HERMON LABORATORIES

Digital T1 and E1 Interfaces

Compliance Requirements Overview

T1 is a digital transmission link with a total transmit and receive rate of 1.544 Mbps (1544000 bits per second).
E1 is a digital transmission link with a total transmit and receive rate of 2.048 Mbps (2048000 bits per second).
T1 is used in North America and Asia and E1 is used in Europe and Australia.
T1 and E1 links enable simultaneous transmission and receiving of several data or voice channels or of unchannelized raw bit stream.
A T1 line uses two balanced pairs of copper wire. One pair is used for transmitting (Tip-Ring) and another pair for receiving (Tip1-Ring1). The nominal impedance of each transmit and receive pair is 100 ohm.
An E1 interface uses the same type of line as T1 but has the nominal impedance of 120 ohm. There are also 75 ohm coax (unbalanced) E1 lines.
In addition to copper lines both T1 and E1 are provided in fiber-optics.
E1 and T1 lines are used to connect between PABXs, COs, routers and switches. They are offered by telephone companies/service providers as point-to-point leased lines and switched lines for applications such as private WAN (Wide Area Network) and public switched data and voice communication, internet access and video conference. The E1/T1 lines can connect equipment within the public network, within the customer private network or connect between public and customer equipment.
T1 and E1 interfaces belong to the physical layer (layer 1) in the OSI reference model, thus higher layer technologies such as ISDN, ATM, Frame Relay, TCP/IP and VoIP can be carried over T1 and E1.

The OSI (Open Systems Interconnection) 7 layer model structure was introduced by ISO to describe convenient break points between various parts of the hardware and software in any data communications system. The communication functions are broken down into a hierarchical set of layers. Each layer performs a related subset of the functions required to communicate with another system. It relies on the next lower layer to perform more primitive functions and to conceal the details of those functions. It provides services to the next higher layer. The layers are defined in such a manner so that changes in one layer do not require changes in the other layers. By partitioning the communication functions into layers, the communication task is much more manageable.

Table 1. OSI reference model.

Layer 1	Physical layer	Deals with physical means of sending data over lines. Includes mechanical, electrical and functional specifications for wiring, power transfer, line signals, timing, frame structure, encoding, multiplexing, error reporting, maintenance and performance monitoring.
Layer 2	Data link layer	Provides data formats, procedures and protocols for operating the communication lines. Provides means for detecting and correcting message errors.
Layer 3	Network layer	Deals with routing and relaying of the communication within and between the individual networks. Provides upper layers with independence from the data transmission and switching technologies used to connect systems. It is responsible for establishing, maintaining and terminating connections.
Layer 4	Transport layer	Defines the rules for information exchange and manages end-to-end delivery of information within and between the networks.
Layer 5	Session layer	Is concerned with dialog management, controlling the use of the communication facilities provided by the transport layer.
Layer 6	Presentation layer	Provides transparent communication services by hiding the differences of various data formats.
Layer 7	Applications layer	Contains functions for specific applications services e.g. access of particular data file formats.

T1 and E1 are part of digital hierarchy, known also as DS (Digital Signal level) hierarchy. Referenced as DS1, T1 and E1 can include several 64 kbps (kilobits per second) DS0 channels, and can be incorporated (multiplexed) into higher rate streams. For example four E1 2048 Mbps streams can be multiplexed into a single 8448 kbps (E2) stream by a second order multiplex. This technique may be repeated to give a third order 34368 kbps (E3) signal, a fourth order (139264 kbps) signal, and so on. The higher order multiplexes need to be able to accommodate small discrepancies in frequency between their input streams (tributaries). They do this by running slightly faster than necessary and adding extra bits (bit stuffing) to account for variations in the input tributaries. This pyramid like structure is called Plesiochronous Digital Hierarchy (PDH). The term Plesiochronous means "nearly synchronous". High rate SONET (Synchronous Optical Network) and SDH (Synchronous Digital Hierarchy) networks provide signals and facilities to synchronize the individual network node and terminal equipment, as opposed to the asynchronous PDH structure. Also the SONET and SDH structure eliminate the need for multiple-stages multiplexing/demultiplexing and add overhead service data necessary for reliable communication at high rates. SONET and SDH are similar and closely related. SONET standards are published by ANSI (American National Standards Institute). SDH is the international version of standards which is published by ITU (International Telecommunications Union).

Table 2. PDH (Plesiochronous Digital Hierarchy).

Line Designation	Line Rate	Equivalent 64 kbps payload (data or voice) channels
North America		
DS0	64 kbps	1
T1 (DS1)	1.544 Mbps	24
T1C (DS1C)	3.152 Mbps	48
T2 (DS2)	6.312 Mbps	96
T3 (DS3)	44.736 Mbps	672
T4 (DS4) NA	139.264 Mbps	2016
T4 (DS4) Canada	274.176 Mbps	4032
Europe		
DS0	64 kbps	1
E1 (DS1)	2.048 Mbps	30
E2 (DS2)	8.448 Mbps	120
E3 (DS3)	34.368 Mbps	480
DS4	139.268 Mbps	1920

Table 3. SDH (Synchronous Digital Hierarchy)

Line Designation	Line Rate	Equivalent capacity
SDH		
STM-0	51.840 Mbps	28 DS1
STM-1	155.520 Mbps	84 DS1
STM-4	622.080 Mbps	356 DS1
STM-16	2488.320 Mbps	1344 DS1
STM-64	9952.280 Mbps	5576 DS1

STM (Synchronous Transport Module)

Table 4. Sonet (Synchronous Optical Network)

Line Designation		Line Rate	Equivalent capacity
Optical	**Electrical**		
OC-1	STS-1	51.840 Mbps	28 DS1
OC-3	STS-3	155.520 Mbps	84 DS1
OC-9	STS-9	466.560 Mbps	252 DS1
OC-12	STS-12	622.080 Mbps	356 DS1
OC-18	STS-18	933.120 Mbps	504 DS1
OC-24	STS-24	1.244 Gbps	572 DS1
OC-36	STS-36	1.866 Gbps	1008 DS1
OC-48	STS-48	2.488 Gbps	1344 DS1
OC-96		4.974 Gbps	2688 DS1
OC-192		9.953 Gbps	5376 DS1

STS (Synchronous Transfer Signal)

In order to read correctly the bit stream, digital equipment must be able to synchronize its receiver circuitry with each incoming bit. A clock signal is used to pace the receiving and transmitting of the data bits. The receive clock can be derived from pulse shape of incoming "1" bits. Such clock extraction is possible because the pulses have about 50% duty cycle, i.e. the peak amplitude is transmitted for 50% of the time. Correct transmitter pulse shape amplitude and time characteristics are important for the receiver ability to extract the clock properly.

Rather than extracting a clock from the data, a clock signal can be directly provided from the network at a separate line. Since this solution requires a separate line, it is more expensive and is seldom used in E1 and T1 systems. The transmit clock can be generated internally (internal clock mode), derived from the received data (also known as recovered clock mode, loop timing or network timing) or taken from external clock source (external clock mode).

Ideally data and clock pulses would come at correct time intervals and with exact nominal frequency. In practice frequency and timing deviations occur. Such deviations can cause clock synchronization problems and lead to errors especially when E1/T1 streams are multiplexed or mapped to higher rates. Thus these deviations must be limited at the output of the equipment. On the other hand the equipment should be able receive the data and operate correctly in presence of the signal frequency and time deviations that may exist on the network.

