binary encoding.

The sampling theorem states that for digitalization without information loss, the sampling frequency *(f_s)* should be *at least twice* the maximum frequency component *(f_{max})* of the analog information:

$$f_s > 2 \cdot f_{max}$$

The frequency $2 \cdot f_{max}$ is called the Nyquist sampling rate. The sampling theorem is considered to have been articulated by Nyquist in 1928, and mathematically proven by Shannon in 1949. Some books use the term *Nyquist sampling theorem*, and others use *Shannon sampling theorem*. They are in fact the same theorem.

PCM involves three phases: sampling, encoding, and quantization:

1. In sampling, values are taken from the analog signal every $1/f_s$ seconds (the sampling period).
2. Quantization assigns these samples a value by approximation, and in accordance with a quantization curve (i.e., A-law of ITU-T [2]).
3. Encoding provides the binary value of each quantified sample.

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2. This is a International Telecommunication Union (ITU-T) ratified audio encoding and compression technique (Rec. G.711). Among other implementations, A-law was originally intended as a phone-communications standard.

A telephone channel admits frequencies of between 300 Hz and 3,400 Hz. Because margins must be established in the channel, the bandwidth is set at 4 kHz. Then the sampling frequency must be $f_S \geq 2 \cdot 4{,}000 = 8{,}000\ Hz$; equivalent to a sample period of $T = 1/8{,}000 = 125\mu s$.

In order to codify 256 levels, 8 bits are needed, where the PCM bit rate (v) is:

$$v = 8{,}000_{\ samples/s} \times 8_{bits/sample} = 64Kbps$$

This bit rate is the subprimary level of transmission networks.

3 PDH and T-Carrier

At the beginning of the 1960s, the proliferation of analog telephone lines, based on copper wires, together with the lack of space for new installations, led the transmission experts to look at the real application of PCM digitalization techniques and TDM multiplexing. The first digital communications system was set up by Bell Labs in 1962, and consisted of 24 digital channels running at what is known as T1.

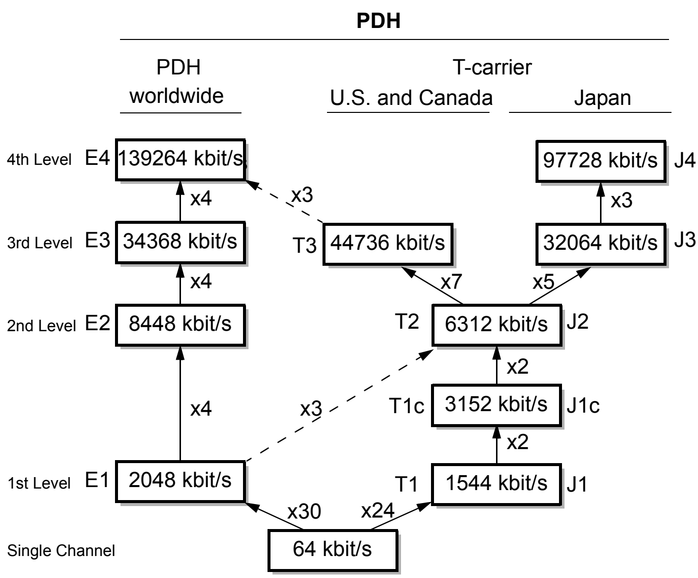

Figure 11 The PDH and T-carrier hierarchies, starting at the common 64-kbit/s channel and the multiplexing levels. Most of the narrowband networks are built on these standards: POTS, FRL, GSM, ISDN, ATM (asynchronous transfer mode), and leased lines to transmit voice, data, and video.

3.1 Basic Rates: T1 and E1

In 1965, a standard appeared in the U.S. that permitted the TDM multiplexing of 24 digital telephone channels of 64 kbit/s into a 1.544-Mbit/s signal with a format called T1 (see Figure 11). For the T1 signal, a synchronization bit is added to the 24 TDM time slots, in such a way that the aggregate transmission rate is:

$$(24_{channels} \times 8_{bit/channel} + 1_{bit}) / 125\mu s = 1{,}544 Mbps$$

125 µs is the sampling period

Europe developed its own TDM multiplexing scheme a little later (1968), although it had a different capacity: 32 digital channels of 64 kbit/s (see Figure 11). The resulting signal was transmitted at 2.048 Mbit/s, and its format was called E1 which was standardized by the ITU-T and adopted worldwide except in the U.S., Canada, and Japan. For an E1 signal, the aggregate transmission rate can be obtained from the following equation:

$$(32_{channels} \times 8_{bit/channel}) / 125\mu s = 2{,}048 Mbps$$

4 THE E1 FRAME

The E1 frame defines a cyclical set of 32 time slots of 8 bits. The time slot 0 is devoted to transmission management and time slot 16 for signaling; the rest were assigned originally for voice/data transport (see Figure 12).

The main characteristics of the 2-Mbit/s frame are described in the following.

