

IEEE Standard Methods and Equipment for Measuring the Transmission Characteristics of Pulse-Code Modulation (PCM) Telecommunications Circuits and Systems

Sponsor
**Transmission Systems Committee
of the
IEEE Communications Society**

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Abstract: Test equipment requirements and methods for testing the transmission characteristics of PCM telecommunications equipment, circuits, and systems are set forth. The requirements are intended for certification, installation, pre-service, out-of-service operational, and in-service operational tests of the PCM transmission facilities. The PCM equipment that may be tested includes primary multiplex equipment containing analog-digital conversion devices, digital multiplex equipment, digital links, and digital sections. This standard is limited to testing at the analog interfaces of the primary multiplex equipment and the digital interfaces at DS1, DS1C, DS2, and DS3 levels of the North American digital hierarchy. Synchronous multiplex equipment and equipment offering other than 64 kb/s coded voiceband channels are not covered in this standard, nor is signaling parameter measurement.

Keywords: analog-digital devices, digital multiplex equipment, digital links, digital sections, multiplex equipment, PCM telecommunications equipment, transmission characteristics

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Foreword

(This Foreword is not a part of IEEE Std 1007-1991, IEEE Standard Methods and Equipment for Measuring the Transmission Characteristics of Pulse-Code Modulation (PCM) Telecommunications Circuits and Systems.)

In 1984, IEEE Std 743-1984, IEEE Standard Methods and Equipment for Measuring the Transmission Characteristics of Analog Voice Frequency Circuits (ANSI), was issued to ensure that analog voice frequency circuits could be measured with meaningful and repeatable results. The pervasive use of pulse-code modulation (PCM) systems necessitated additional standards to ensure that telecommunications circuits and systems using PCM to digitally transmit analog voice frequency signals also could be measured in ways that were meaningful and repeatable. This standard was developed to provide those additional methods and procedures.

Suggestions for improvement of this standard will be welcome. They should be sent to the

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This standard was prepared by the PCM Measurements Working Group of the Subcommittee on Telecommunications Measurements and Standards of the IEEE Communications Society Standards Coordinating Committee. During the approval process, this working group was reassigned to the Transmission Systems Committee of the IEEE Communications Society. The members of the PCM Measurements Working Group at the time this standard was prepared were as follows:

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IEEE Standard Methods and Equipment for Measuring the Transmission Characteristics of Pulse-Code Modulation (PCM) Telecommunications Circuits and Systems

1. Introduction

1.1 Scope and Objectives

The purpose of this standard is to provide test equipment requirements and methods for testing the transmission characteristics of PCM telecommunications equipment, circuits, and systems. The test equipment requirements are intended to be suitable for certification, installation, pre-service, and out-of-service operational and in-service operational tests of the PCM transmission facilities. The PCM equipment that may be tested includes primary multiplex equipment containing A-D (Analog-Digital) conversion devices, digital multiplex equipment, digital links, and digital sections. This document is currently limited to testing at the analog interfaces of the primary multiplex equipment and the digital interfaces at DS1, DS1C, DS2, and DS3 levels of the North American digital hierarchy. Synchronous multiplex equipment and equipment offering other than 64 kb/s coded voiceband channels (e.g., 32 kb/s voiceband coding, 256 kb/s program, or 64 kb/s data) are not covered in this standard. Signaling parameter measurement is outside the scope of this document.

1.2 References

This standard shall be used in conjunction with the following publications. When the following standards are superseded by an approved revision, the revision shall apply.

[1] IEEE Std ANSI T1.101-1987, American National Standard for Telecommunications-Synchronization Interface Standards for Digital Networks.¹

¹ANSI publications are available from the Sales Department, American National Standards Institute, 11 West 42nd Street, 13th Floor, New York, NY 10036 USA.

- [2] IEEE Std ANSI T1.102-1987, American National Standard for Telecommunications—Digital Hierarchy—Electrical Interfaces.
- [3] IEEE Std ANSI T1.107-1988, American National Standard for Telecommunications—Digital Hierarchy—Formats Specifications.
- [4] IEEE Std ANSI T1.403-1989, American National Standard for Telecommunications—Carrier-to-Customer Installation—Metallic Interface.
- [5] IEEE Std ANSI T1.404-1989, American National Standard for Telecommunications—Carrier-to-Customer Installation—DS3 Metallic Interface Specification.
- [6] IEEE Std ASTM D 2197-1986, Test Methods for Organic Coatings by Spray Adhesives.²
- [7] IEEE Std EIA Std RS-232-E (July 1991), Interface Between Data Terminal Equipment and Circuit Terminal Equipment Employing Serial Binary Data Interchange.³
- [8] IEC 320 (1981), Appliance couplers for household and similar general purposes.⁴
- [9] IEC 348 (1978), Safety requirements for electronic measuring apparatus.
- [10] IEEE Std 100-1988, IEEE Standard Dictionary of Electrical and Electronic Terms — 4th ed. (ANSI).⁵
- [11] IEEE Std 455-1985, IEEE Standard Test Procedure for Measuring Longitudinal Balance of Telephone Equipment Operating in the Voice Band (ANSI).
- [12] IEEE Std 488.1-1987, IEEE Standard Digital Interface for Programmable Instrumentation (ANSI).
- [13] IEEE Std 743-1984, IEEE Standard Methods and Equipment for Measuring the Transmission Characteristics of Analog Voice Frequency Circuits (ANSI).
- [14] AT&T Bell System Technical Reference, Pub. 43801, *Digital Channel Bank Requirements and Objectives*, Nov. 1982.⁶
- [15] AT&T Bell System Technical Reference, Pub. 43802, *Digital Multiplexes—Requirements and Objectives*, July 1982.
- [16] Bellcore Technical Reference, TR-TSY-000009, *Asynchronous Digital Multiplexes Requirements and Objectives*, Issue 1, Dec. 1984.⁷
- [17] Bellcore Technical Reference, TR-TSY-000064, *LATA Switching Systems Generic Requirements (LSSGR)*, Issue 2, July 1987.