The maximum frequency deviation specified by the standards is 50 ppm (parts per million) for E1 and 32 ppm for T1.

The timing or phase deviation of a signal is called jitter. Very slow jitter with frequency below 10 Hz is called wander.

Jitter can be caused by impulsive noise, crosstalk, distortion, oscillator drift due to thermal noise, delay fluctuations and clock differences and modulation due to multiplexing and mapping.

Jitter amplitude is expressed in unit intervals (UI), where 1 UI corresponds to phase deviation of one bit. The standards specify maximum limits for output jitter for all the possible clock sources at a range of input jitter and frequency offset signals. Also the equipment should be able to receive the data without errors or alarms, while jitter is applied to its input.

Line signals

The data bits are transmitted to the line as pulses representing "1" and spaces (no pulse) representing "0".

The nominal "1" pulse (mark) voltage is 3V for T1 and 120 ohm E1 and 2.37 V for 75 ohm E1.

If pulses had only one polarity - that would introduce a DC component to the line. The signal having a DC component can't be transmitted, because the repeaters, placed along the line to retransmit the signal, have DC power feeding supplied on the same line. To overcome this problem the polarity of each pulse is inverted with regards to the preceding pulse. This polarity inversion is called AMI (Alternate Mark Inversion) signal (or line) coding. Such bipolar pattern also halves the fundamental frequency of the signal which results in less attenuation and group delay. The AMI signal coding does not address another problem, though. When there is long sequence of zeros the equipment at the other end of the line can't synchronize and thinks that the link is lost.

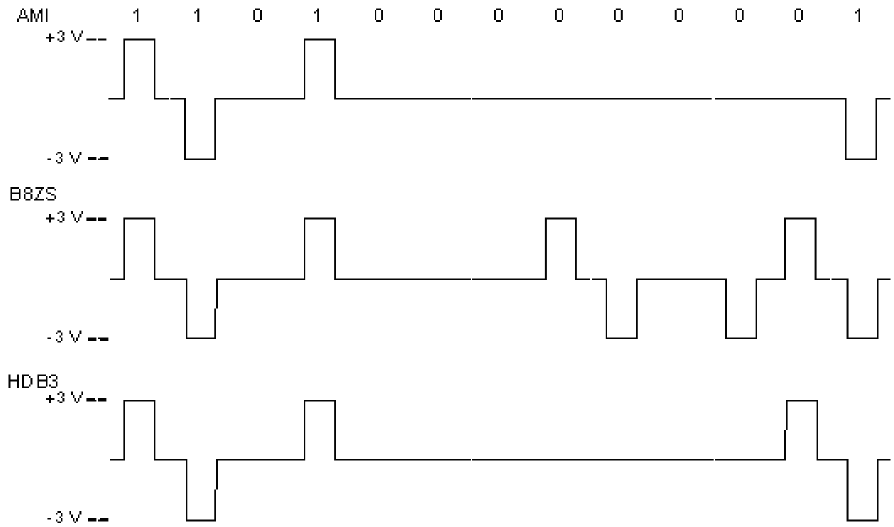
To solve this problem pulses are inserted in each sequence of 8 (in T1) or 4 (E1) continuous zeros. Such "artificial" pulse has the polarity opposite to what is required by the AMI rule.

This results in line code with maximum three consecutive zeros followed by the bipolar violation pulse. Such line code is called HDB3 (High Density Bipolar) and it is used in E1.

T1 systems use B8ZS (Bipolar with Eight Zero Substitution) line code substitutes each sequence of 8 consecutive zeros with 00VP0VP. Where V is a pulse of the same polarity (bipolar violation) as the previous pulse and P is a pulse with the opposite polarity.

ITU-T G.703 allows B8ZS or AMI line codes for use in T1 interfaces. For AMI it requires maximum 15 consecutive zeros and limits the maximum ratio between zeros and ones.

Figure 1. AMI, B8ZS, HDB3 Line Coding.



The standards limit the pulse shape of a one by a mask which consists of maximum and minimum voltage vs time limits. For E1 interface the amplitude ratio and widths ratio between positive and negative pulses is limited between 0.95 to 1.05, space peak voltage is limited by +/- 0.3 V for 120 and +/-0.237 for 75 ohm.
The pulses are attenuated and are spread out in time domain as they travel down the line.
FCC Part 68/TIA-968 specifies pulse shape limits that correspond to 0 dB line loss (option A), 7.5 dB loss (option B) and 15 dB (option C). These pulse shape characteristics, called line buildout (LBO), should match the telephone company requirements due to location of its repeater equipment, when the T1 terminal equipment is installed (connected to the telephone company network).
In addition to pulse shape requirements FCC Part 68/TIA-968 limits the maximum T1 output power at the 772 kHz (carrier) by 19 dBm and the maximum second harmonic power (1.544 MHz) is limited by at least 25 dB below the carrier. The measurements are performed with all ones output signal, 3 kHz bandwidth and at 100 ohm load termination. The G.703 limits the T1 output power between 12.6 dBm to 17.9 dBm and the second harmonic shall be 29 dB at least below the carrier.

Figure 2. E1 (2.048 MBps) Pulse Shape per G.703 / TBR 12, TBR 13, TBR 4.

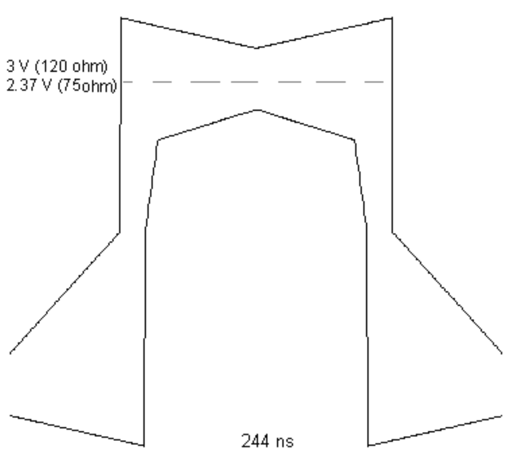


Figure 3. T1 (1.544 Mbps) Option A Pulse Shape per FCC Part 68 / TIA 968-A.

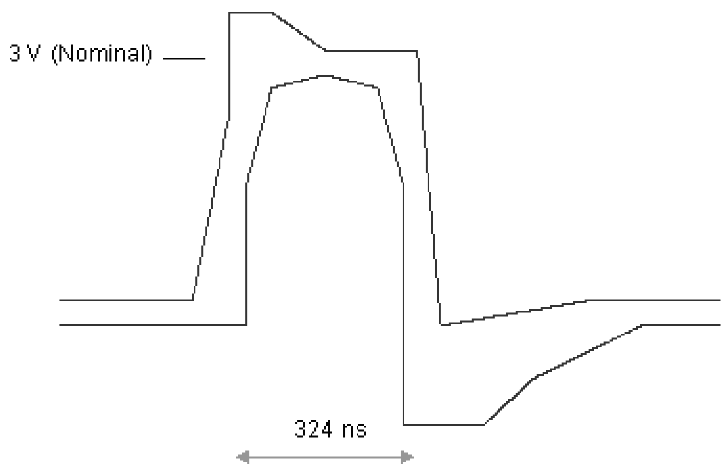


Figure 4. T1 (1.544 Mbps) Option B Pulse Shape per FCC Part 68 / TIA 968-A.