4.1 Frame Alignment

In an E1 channel, communication consists of sending consecutive frames from the transmitter to the receiver. The receiver must receive an indication showing when the first interval of each frame begins, so that, since it knows to which channel the information in each time slot corresponds, it can demultiplex correctly. This way, the bytes received in each slot are assigned to the correct channel. A synchronization process is then established, and it is known as frame alignment.

4.2 Frame Alignment Signal

In order to implement the frame alignment system so that the receiver of the frame can tell where it begins, there is what is called a frame alignment signal (FAS) (see Figure 13). In the 2Mbit/s frames, the FAS is a combination of seven fixed bits ("0011011") transmitted in the first time slot in the frame (*time slot zero* or TS0). For the alignment mechanism to be maintained, the FAS does not need to be trans-

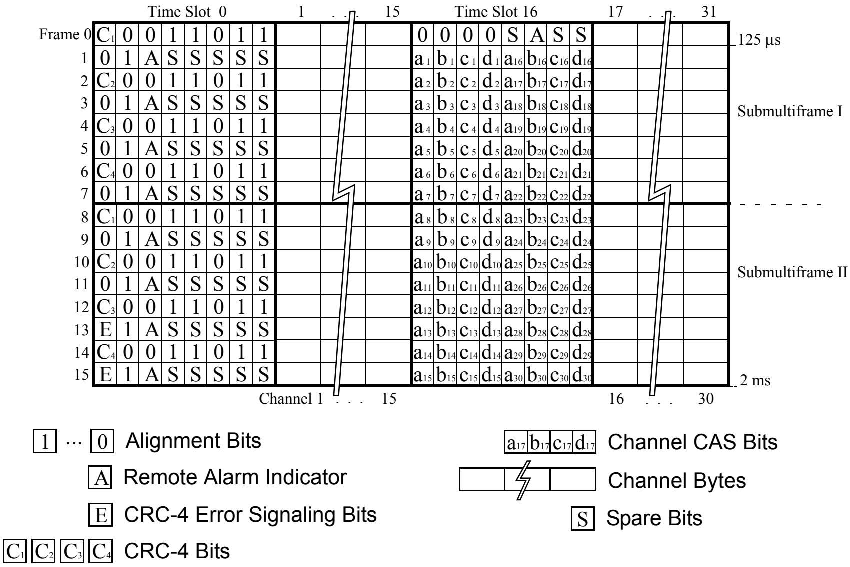

Figure 12 The E1 frame is the first hierarchy level, and all the channels are fully synchronous.

mitted in every frame. Instead, this signal can be sent in alternate frames (in the first, in the third, in the fifth, and so on). In this case, TS0 is used as the synchronization slot. The TS0 of the rest of the frames is therefore available for other functions, such as the transmission of the alarms.

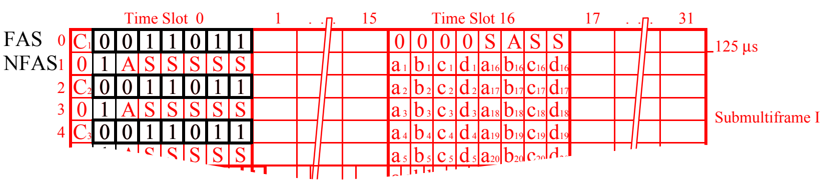

Figure 13 The E1 multiframe uses the FAS code only transmitted in even frames. The NFAS frames are the odd ones, using a bit equal to “1” to avoid coincidences.

4.3 Multiframe CRC-4

In the TS0 of frames with FAS, the first bit is dedicated to carrying the *cyclic redundancy checksum* (CRC). It tells us whether there are one or more bit errors in a specific group of data received in the previous block of eight frames known as submultiframe (see Figure 14).

4.3.1 The CRC-4 procedure

The aim of this system is to avoid loss of synchronization due to the coincidental appearance of the sequence "0011011" in a time slot other than the TS0 of a frame with FAS. To implement the CRC code in the transmission of 2-Mbit/s frames, a CRC-4 multiframe is built, made up of 16 frames. These are then grouped in two blocks of eight frames called submultiframes, over which a CRC checksum or word of four bits (CRC-4) is put in the positions C_i (bits #1, frames with FAS) of the next submultiframe.

At the receiving end, the CRC of each submultiframe is calculated locally and compared to the CRC value received in the next submultiframe. If these do not coincide, one or more bit errors is determined to have been found in the block, and an alarm is sent back to the transmitter, indicating that the block received at the far end contains errors (see Table 2).

4.3.2 CRC-4 multiframe alignment

The receiving end has to know which is the first bit of the CRC-4 word (C1). For this reason, a CRC-4 multiframe alignment word is needed. Obviously, the receiver has to be told where the multiframe begins (synchronization).

The CRC-4 multiframe alignment word is the set combination "001011," which is introduced in the first bits of the frames that do not contain the FAS signal.