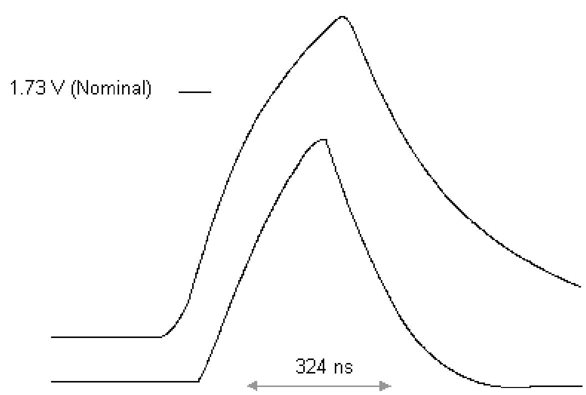
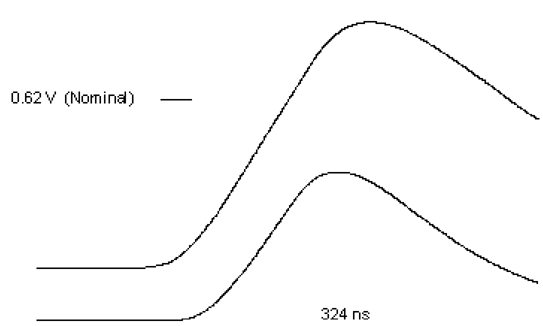


Figure 5. T1 (1.544 Mbps) Option C Pulse Shape per FCC Part 68 / TIA 968-A.



The equipment should be able to correctly receive the possible range of signals with characteristics representing the conditions that can exist on the line, such as losses, reflections and interference.
G.703 and TBR 4 require E1 interfaces to receive signals without errors for any combination of the following signal characteristics:
Carrier signal level: Nominal +/- 10%
Attenuation (Cable loss simulation): 0 - 6 dB at 1.024 MHz following sqrt(f) law
Reflection simulating signal mixed with the carrier: 18 dB below the carrier, 2^15-1 pattern.
TBR 4 requires also receiver tolerance to longitudinal (common mode) 2Vrms sine wave signals in the 10 Hz - 150 kHz frequency range.

If the impedance of the equipment is mismatched i.e. differs significantly from the line impedance it results in reflections and signal losses that can cause errors. Return loss measurement is used to access the impedance matching. According to G.703 requirements for E1 interfaces the return loss around the carrier frequency should at least 18 dB at the input port and 8 dB at least at the output port. For T1 interfaces the return loss requirements are not specified.

Since T1 and 120 ohm E1 interfaces use balanced lines, they should be balanced as well i.e. have symmetrical impedance between each lead and the ground. Lack of balance can cause crosstalk from and to adjacent leads. FCC Part 68/TIA-968-A requires transverse balance to be 35 dB at least in the 12 - 1544 kHz frequency range. TBR 4 requires 1 kohm minimum impedance towards ground in the 10 kHz - 500 kHz range and minimum 500 ohm in the 500 kHz - 1 MHz range.

PCM

Originally the digital T1 and E1 interfaces were designed to carry telephone voice calls within the telephone company network, allowing to transmit to greater distances with higher quality and to increase the network capacity, as opposed to the analog transmission and multiplexing techniques that were used in the older networks.
Today due to development of the Internet the digital lines are increasingly used for data transmission. Still voice calls account for major part of telecommunication.
In modern PSTNs the analog voiceband signal coming from the subscriber terminal equipments is converted to a digital signal at the Central Office, transmitted digitally through the PSTN and converted back to analog signal at the second subscriber end.
Usually it is cheaper to have one T1/E1 line than 20-25 or more separate analog lines. In this case multiple analog or digital telephones can be connected to the customer PBX. The customer PBX multiplexes/demultiplexes the calls between the single T1/E1 line and the telephones.

Figure 6. E1/T1 customer PBX connection

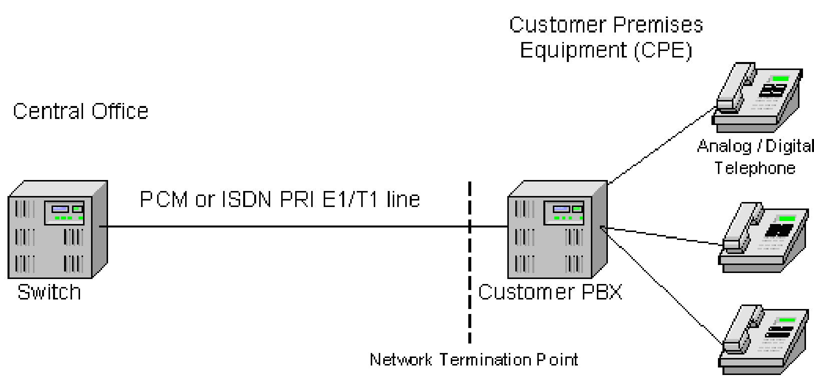
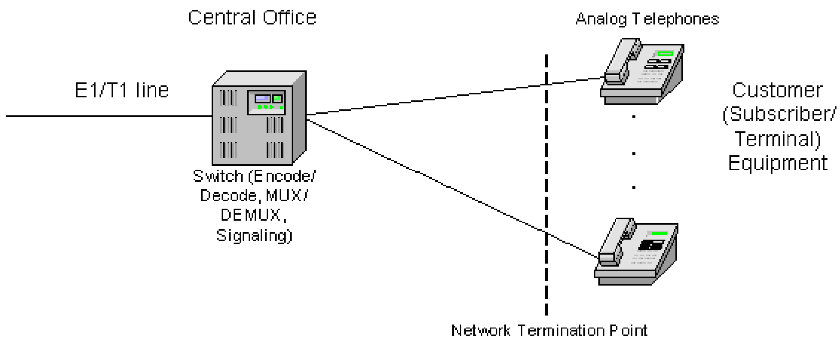


Figure 7. Central Office Analog/Digital Switch.



The analog to digital signal conversion, performed at the CO, customer PBX or in a digital telephone uses a technique called sampling, encoding or PCM (Pulse Code Modulation).

The analog signal is sampled at 125 us equal intervals. Each sample is a 8-bit word (also known as PCM word) which represents the amplitude of the signal at the time of sampling. The maximum level corresponds to about +3 dBm. There are total 2^8 = 256 sample levels. Due to this limited vertical resolution of the A/D conversion a certain amount of distortion is introduced to the signal. Since the lower signal levels suffer more from the vertical resolution error, non-linear conversion is used allocating more resolution to lower levels than to higher levels. Different non-linear level conversions are used for T1 and E1 systems.

U-law is used for T1 systems and A-law is used for E1 systems. U-law and A-law encoding is defined in ITU-T G.711.

There are 8000 samples per second (1/125 us = 8 kHz). According to the Nyquist–Shannon sampling theorem the minimum A/D sampling frequency should be twice the maximum frequency component of the analog signal in order to correctly reproduce the signal. Thus the 8 kHz frequency is just sufficient for sampling the 4 kHz voice band.

The bit rate of a single digital voice channel is 8000 samples/s x 8 bit = 64 kbps.

30 voice channels are transmitted over E1 and 24 channels are transmitted over T1 using Time Division Multiplexing (TDM).

E1 multiplexor takes 8 bits from each 64 kbps channels in turn and re-transmits them at 2048 kbps.

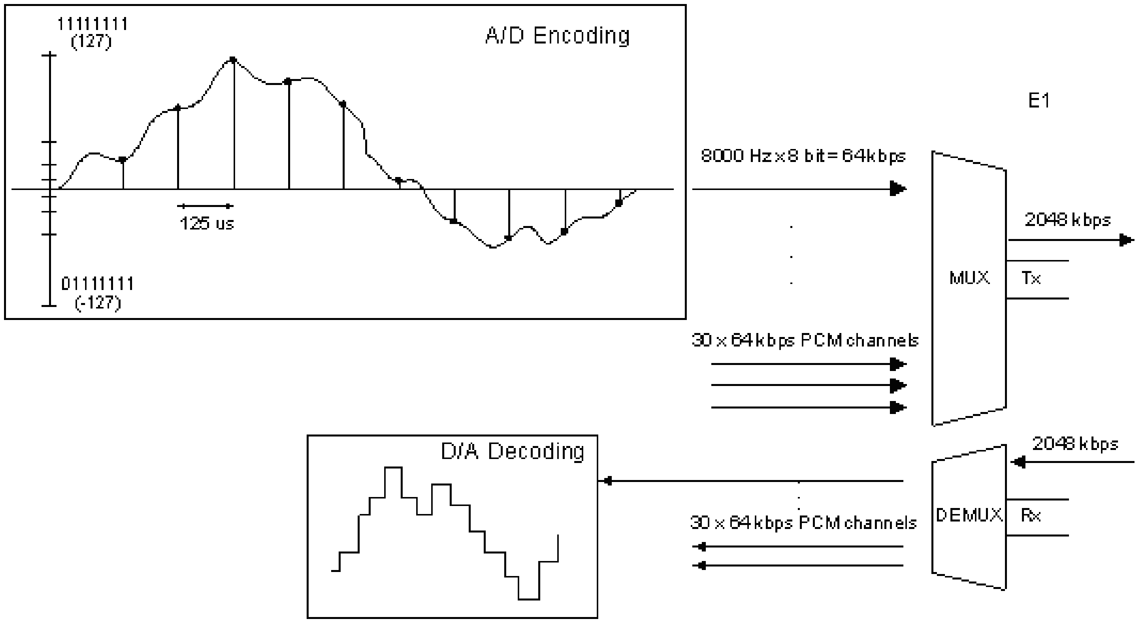
Besides the 30 voice channels there are two additional 64 kbps channels used to transmit synchronization, service and signaling data. Thus E1 transmission rate 2048 kbps = 32 * 64 kbps.

On the receive side of the PCM system the process is reversed - the E1 stream is demultiplexed into 64 kbps channels and analog voice signals are decoded from the 64 kbps channels.

To protect the PSTN from excessive signal levels, FCC Part 68/TIA-968-A limits the maximum encoded level representing analog signals by -12 dBm to -3 dBm and the gain (Net Amplification) of Analog-to-Digital, Digital-to-Analog, Digital-Digital through-transmission paths of T1 PCM system by -3 dB to +6 dB. The actual Encoded Analog Content and Net Amplification limits are set according to the type of the equipment. The 2600 Hz single-frequency (SF) signaling used for transmission of supervisory information between voice network and switches is protected by limiting the energy and gain in the frequency bands preceding and following the SF.

ITU-T G.712 limits additional A-A, A-D, D-A, D-D characteristics of E1 and T1 PCM systems such gain variation vs frequency and vs level, idle channel noise, crosstalk, distortion and group delay.

Figure 8. E1 PCM system.



E1 Frame structure

Each 8-bit sample or data word occupies a Timeslot in the E1 stream. A cycle of 32 timeslots repeats every 125 us. A complete cycle of the 32 timeslots is called a Frame. The timeslots in a frame are numbered TS0 to TS31. The first timeslot sent - the TS0 (also called overhead byte/timeslot/channel) contains synchronization and service data. The timeslots that carry user data are called payload.

Table 5. E1 timeslots.

TS0	TS1	....	TS16	....	TS31

A fixed 7-bit pattern "0011011" known as FAS (Frame Alignment Signal) is transmitted in timeslot 0 in each even frame. The second bit of timeslot 0 alternates between 0 and 1. When the receiver detects this pattern it achieves frame alignment ("knows" where each frame begins). If three or more incorrect FAS patterns are received in a row, the frame alignment is lost.

When frame alignment can't be achieved or no signal is received by the E1 equipment it transmits RAI (Remote Alarm Indication) by setting the A bits of the TS0 to 1 (see table 6) or by sending all ones pattern called AIS (Alarm Indication Signal). The AIS is usually sent by network equipment.

E1 CRC Multiframe structure

In addition to the frame structure, the CRC-4 Multiframe structure is recommended as an option by ITU-T G.704 and is mandatory for terminal (customer) equipment according to the European TBR 13, TBR 4 and the Australian TS016 standards.

Each multiframe (MF) consists of 16 consecutive frames divided into two 8-frame Sub Multiframes (SMF1 and SMF2).

When CRC is used a Multiframe alignment bit pattern "001011" (called MFAS - Multiframe Alignment Signal) is transmitted in the first bit of the timeslot 0 in the odd frames 1 to 11. This pattern is used by the receiver to synchronize on the multiframe structure.

CRC-4 (cyclic redundancy check) is an algorithm used for error checking. The transmitter performs a certain bit calculation on each SMF data it sends and puts the result of this calculation in the timeslot 0 of the next SMF. The CRC-4 result is 4 bits which are inserted in the TS0 first bit of the 4 even frames of the SMF. The receiver performs the same CRC-4 calculation and compares the result with the four CRC bits it received in the timeslot 0. If the bits don't match the receiver sets the E-bit alarm (first bit in TS0 in frames 13 and 15 is set to 0) notifying the transmitter that an error has occurred.

Table 6. CRC Multiframe.

	Sub-multiframe (SMF)	Frame number	Bits 1 to 8 of the frame (Timeslot 0)							
			1	2	3	4	5	6	7	8
Multiframe	**I**	**0**	C_1	0	0	1	1	0	1	1
		1	0	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
		2	C2	0	0	1	1	0	1	1
		3	0	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
		4	C3	0	0	1	1	0	1	1
		5	1	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
		6	C4	0	0	1	1	0	1	1
		7	0	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
	II	**8**	C_1	0	0	1	1	0	1	1
		9	1	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
		10	C_2	0	0	1	1	0	1	1
		11	1	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
		12	C_3	0	0	1	1	0	1	1
		13	E	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}
		14	C_4	0	0	1	1	0	1	1
		15	E	1	A	S_{a4}	S_{a5}	S_{a6}	S_{a7}	S_{a8}

NOTES
1 E = CRC-4 error indication bits (FEBE).
2 S_{a4} to S_{a8} = Spare bits (International bits). If not in use, they are usually set to 1.
3 C_1 to C_4 = Cyclic Redundancy Check-4 (CRC-4) bits.
4 A = Remote Alarm Indication (RAI).