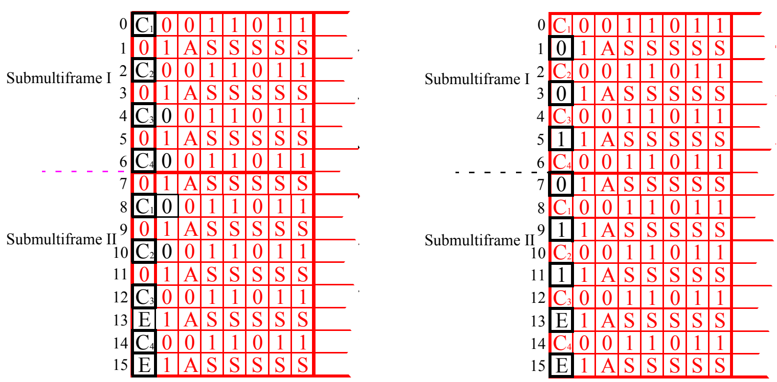

Figure 14 The CRC-4 provides error monitoring by means of four Ci bits that correspond to the previous submultiframe. If the receiver detects errors, it sets the E-bit to indicate the error. The "001011"sequence is used to synchronize the submultiframe.

4.3.3 Advantages of the CRC-4 method

The CRC-4 method is mainly used to protect the communication against a wrong frame alignment word, and also to provide a certain degree of monitoring of the *bit error rate* (BER), when this has low values (around 10^{-6}). This method is not suitable for cases in which the BER is around 10^{-3} (where each block contains at least one errored bit).

Another advantage in using the CRC is that all the bits transmitted are checked, unlike those systems that only check seven bits (those of the FAS, which are the only ones known in advance) out of every 512 bits (those between one FAS and the next). However, the CRC-4 code is not completely infallible, since there is a probability of around 1/16 that an error may occur and not be detected; that is, that 6.25% of the blocks may contain errors that are not detected by the code.

4.3.4 Monitoring errors

The aim of monitoring errors is to continuously check transmission quality without disturbing the information traffic and, when this quality is not of the required standard, taking the necessary steps to improve it. Telephone traffic is two way, which means that information is transmitted in both directions between the ends of the communication. This in its turn means that two 2-Mbit/s channels and two directions for transmission must be considered.

The CRC-4 multiframe alignment word only takes up six of the first eight bits of the TS0 without FAS. There are two bits in every second block or submultiframe, whose task is to indicate block errors in the far end of the communication. The mechanism is as follows: Both bits (called E-bits) have “1” as their default value. When

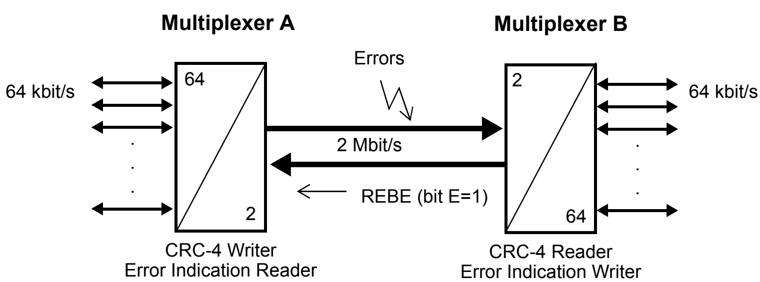

Figure 15 The A multiplexer calculates and writes the CRC code, and the multiplexer B reads and checks the code. When errors affect the 2-Mbit/s frame, the multiplexer B indicates the problem by means of the E-bit of the frame which travels toward the multiplexer B.

the far end of the communication receives a 2Mbit/s frame and detects an erroneous block, it puts a “0” in the E-bit that corresponds to the block in the frame being sent along the return path to the transmitter (see Figure 15). This way, the near end of the communication is informed that an erroneous block has been detected, and both ends have the same information: one from the CRC-4 procedure and the other from the E bits. If we number the frames in the multiframe from 0 to 15, the E-bit of frame 13 refers to the submultiframe I (block I) received at the far end, and the E-bit of frame 15 refers to the submultiframe II (block II).

4.4 Supervision Bits

The bits that are in position 2 of the TS0 in the frame that does not contain the FAS are called supervision bits and are set to “1,” to avoid simulations of the FAS signal.

4.5 NFASs - Spare Bits

The bits of the TS0 that do not contain the FAS in positions 3 to 8 make up what is known as the *non-frame alignment signal* or NFAS. This signal is sent in alternate frames (frame 1, frame 3, frame 5, etc.). The first bit of the NFAS (bit 3 of the TS0) is used to indicate that an alarm has occurred at the far end of the communication. When operating normally, it is set to “0,” while a value of “1” indicates an alarm.

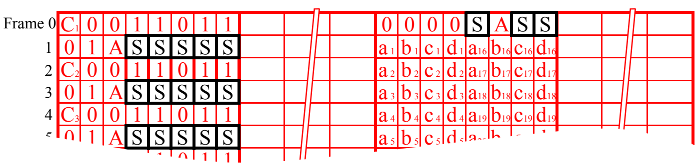

Figure 16 Spare bits in the E1 frame.

The bits in positions 4 to 8 are spare bits (see Figure 16), and they do not have one single application, but can be used in a number of ways, as decided by the telecommunications carrier. In accordance with the ITU-T Rec. G.704, these bits can be used in specific point-to-point applications, or to establish a data link based on messages for operations management, maintenance or monitoring of the transmission quality, and so on. If these spare bits in the NFAS are not used, they must be set to “1” in international links.