Signaling

In E1 PCM and ISDN PRI systems the timeslot 16 is usually used to transmit the signaling information such as line status and call alerting (on-hook, off-hook, service request) addressing (destination and routing), calling party identification; employing. standard or proprietary protocols.
There are two distinct signaling modes with regards to the data structure utilization - CAS and CCS.
In CAS (Channel Associated Signaling) mode the multiframe (MF) consists of 16 consecutive frames. The first frame of the MF is identified by "0000" pattern in the first four bits of the timeslot 16. The 6th bit of the timeslot 16 in the first frame is used for alarm indication (Distant Multiframe alarm). This bit is set to 1 if the receiver does not detect the correct CAS multiframe structure. In each of the remaining 15 frames of the MF the timeslot 16 is divided into two 4-bit words (called abcd bits or CAS bits). The signaling data associated with the 30 payload channels is transmitted in those abcd bits. Channel 1 and channel 2 associated signaling is transmitted in the first and the second abcd words of the second frame, channel 3 and channel 4 signaling is transmitted in third frame and so on.
In the past the CAS bits were used for transmitting the number dialed. This method resulted in a slow connection establishment.
Today the dialed number (and other call information) is sent using DTMF or MF signals encoded in the calling payload channel (in-band signaling). A commonly used E1 CAS signaling is MFC-R2 (Multi-Frequency Compelled R2) uses ABCD bits for supervisory information (on hook/off hook) and MF for register addressing signaling (called/calling party number, call type, etc).

Table 7. CAS multiframe.

Frame	TS 0									TS 16									TS 31							
0	Si	0	0	1	1	0	1	1	...	0	0	0	0	X	Y	X	X	...								
1	Si	0	0	Sn	Sn	Sn	Sn	Sn	...	a	b	c	d	a	b	c	d	...								
										Channel 1				Channel 16												
2	Si	0	0	1	1	0	1	1	...	a	b	c	d	a	b	c	d	...								
										Channel 2				Channel 17												
3	Si	0	0	Sn	Sn	Sn	Sn	Sn	...	a	b	c	d	a	b	c	d	...								
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14	Si	0	0	1	1	0	1	1	...	a	b	c	d	a	b	c	d	...								
15	Si	0	0	Sn	Sn	Sn	Sn	Sn	...	a	b	c	d	a	b	c	d	...								
										Channel 15				Channel 30												

The CCS (Common Channel Signaling) users higher level messages for signaling. The signaling messages are in most cases based on HDLC (High Level Data Control) protocol. In E1 systems timeslot 16 is used for CCS signaling.
CCS is much more efficient than CAS.
ISDN PRI messages encoded in timeslot 16 is one example of common channel signaling and SS7 (Signaling System 7) is another example.
An E1 link with 30 64 kbps user channels has total 1.920 Mbps payload capacity.

T1 framing structure

The T1 interface can provide up to 24 64 kbps payload channels (timeslots).
A T1 frame consists of 24 8-bit timeslots and one framing bit. Thus there are 193 bits (24*8+1) in the T1 frame.
The "F" framing bit (the first bit in the frame) is used for synchronization and basic error indication.

Table 8. T1 frame.

F bit	Channel 1	....	Channel 24

The framing bit takes up 8 kbps reducing the useful capacity from 1.544 Mbps to 1.536 Mbps (24 x 64 kbps = 1.536 Mbps).
Multiple frames are grouped together to form structures called superframes.
There are two main superframe formats in use - SF (superframe also known as D4 or 193S) and ESF (Extended Superframe or 193E).
Services that give users a portion of T1 bandwidth are provided and called Fractional T1 or subrate service.
A T1 customer equipment (known as CPE - Customer Premises Equipment or Terminal Equipment) which implements the T1 layer 1 interface functionality is called CSU (Channel Service Unit). Equipment that connects DTE (Data Terminal Equipment such as PC) to T1 line is called DSU. In many cases these two functions are combined in one device, known as CSU/DSU.

SF (D4) format

The SF (Superframe) structure consists of 12 consecutive frames called superframe. The framing bits of the 12 frames form a fixed bit sequence 100011011100. The odd bits in the sequence (101010) are called terminal framing bits or Ft bits, and the even bits (001110) are called signaling frame bits or Fs bits.
When no incoming signal is detected Blue alarm of AIS (Alarm Indication Signal) is sent by transmitting all ones unframed pattern.
Yellow alarm (also called RAI - Remote Alarm Indication) is sent, by setting the second bit of each timeslot to 0, when receiver framing synchronization is lost.

ESF format

The ESF Extended Superframe structure consists of 24 consecutive frames.
Of the 24 framing bits, six bits (every fourth frame, starting from frame #4) are used for the 001011 framing synchronization pattern. Another 6 bits (in frames 2,6,10,14,18,12) are used to transmit the results of CRC-6 calculation performed on the previous superframe data. The remaining 12 bits of the framing bit sequence are used to create a 4 kbps channel called FDL (Facilities Data Link). The FDL channel is used for alarms and performance reports. It can be also used for signaling, control or other (user) messages.
The Yellow alarm (RAI) is sent over the FDL channel, when receiver framing synchronization is lost, by transmitting 11111111 00000000 word. The AIS/Blue alarm is sent by transmitting all ones unframed pattern.

Table 9. SF - Superframe Format (D4)

	F bits			Bit use in each channel time slot		
Frame number within superframe	Bit number within Superframe	Terminal Framing Bit (Ft)	Signaling Bit (Fs)	Data	Robbed-Bit Signaling [a)]	Signaling Channel (Signaling bit) [a)]
1	1	1	-	1-8	-	
2	194	-	0	1-8	-	
3	387	0	-	1-8	-	
4	580	-	0	1-8	-	
5	773	1	-	1-8	-	
6	966	-	1	1-7	8	A
7	1159	0	-	1-8	-	
8	1352	-	1	1-8	-	
9	1545	1	-	1-8	-	
10	1738	-	1	1-8	-	
11	1931	0	-	1-8	-	
12	2124	-	0	1-7	8	B

NOTES:

1. A Superframe consists of 12 consecutive frames. The F bits are used for framing only, and are divided into two groups.
   a. Terminal framing (Ft) bits are used to identify frame boundaries.
   b. Signal framing (Fs) bits are used to identify Superframe boundaries.
2. If robbed-bit signaling is not implemented, all 8 bits may be available for data.
3. Frames 6 and 12 are called signaling frames.

a) Only applicable in the case of channel associated signalling (CAS Robbed-Bit signaling). If Robbed-Bit signaling is not used, all 8 bits of each channel timeslot are available for data.

Table 10. Extended Superframe Format (ESF)

	F-bit						
Frame number	Bit number within	Assignments			Bit use in each channel time slot		Signalling channel
within Extended Superframe	Extended Superframe	FAS	FDL	CRC	Data	Robbed-Bit Signaling[a)]	(Signaling bit)[a)]
1	1	–	m	–	1-8	–	
2	194	–	–	C1	1-8	–	
3	387	–	m	–	1-8	–	
4	580	0	–	–	1-8	–	
5	773	–	m	–	1-8	–	
6	966	–	–	C2	1-7	8	A
7	1159	–	m	–	1-8	–	
8	1352	0	–	–	1-8	–	
9	1545	–	m	–	1-8	–	
10	1738	–	–	C3	1-8	–	
11	1931	–	m	–	1-8	–	
12	2124	1	–	–	1-7	8	B
13	2317	–	m	–	1-8	–	
14	2510	–	–	C4	1-8	–	
15	2703	–	m	–	1-8	–	
16	2896	0	–	–	1-8	–	
17	3089	–	m	–	1-8	–	
18	3282	–	–	C5	1-7	8	C
19	3475	–	m	–	1-8	–	
20	3668	1	–	–	1-8	–	
21	3861	–	m	–	1-8	–	
22	4054	–	–	C6	1-8	–	
23	4247	–	m	–	1-8	–	
24	4440	1	–	–	1-7	8	D

FAS Frame alignment signal (. . . 001011 . . .).
FDL 4 kbit/s Facility Data Link (message bits m).
CRC CRC-6 block check field (check bits C1 to C6).
a) Only applicable in the case of channel associated signalling (CAS Robbed-Bit signaling). If Robbed-Bit signaling is not used, all 8 bits of each channel timeslot are available for data.