4.6 NFAS - Alarm Bit

The method used to transmit the alarm makes use of the fact that in telephone systems, transmission is always two way (see Figure 17). Multiplexing/demultiplexing devices (known generically as multiplex devices) are installed at both ends of the communication for the transmission and reception of frames. An alarm must be sent to the transmitter when a device detects either a power failure or a failure of the coder/decoder, in its multiplexer; or any of the following in its demultiplexer: *loss of the signal* (LOS), *loss of frame alignment* (LOF), or a BER greater than 10^{-3}.

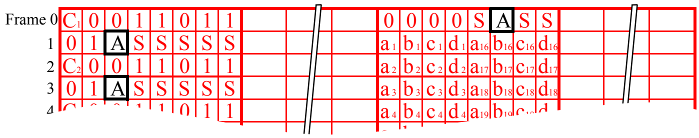

Figure 17 The alarm indication signal is used to send alarms to the remote end to indicate a power fault, loss of incoming signal, loss of frame, coder/decoder fault or a high bit error rate, among others.

The *remote alarm indication* (RAI) is sent in the NFAS of the return frames, with bit 3 being set to "1." The transmitter then considers how serious the alarm is, and goes on generating a series of operations, depending on the type of alarm condition detected (see Table 2).

4.7 Signaling Channel

As well as transmitting information generated by the users of a telephone network, it is also necessary to transmit signaling information. Signaling refers to the protocols that must be established between exchanges so that the users can exchange information between them.

There are signals that indicate when a subscriber has picked up the telephone, when he or she can start to dial a number, and when another subscriber calls, as well as signals that let the communication link be maintained, and so on.

In the E1 PCM system, signaling information can be transmitted by two different methods: the *common channel signaling* (CCS) method and the *channel associated signaling* (CAS) method. In both cases, the time slot TS16 of the basic 2-Mbit/s frame is used to transmit the signaling information (see Figure 18).

For CCS signaling, messages of several bytes are transmitted through the 64-kbit/s channel provided by the TS16 of the frame, with these messages providing the signaling for all the channels in the frame. Each message contains information that determines the channel that is signaling. The signaling circuits access the 64-kbit/s channel of the TS16, and they are also common to all the channels signaled. There are different CCS systems that constitute complex protocols. In the following section and by way of example, channel associated signaling will be looked at in detail. CAS is defined in the ITU-T Rec. G.704, which defines the structure of the E1 frame.

In CAS signaling, a signaling channel is associated with each information channel (there is no common signaling channel), meaning that the signaling circuits are personalized for each channel.

4.8 CAS Signaling Multiframe

In the case of channel associated signaling, each 64kbit/s telephone channel is assigned 2 kbit/s for signaling. This signaling is formed by a word of 4 bits (generically known as a, b, c, and d) that is situated in the TS16 of all the frames sent. Each TS16 therefore carries the signaling for two telephone channels.

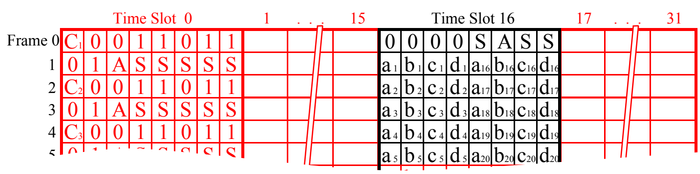

Figure 18 When the CAS method is used, each of the channels has an associated 2-kbit/s channel (a_i b_i c_i d_i) in the time slot 16. This multiframe also has an alignment signal “0000”; spare and alarm bit to be used specifically by the signaling multiframe.

Given that there are only four signaling bits available for each channel, to transmit all the signaling words from the 30 PCM channels that make up a 2-Mbit/s frame (120 bits), it is necessary to wait until the TS16 of 15 consecutive frames have been received. The grouping of frames defines a CAS signaling multiframe, which consists of a set of the TS16 of 16 consecutive E1 frames.

4.8.1 CAS multiframe alignment signal

In order to synchronize the CAS multiframe, that is to identify where it begins, a *multiframe alignment signal* (MFAS) is defined, made up of the sequence of bits “0000” located in the first four bits of the TS16 of the first frame of the CAS multiframe.

4.8.2 CAS non-multiframe alignment signal

The remaining four bits of the interval are divided between one alarm bit and three spare bits, making up the *non-multiframe alignment signal* (NMFAS). In short, the signaling information for the 30 channels is transmitted in 2 ms, which is fast enough if we consider that the shortest signaling pulse (the one that corresponds to dialing the number) lasts for 100 ms.

The alarm bit in the NMFAS is dealt with in a similar way to the NFAS. In this case, the alarms are transmitted between multiplex signaling devices connected to the 64-kbit/s circuits that correspond to signaling (TS16). The alarm is sent when the CAS multiplexer detects:

- A power failure;
- Loss of incoming signaling;
- Loss of CAS multiframe alignment.

An indication must be sent to the multiplex signaling device at the far end (see Table 2), setting bit 6 of the TS16 in the return frame 0 to "1." Additionally, the value "1" is applied to all the signaling channels (see Figure 22).

Example: A remote multiplexer is considered to have lost multiframe alignment when it receives two consecutive MFAS words with error, that is, with a value other than "0000." In this case, bit 6 of the TS16 of the frame 0 that this multiplexer transmits to the near-end multiplexer is set to "1." When it receives this indication of loss of multiframe alignment at the far end, the near end multiplexer sends a signal made up entirely of bits at "1," known as AIS64 (*alarm indication signal* - 64 kbit/s) in the TS16 (64-kbit/s channel).

Table 2
2-Mbit/s events: Alarms, errors, and event indications.

ID	*Explanation*
AIS	Alarm indication signal. It is detected if there are two or less zeros (ITU-T G.775).
LOF	Loss of frame alarm. It is raised after three consecutive frames with FAS error or three consecutive signalling bits (ITU-T G.706).
LOS	Loss of frame signal alarm.
RAI	Remote alarm indication. It is detected after three consecutive frames with the A bit equals to 1 (ITU-T G.732).
FAS error	Frame alignment signal error, indicating an incorrect bit in the alignment word.
Bit error	Bit sequence mismatch (when the transmitted pattern is known).
Code error	Violation on coding sequence.
CRC-LOM	Cyclic redundancy checksum - loss of multiframe. It is activated if there is LOF and deactivated after one correct FAS and two correct CRC-MFAS (ITU-T G.706).
CAS-LOM	Channel associated signaling-loss of multiframe. It is raised after two consecutive MFAS errors or two multiframes with time-slot 16 bits equal to 0 (ITU-T G.732).

Table 2
2-Mbit/s events: Alarms, errors, and event indications.

CAS-MRAI	Channel associated signaling-multiframe remote alarm indication. Detected after two consecutive frames with the MRAI bit equal to 1 (ITU-T G.732).
CAS-MAIS	Channel associated signaling-multiframe alarm indication signal. It is detected if there are less than three zeros in the time slot 16 during two consecutive multi-frames.
CRC error	Cyclic redundancy check error. It is raised whenever one or more bits are erroneous, whenever CRC-LOM is off (ITU-T G.706).
REBE	Remote end block error. It is erased if the first bit of the frames 14 and 16 is 0 (ITU-T G.706).

5 The Plesiochronous Digital Hierarchy

Based on the E1 signal, the ITU-T defined a hierarchy of plesiochronous signals that enables signals to be transported at rates of up to 140 Mbit/s (see Figure 19). This section describes the characteristics of this hierarchy and the mechanism for dealing with fluctuations in respect to the nominal values of these rates, which are produced as a consequence of the tolerances of the system.

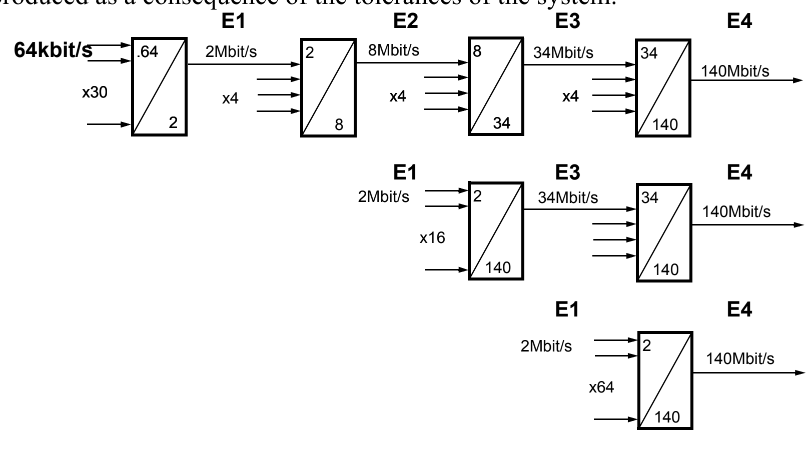

Figure 19 The PDH hierarchy, with four levels from 2 to 140 Mbit/s. A bit-oriented justification process is used to fit tributaries created with different clocks in the second, third, and fourth hierarchy. The first hierarchy, 2 Mbit/s, is the only fully synchronous frame.

5.1 Higher Hierarchical Levels

As is the case with level 1 of the plesiochronous digital hierarchy (2 Mbit/s), the higher levels of multiplexing are carried out bit by bit (unlike the multiplexing of 64-kbit/s channels in a 2-Mbit/s signal, which is byte by byte), thus making it impossible to identify the lower level frames inside a higher level frame. Recovering the tributary frames requires the signal to be fully demultiplexed.

The higher hierarchical levels (8,448, 34,368, and 139,264 Mbit/s, etc.; referred to as 8, 34, and 140 Mbit/s for simplicity) are obtained by multiplexing four lower level frames within a frame whose nominal transmission rate is more than four times that of the lower level (see Table 3), in order to leave room for the permitted variations in rate (justification bits), as well as the corresponding FAS, alarm, and spare bits (see Figure 19).