Robbed-Bit signaling

Bit robbing (or Robbed bit signaling) is a technique, which is commonly used in T1 PCM systems for telephone call control signaling. The least significant bit of each channel in each sixth frame is used to transmit the signaling data. Such "stealing" of the least significant bit has a negligible effect on the voice quality. Changing bits is unacceptable, though, with data transmission. Thus when robbed-bit signaling is used with T1 data communication, the least

significant bit is ignored which reduces the throughput of each channel from 64 kbps to 56 kbps. The total available capacity of a T1 link falls from 1.536 Mbps to 1.344 Mbps.
Since every sixth frame is used for signaling, the SF-framed links use two bits (A, B bits) and the ESF-framed links can use four bits (A, B, C, D bits) for signaling associated with each voice channel.
The Robbed-Bit signaling is CAS (Channel Associated Signaling).

E & M wink start signaling is one of the procedures commonly used in the US for signaling between two switches (CO or local PBX) connected by a T1 link.
Initially the two switches are in the idle on-hook state. The on-hook state is identified by both switches sending the A,B signaling bits set to 0. The switch originating a call goes to off-hook state which is signaled by sending A=1, B=1 to the destination switch. After the originating switch goes off-hook, it waits for acknowledgement from the destination switch. That acknowledgement comes when the destination switch "winks" off-hook" for 140-200 ms (i.e. goes off-hook for 140 - 200 ms and returns to on-hook). When the originating switch receives the wink it can transmit the destination telephone number after a delay of at least 210 ms.
The number is transmitted using MF (multi-frequency) or DTMF signals encoded in the calling channel.
After receiving the telephone number the destination switch goes off-hook (sends A=1, B=1) and the call connection is established.
This three-stage handshaking procedure is used to prevent a problem (called glare) arising from the case when both ends try to use the same channel at the same time?
Other types of Robbed-Bits (CAS) signaling are Ground Start, Loop Start and E&M immediate.

The call information (called/calling numbers, and so on) is transferred using tones in the time slot being used for the call (in-band signaling).
R1 signaling uses six signaling frequencies that are 700 to 1700 Hz in 200-Hz steps. These signals are called multifrequency inter-register signals. Each signal is a combination of two frequencies out of set six frequencies.

Rather than implementing bit-robbing, ISDN PRI uses the twenty-fourth 64 kbps channel for (CCS) signaling. This channel is called D-channel. The remaining 23 channels are called B channels (bearer channels). Multiple PRIs can share a single D channel. Such sharing of a signaling channel by several systems is known as NFAS (Network Facilities Associated Signaling). Those PRIs that don't have the D channel can use all 24 channels for payload data.

Circuit-switched versus Packet-switched networks

In circuit-switched communication after the connection (or a call) has been established the predefined network path is fully reserved, while communication lasts. The older PCM based PSTN and ISDN are examples of circuit-switched networks. Such circuit-switched network is optimized for uncompressed, real-time voice communications. But it is also inefficient, since the path is committed during the connection even when no information is transmitted. Packet switching was originally created for data sending. The data to be send is sliced into small pieces called packets. The packets can be sent via different network routes or logical channels, depending on the availability of such routes. The packet-switching uses the network bandwidth much more efficiently than the circuit-switching and thus it is cheaper. On the other hand breaking up the data and reassembling the packets arriving at different time at the other end, introduces some delay (called latency). For data communication this latency is barely noticeable. For real-time communication such as voice or video, the latency can be noticeable. That is why packet switching was not used for real-time applications until recently when ATM and IP telephony were introduced.

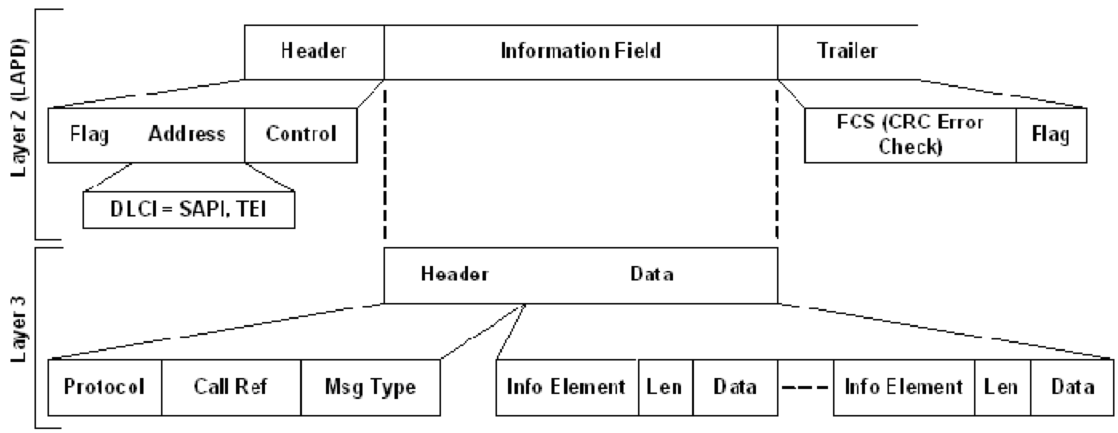
ISDN

ISDN (Integrated Services Digital Network) is an end-to-end digital network. It can be used for basic-quality 3.4 kHz voice, better quality 7 kHz voice, data and video communication. Various combinations of these communication types can be provided simultaneously over the same ISDN link. ISDN provides mostly for circuit-switched communication.

There are two main types of ISDN services in use - BRI and PRI.
ISDN BRI (Basic Rate Interface) provides 128 kbps of user bandwidth in two 64 kbps bearer (B) channels and one 16 kbps D channel used for signaling. ISDN BRI physical line (layer 1) is different from E1 and T1 lines.
ISDN PRI (Primary Rate Interface) uses E1 (30B+D) or T1 (23B+D) as its physical layer. The ISDN-specific features are provided by layer 2 and layer 3 definitions for signaling. The signaling is implemented through data messages coded into the 64 kbps D channel, which is timeslot 16 in E1 and timeslot 24 in T1.
The data link layer (layer 2) describes message delimitation into frames, error detection and correction, flow control and subdivision of the D channel into multiple logical channels. The data link layer also called LAP-D (Link Access Procedure D Channel) is specified in ITU-T Q.921.

The network layer (layer 3) provides a set of coded messages to be exchanged between user and network-switch equipment for call establishment and disconnection control. The network layer messages are carried within the layer 2 frames. The ITU-T Q.931 gives layer 3 specifications.

Figure 9. ISDN Layer 2 and Layer 3 structure.



ISDN Layer 2 - Q.921 (LAPD)

Address field

DLCI - Data Link Connection Identifier includes SAPI and TEI.
SAPI - Service Access Point Identifier points to particular type of processing (service) required.

TEI - (Terminal Equipment Identifier) identifies user terminal equipment.
TEI values:
0-63 Used by non-automatic TEI assignment user equipment.
64-126 Used by automatic TEI assignment equipment.
127 Used for a broadcast connection meant for all Terminal Endpoints.
Other address subfields:
EA1 First Address Extension bit which is always set to 0.
C/R Command/Response bit. Frames from the user with this bit set to 0 are command frames, as are frames from the network with this bit set to 1. Other values indicate a response frame.
EA2 Second Address Extension bit which is always set to 1.
Control field identifies frame type (Supervisory and Unnumbered frame types) and includes sequence numbers, control features and error tracking according to the frame type.
FCS field (Frame Check Sequence) carries CRC results inserted by the transmitter. The receiver performs the same CRC calculation and compares the results with ones received in this field.

ISDN Layer 3 - Q.931

Protocol -Protocol Discriminator, usually 08, meaning Q.931 call maintenance
Call Ref - Call Reference Value (CRV). In PRI is a 15-bit value which is used to associate messages with a particular channel connection. Also there are Length of reference call and flag fields in layer 3 header.
Msg Type - Message type. There are four general categories of messages:

1. Call Establishment
2. Call Information
3. Call Clearing
4. Miscellaneous

Table 11. ISDN Layer 3 messages.

	Message type	Message	Message value
1	Call Establishment		
		Alerting	000 00001
		Call Proceeding	000 00010
		Progress	000 00011
		Setup	000 00101
		Connect	000 00111
		Setup Acknowledge	000 01101
		Connect Acknowledge	000 01111
2	Call Information		
		User Information	001 00000
		Suspend Reject	001 00001
		Resume Reject	001 00010
		Hold	001 00100
		Suspend	001 00101
		Resume	001 00110
		Hold Acknowledge	001 01000
		Suspend Acknowledge	001 01101
		Resume Acknowledge	001 01110
		Hold Reject	001 10000
		Retrieve	001 10001
		Retrieve Acknowledge	001 10011
		Retrieve Reject	001 10111
2	Call Clearing		
		Disconnect	010 00101
		Restart	010 00110
		Release	010 01101
		Restart Acknowledge	010 01110
		Release Complete	010 11010
	Miscellaneous		
		Segment	011 00000
		Facility	011 00010
		Register	011 00100
		Notify	011 01110
		Status inquiry	011 10101
		Congestion Control	011 11001
		Information	011 11011
		Status	011 11101

Information Elements can be either single octet or variable-length

Single octet information elements:
1 000 ---- Reserved
1 001 ---- Shift
1 010 0000 More data

1 010 0001 Sending Complete
1 011 ---- Congestion Level
1 101 ---- Repeat indicator

Variable-length information elements:
0 0000000 Segmented Message
0 0000100 Bearer Capability (Identifies requirements of the requested B channel)
0 0001000 Cause (Identifies reasons for disconnect or incomplete calls)
0 0010100 Call identify
0 0010100 Call state
0 0011000 Channel identification (Identifies requested B channel)
0 0011100 Facility
0 0011110 Progress indicator (Indicates status of outgoing call)
0 0100000 Network-specific facilities
0 0100111 Notification indicator
0 0101000 Display
0 0101001 Date/time
0 0101100 Keypad facility
0 0110100 Signal
0 0110110 Switchhook
0 0111000 Feature activation
0 0111001 Feature indication
0 1000000 Information rate
0 1000010 End-to-end transit delay
0 1000011 Transit delay selection and indication
0 1000100 Packet layer binary parameters
0 1000101 Packet layer window size
0 1000110 Packet size
0 1101100 Calling party number
0 1101101 Calling party subaddress
0 1110000 Called party number (destination number)
0 1110001 Called Party subaddress
0 1110100 Redirecting number
0 1111000 Transit network selection
0 1111001 Restart indicator
0 1111100 Low layer compatibility
0 1111101 High layer compatibility
0 1111110 User-user
0 1111111 Escape
Other values Reserved

Table 12. ISDN call example.

#	Hex	Bin	Description
1	Calling Terminal Equipment sends a SETUP to the Network Switch:		
	02010000		Layer 2 header
	Layer 3:		
	08	00001000	Protocol discriminator - Q931
	01 2B	00000001 00101011	Call Reference Value allocated by user side
	05	00000101	Message Type = Setup
	04	00000100	Bearer capability
	03	00000100	Length = 3 octets
	80	10000000	Speech
	90	10010000	64 kbps circuit mode
	A2	10100010	u-law
	18	00011000	Channel ID

#	Hex	Bin	Description
	03	00000100	Length = 3 octets
	A9	10101001	This interface, PRI, exclusive, not D, B1
	83	10000011	CCITT, number, B-channels
	89	1 0001001	Slot 15
	1E	00011110	Progress indicator
	6C	01101100	Calling party number
	09	00001001	Length = 9 octets
	41	01000001	Local ISDN number
	A0	10100000	Presentation prohibited
	37	00110111	digit 7
	36	00110110	digit 6
	38	00111000	digit 8
	35	00110101	digit 5
	30	00110000	digit 0
	37	00110111	digit 7
	30	00110000	digit 0
	70	01110000	Called party number
	08	00001000	Length = 8 octets
	41	01000001	Local ISDN number
	38	00111000	digit 8
	33	00110011	digit 3
	34	00110100	digit 4
	35	00110101	digit 5
	30	00110000	digit 0
	32	00110010	digit 2
	31	00110001	digit 1
2	If the SETUP is correct, the Network Switch sends a CALL PROCEEDING to the Calling Terminal Equipment, and then sends a SETUP to the Receiver.		
3	The Receiver gets the SETUP. If it is correct it indicates alert (e.g. rings a phone) and sends an ALERTING MESSAGE to the Network Switch.		
4	The Network Switch forwards the ALERTING message to the Caller.		
5	When the receiver answers the call, it sends a CONNECT message to the Network Switch.		
6	The Network Switch forwards the CONNECT message to the Caller.		
7	The Caller sends a CONNECT ACKNOWLEDGE message to the switch.		
8	The Network Switch forwards the CONNECT ACKNOWLEDGE message to the Receiver.		
	The call establishment is completed.		

ETSI TBR 4 standard defines layer 1,2,3 requirements and tests for E1 ISDN PRI terminal equipment.
The layer 2 and 3 tests are specified as a series of TTCN test cases.

TTCN (Tree and Tabular Combined Notation) is a programming–like language that describes test procedure including protocol simulation, analysis, and verdict criteria. The TTCN test cases are intended for machine processing.
Different flavors of ISDN are in use in US and other countries, varying by Q.921 and Q.931 based supported features and services.

Table 13. Comparison of regulatory standard requirements for E1 interface.