5.2 Multiplexing Level 2: 8 Mbit/s

The 8-Mbit/s frame structure is defined in the ITU-T Rec. G.742 (see Figure 20). The frame is divided into four groups:

- Group I contains the FAS, with sequence “1111010000”; the A-bit (remote alarm); the S-bit (spare); and 200 T-bits (tributary) transporting data.
- Groups II and III contain a block of four J-bits (justification control) and 208 T-bits transporting data.
- Group IV contains a block of four J-bits, a block of R-bits (justification opportunity), one per tributary, and 204 T-bits. To check whether R-bits have been used, the J-bits are analyzed in each of the groups II, III, and IV (there are three per tributary). Ideally the R-bit does not carry useful information on 42.4% of the occasions. In other words, this percentage is the probability of justification or the insertion of stuffing bits.

5.3 Multiplexing Level 3: 34 Mbit/s

The structure of this frame is described in the ITU-T Rec. G.751 (see Figure 20). As in the previous case, the frame is divided into four groups:

- Group I contains the FAS, with sequence “1111010000”; the A-bit (remote alarm); the S-bit (spare); and 372 T-bits (tributary) transporting data.
- Groups II and III contain a block of four J-bits (justification control) and 380 T-bits transporting data.
- Group IV contains a block of four J-bits, a block of R-bits (justification opportunity) one per tributary, and 376 T-bits. To check whether R-bits have been used, the J-bits are analyzed in each of the groups II, III, and IV (there are three per tributary). Ideally the R-bit does not carry useful information on 43.6% of the occasions.

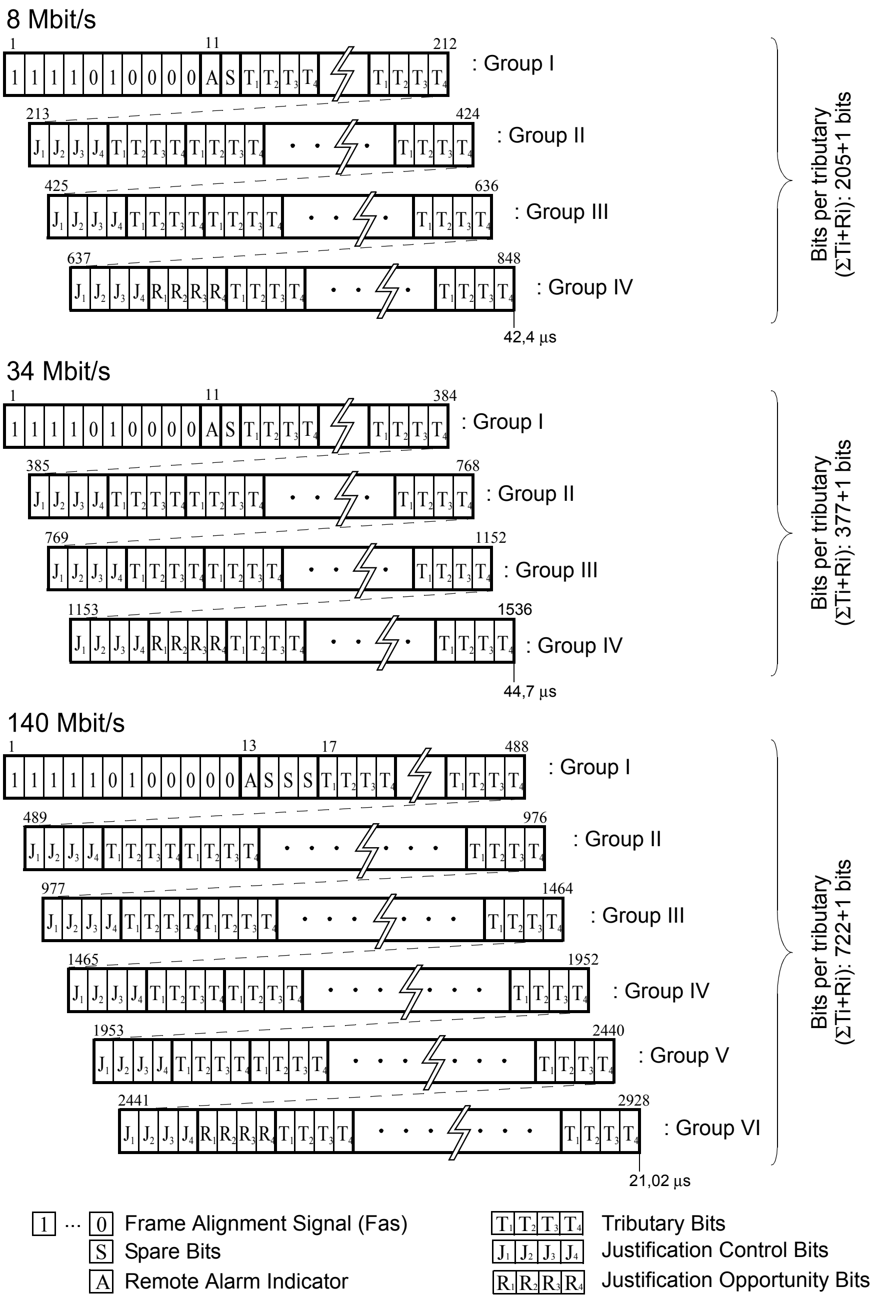

Figure 20 The PDH higher hierarchies. A bit-oriented justification process is used to fit tributaries created with clock impairments.

5.4 Multiplexing Level 4: 140 Mbit/s

The structure of this frame is described in the ITU-T Rec. G.751 (see Figure 20). In this case, the frame is divided into six groups:

- Group I contains the FAS, with sequence “111110100000;” the A-bit (remote alarm); the S-bit (spare); and 472 T-bits (tributary) transporting data.
- Groups II, III, IV, and V contain a block of four J-bits (justification control) and 484 T-bits transporting data.
- Group VI contains a block of four J-bits, a block of R-bits (justification opportunity), one per tributary, and 376 T-bits. To check whether R-bits have been used, the J-bits are analyzed in each of the groups II, III, IV, V, and VI (there are five per tributary). Ideally the R-bit does not carry useful information on 41.9% of the occasions.

Table 3
The PDH hierarchy, with four levels from 2 to 140 Mbit/s. A bit-oriented justification process is used to fit tributaries created with different clocks in the second, third, and fourth hierarchy.

Standard	*Binary Rate*	*Size*	*Frame/s*	*Code*	*Amplitude*	*Attenuation*
G.704/732	2,048 kbit/s±50 ppm	256 bits	8,000	HDB3	2.37-3.00V	6 dB
G.742	8,448 kbit/s±30 ppm	848 bits	9,962.2	HDB3	2.37V	6 dB
G.751	34,368 kbit/s±20 ppm	1536 bits	22,375.0	HDB3	1.00V	12 dB
G.751	139,264 kbit/ s±15 ppm	2928 bits	47,562.8	CMI	1.00V	12 dB

5.5 Service Bits in Higher Level Frames

In any of the groups containing the FAS in the 8-, 34-, and 140-Mbit/s frames, alarm bits and spare bits are also to be found. These are known as service bits. The A-bits (alarm) carry an alarm indication to the remote multiplexing device, when certain breakdown conditions are detected in the near-end device. The spare bits are designed for national use, and must be set to “1” in digital paths that cross international boundaries.

5.6 Plesiochronous Synchronization

As far as synchronization is concerned, the multiplexing of plesiochronous signals is not completely trouble free, especially when it comes to demultiplexing the circuits. In a PCM multiplexer of 30 + 2 channels, a sample of the output signal clock (1/32) is sent to the coders, so that the input channels are synchronized with the output frame. However, higher level multiplexers receive frames from lower level multiplexers with clocks whose value fluctuates around a nominal frequency value within certain margins of tolerance.

The margins are set by the ITU-T recommendations for each hierarchical level. The signals thus formed are almost synchronous, except for differences within the permitted margins of tolerance, and for this reason they are called plesiochronous (see Figure 21).

5.7 Positive Justification

In order to perform bit-by-bit TDM, each higher-order PDH multiplexer has elastic memories in each of its inputs in which the incoming bits from each lower level signal line or tributary are written. Since the tributary signals have different rates, they are asynchronous with respect to each other. To prevent the capacity of the elastic memories from overflowing, the multiplexer reads the incoming bits at the maximum rate permitted within the range of tolerances.

When the rate of the incoming flow in any of the tributary lines is below this reading rate, the multiplexer cannot read any bits from the elastic memory, and so it uses a stuffing bit or justification bit (called justification opportunity) in the output aggregate signal. Its task is that of adapting the signal that enters the multiplexer to the rate at which this signal is transmitted within the output frame (its highest clock value). This type of justification is called positive justification.

Justification bits, together with other overhead bits, make the output rate higher than the total of the input signals.

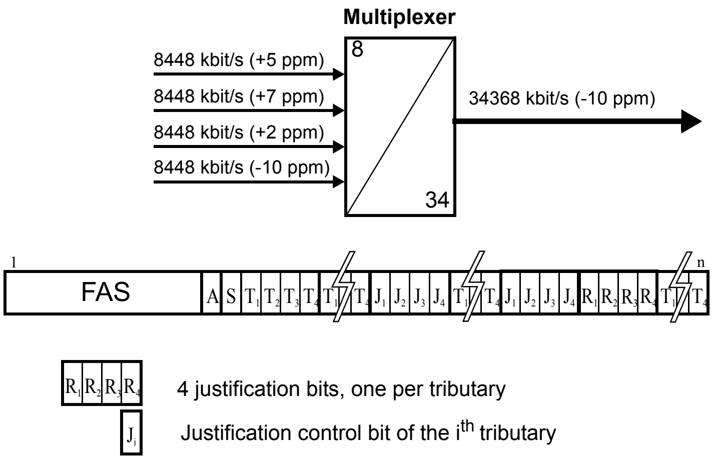

Figure 21 The PDH and the T-carrier hierarchies are not synchronous and variations can be expected in the bit rate clock, shown in this figure as parts per million (ppm). The justification mechanism is implemented in the E2, E3, and E4 frames. If all J_i=1, then R_i is a justification bit that does not contain information. If all J_i=0, then R_i contains information. If all are not 0 or 1, the decision is based on the majority.