Regulatory Standards						ITU Standards
	European TBR 4 for ISDN PRI Terminal Equipment	**European TBR 13, Framed interfaces**	**European TBR 12, Unframed interfaces**	**Australia (AS/ACIF S016)**	**Japan JATE**	**ITU-T G.703, G.704, G.706, G.711, G.712, G.823**
Layer1						
Specification at the output port						
Waveform shape	Positive and Negative pulse shape mask, Space peak voltage, Pulse amplitude ratio, Pulse width ratio	Same as TBR 4	Same as TBR 4	Positive and Negative pulse shape mask, Space peak voltage	Positive and Negative pulse shape mask	Positive and Negative pulse shape mask, Space peak voltage, Pulse amplitude ratio, Pulse width ratio
Bit rate/clock accuracy	+/- 50 ppm	+/- 50 ppm	+/- 50 ppm	+/- 50 ppm		+/- 50 ppm
Line Coding	HDB3	HDB3	HDB3	HDB3	HDB3	HDB3
Output Jitter	Intrinsic Jitter, Output Jitter vs Jitter input to Data/Clock with carrier frequency offset	Intrinsic Jitter, Output Jitter vs Jitter input to Data/Clock with carrier frequency offset	Intrinsic Jitter, Output Jitter vs Jitter input to Data/Clock with carrier frequency offset	Intrinsic Jitter		G.823: ntrinsic Jitter, Output Jitter vs Jitter input to Data/Clock with carrier frequency offset. Jitter Transfer.
Specification at the input port						
Return loss	51 to 102 kHz: 12 dB 102 to 2048 kHz: 18 dB 2048 to 3072 kHz: 14 dB			Same as TBR 4		Same as TBR 4
Immunity to attenuation and reflections	Cable simulation 0 – 6 dB, - 18 dBc interfering signal, Nominal pulse amplitude +/- 10 %			Cable simulation 0 – 6 dB, - 18 dBc interfering signal, Nominal pulse amplitude +/- 10 %		Cable simulation 0 – 6 dB, - 18 dBc interfering signal, Nominal pulse amplitude +/- 10 %
Receiver sensitivity	-			Cable simulation 0 – 6 dB, - 18 dBc		Cable simulation 0 – 6 dB, - 18 dBc
Jitter tolerance	Tolerance to input jitter in 20 Hz – 100 kHz			Jitter/Wander tolerance		Jitter/Wander tolerance (G.823)
Tolerable longitudinal voltage	Immunity to longitudinal 2Vrms sine signal in 10Hz – 150kHz					
Impedance towards ground	At least 1kohm to 500ohm at 10Hz – 1MHz					
Frame Structure	Use of bit 1 in 2048 kbit/s CRC-4 multiframe against simulation with and without errors	Frame and MF Structure, Use of A bit				Frame and MF Structure (G.704)

Regulatory Standards						ITU Standards
	European TBR 4 for ISDN PRI Terminal Equipment	**European TBR 13, Framed interfaces**	**European TBR 12, Unframed interfaces**	**Australia (AS/ACIF S016)**	**Japan JATE**	**ITU-T G.703, G.704, G.706, G.711, G.712, G.823**
Operational functions	Maintenance functions, Loss of frame alignment, Strategy for frame alignment recovery, CRC multiframe alignment, CRC bit monitoring, Monitoring for false frame alignment					RAI, Loss of frame alignment, Strategy for frame alignment recovery, CRC multiframe alignment, CRC bit monitoring, Monitoring for false frame alignment (G.706)
PCM						G.711, G.712: a-law encoding/decoding, A-A, A-D, D-A, D-D Levels, Load capacity, Variation of loss vs time, Impedance, Return loss, LCL, Attenuation/Frequency distortion, Group delay, Group delay distortion, Idle channel noise, SF interference, In-band, out-of-band input and output signals, Spurious signals, THD, Gain variation, Crosstalk, Interference from signaling, Echo, TBRL, SL
CAS signaling						Q.4xx series standards
Overvoltage protection	Impulse transfer from mains, common mode, Impulse transfer from mains, transverse mode, Conversion of common mode to transverse mode	Same as TBR 4	Same as TBR 4	Same as TBR 4	Same as TBR 4	Same as TBR 4
ISDN PRI Layer 2	TTCN-based simulation and monitoring tests	-	-	-	-	Q.921
ISDN PRI Layer 3	TTCN-based simulation and monitoring tests	-	-	-	-	Q.931

Table 14. Comparison of regulatory standard requirements for T1 interface.

	Regulatory standards		ANSI standards	ITU standards
	US FCC Part 68 / TIA-968-A	**Japan JATE**	**ANSI T1.403**	**ITU-T G.703, G.704, G.706, G.711, G.712, G.823, I.431**
Layer1				
Signal Power Limitations				
Pulse shape	Options A, B, C		Pulse shape, Pulse imbalance, 60 Hz variations in pulse amplitude	
Output power	Power at 772 and second harmonic power with 3 kHz BW			
Bit rate	+/- 32 ppm		+/- 32 ppm	+/- 32 ppm
Line coding			AMI (with minimum pulse density), B8ZS	AMI, B8ZS
Line Buildout	Options A, B, C		Options A, B, C	
PCM				
Encoded analog content	Maximum encoded power < -3 dBm			G.711, G.712: a-law encoding/decoding, A-A, A-D, D-A, D-D Levels, Load capacity, Variation of loss vs time, Impedance, Return loss, LCL, Attenuation/Frequency distortion, Group delay, Group delay distortion, Idle channel noise, SF interference, In-band, out-of-band input and output signals, Spurious signals, THD, Gain variation, Crosstalk, Interference from signaling, Echo, TBRL, SL
Net amplification	Maximum A-A, A-D, D-D gain			
SF cutoff	Signals with energy in the 2450 to 2750 Hz band are not through transmitted unless there is at least an equal amount of energy in the 800 to 2450 Hz band within 20 ms of application of a signal			
Through transmission – SF/Guard band	The A-A, A-D, D-D in the 800 to 2450 Hz band shall not exceed the loss at any frequency in the 2450 to 2750 Hz band by more than 1 dB			
Jitter			Output Jitter (10 Hz – 40 kHz),Output Wander (ANSI T1.101)	G.824: Jitter and Wander tolerance, Intrinsic jitter, jitter transfer
Return Loss			Receiver Return loss > 18 dB in 100 kHz – 1.544 MHz	
Transverse Balance	Transverse Balance > 35 dB in 12 kHz – 1.544 MHz		Transverse Balance > 35 dB in 100 kHz – 1.544 MHz	
Billing protection				
Signaling interference	TE shall not deliver digital signals to the telephone network with encoded analog content energy in the 2450 to 2750 Hz band unless at least an equal amount of encoded analog energy is present			

	Regulatory standards		ANSI standards	ITU standards
	US FCC Part 68 / TIA-968-A	**Japan JATE**	**ANSI T1.403**	**ITU-T G.703, G.704, G.706, G.711, G.712, G.823, I.431**
	in the 800 to 2450 Hz band for the first two seconds after going to the off-hook state			
On-hook level	< -55 dBm			
Off-hook signal requirements	The off-hook shall be transmitted at least 5 s as response to alerting			
Operating requirements for direct inward dialing	The off-hook state shall be transmitted not later than 0.5 s after answering a call.			
Framing			SF and ESF formats, RAI, AIS	G.704
CAS Signaling			Robbed bit signaling defined in T1.403.02, T1.401, T1.401.01, T1.405, T1.407, T1.409, T1.411	Q.310 - Q.331
Maintenance			Loopback activation and control, Data Link, HDLC, LAPD messages, error and performance reports	I.431
Hearing aid compatibility	Magnetic field			
Equal access to common carriers				
Environmental Simulation				
Mechanical shock	Drop test			
Surges				
Leakage current and Hazardous voltage limitations	Leakage current, General (T&R), Physical separation of leads, Intentional Operational Paths to Ground, Intentional Protective Paths to Ground			