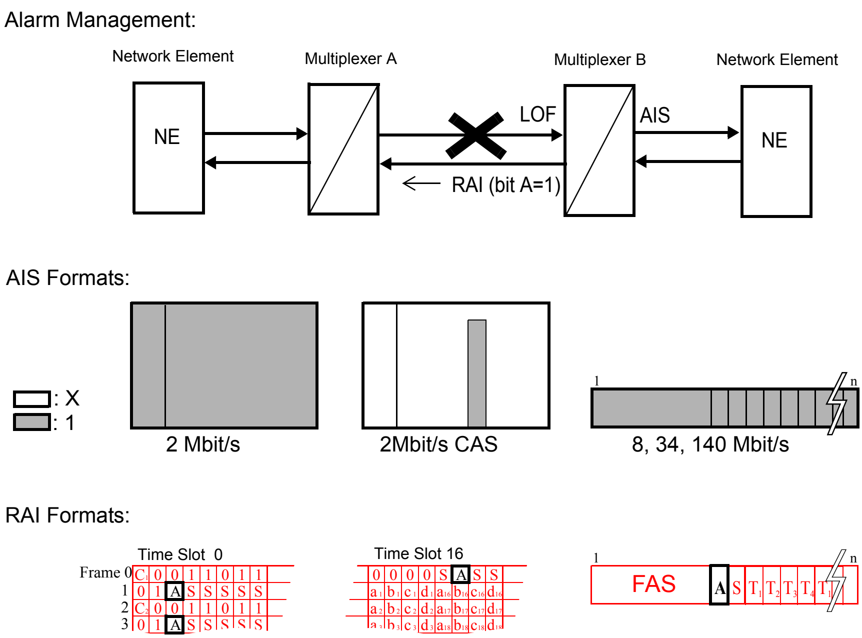

Figure 22 When a multiplexer detects an LOS or LOF, it sends a remote alarm indication (RAI) to its partner multiplexer and forwards an AIS to the next network element, because it has not been possible to recover any information.

5.7.1 Justification opportunity bits

The task of the *justification opportunity bits* (R-bits) is to be available as extra bits that can be used when the rate of the incoming tributaries is higher than its nominal value (within the margin specified by ITU-T) by an amount that makes this necessary. In this case, the opportunity bit is no longer mere stuffing, but becomes an information bit instead.

In order for the device that receives the multiplexed signal to be able to determine whether a justification opportunity bit contains useful information (i.e., information from a tributary), *justification control bits* (J-bits) are included in the frame. Each group of control bits refers to one of the tributaries of the frame. All of them will be set to "0" if the associated opportunity bit is carrying useful information; otherwise they will be set to "1."

Several bits are used instead of just one, to provide protection against possible errors in transmission. On examining the control bits received, if they do not all have the same value, it is decided that they were sent with the majority value (a "1" if there are more 1s than 0s, for instance; it is assumed that there has been an error in the bits that are at 0).

It can be seen that there is a dispersion of the control bits referring to a tributary that causes them to be located in separate groups. Spreading out the J-bits (control bits), reduces the probability of errors occurring in them, and a wrong decision being made as to whether or not they have been used as a useful data bit. If the wrong decision is made, there is not only an error in the output data, but also a slip of one bit; that is, the loss or repetition of one bit of information.

6 MANAGING ALARMS IN HIGHER LEVEL HIERARCHIES

The A-bit of the FAS in 8-, 34-, and 140-Mbit/s frames enables the multiplexers that correspond to these hierarchies to transmit alarm indications to the far ends (see Figure 22) when a multiplexer detects an alarm condition (see Table 4).

In addition, 140-Mbit/s multiplexers also transmit an alarm indication when faced with the loss of frame alignment of the 34-Mbit/s signals received inside the 140-Mbit/s signals, as well as in the NFAS of the 34-Mbit/s signal that has lost its alignment (bit 11 of group I changes from "0" to "1") in the return channel (see Figure 21).

Table 4
PDH events: Alarms, errors, and event indications.

ID	*Explanation*
AIS	Alarm indication signal. This is detected if less than six zeros in a frame in the case of 140 Mbit/s, or less than three zeros in 34 Mbit/s, and 8 Mbit/s (ITU-T G.751 and ITU-T G.742).
LOF	Loss of frame alarm. It is raised after four consecutive frames with FAS error (ITU-T G.751 and ITU-T G.742).
LOS	Loss of frame signal alarm.
RAI	Remote alarm indication. It is detected after two consecutive frames with the A bit equal to 1 (ITU-T G.751 and ITU-T G.742).
FAS error	Frame alignment signal error. One or more incorrect bits in the alignment word.

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