²ASTM publications are available from the Customer Service Department, American Society for Testing and Materials, 1916 Race Street, Philadelphia, PA 19103, USA.

³EIA publications are available from the Electronics Industries Association, Standards Sales, 2001 Pennsylvania Avenue NW, Washington, DC 20006, USA.

⁴IEC publications are available from IEC Sales Department, Case Postale 131, 3 rue de Varembe, CH 1211, Genee 20, Switzerland/Suisse. IEC publications are also available in the United States from the Sales Department, American national Standards Institute, 11 West 42nd Street, 13th Floor, New York, NY 10036, USA.

⁵IEEE publications are available from the Institute of Electrical and Electronics Engineers, Service Center, 445 Hoes Lane, P. O. Box 1331, Piscataway, NJ 08855-1331, USA.

⁶AT&T Technical Reference publications are available from AT&T Technologies, Inc., Customer Information Center, P. O. Box 19901, Indianapolis, IN 46219.

⁷Bellcore Technical Reference publications are available from Bellcore, 60 New England Avenue, Piscataway, NJ 08854.

- [18] Bellcore Technical Reference, TR-TSY-000191, *Alarm Indication Signal Requirements and Objectives*, Issue 1, May 1986.
- [19] “Characteristics of Digital Multiplex Equipments Based on a Second Order Bit Rate of 6312 kbit/s and Using Positive Justification.” 1988 CCITT Recommendations, Vol. III-Fascicle III.3, Rec G. 752.⁸
- [20] “Digital Networks—Transmission Systems and Multiplexing Equipment.” 1988 CCITT Recommendations, Vol. III—Fascicle III.3, Rec G. 732.
- [21] “Pulse Code Modulation (PCM) of Voice Frequencies.” 1988 CCITT Recommendations, Vol. III—Fascicle III.3, Rec G.711.
- [22] “Specification for Equipment to Measure the Performance of PCM Encoders and Decoders.” 1988 CCITT Recommendations, Vol. IV—Fascicle IV.4, Rec 0.133.
- [23] “Specifications of Measuring Equipment.” 1988 CCITT Recommendations of the O Series, Volume IV—Fascicle IV.4.
- [24] “Specifications of Signaling System No. 7.” 1988 CCITT Recommendations, Vol. VI-Fascicle VI.7 and VI.8.
- [25] T1X1.4/91-004R1, July 1991, Digital Hierarchy—Formats Specifications.⁹
- [26] U.S. Dept. of Radiational Health, Code of Federal Regulations, 21 CFR 1040.10 and 1040.11.¹⁰

2. Terms and Definitions

The definitions provided here are those applying to pulse code modulation (PCM) and voice frequency analog telephony. These are general definitions, and when greater detail is required it will appear in the applicable section. There may be some situations where a specific definition for one section will differ from that in another. Some definitions may also appear in IEEE Std 100-1988 [10],¹¹ and where these definitions are suitable for this standard, IEEE Std 100-1988 will be cited. It should be noted that IEEE Std 100-1988 provides definitions for terms used in all fields of electrical and electronics activity, and frequently a definition in one field is not applicable to another.

2.1 Digital Terms and Definitions

alarm, blue* : *See*: alarm indication signal (AIS)

*Deprecated

alarm indication signal (AIS): A signal that replaces the normal traffic signal when a maintenance alarm indication has been activated.

alarm, red: A locally detected failure in a sink device, e.g., a primary multiplex equipment. Examples of locally detected failures could be loss of synchronization, incoming signal failure, a blown fuse, etc.

alarm, yellow: An indication provided to a source device indicating a signal failure condition in a sink device. Also known as a remote alarm indication (RAI).

⁸CCITT publications are distributed in the U. S. through OMNICON, Inc., 115 Park Street SE, Vienna, VA 22180.

⁹This Working Group document is available from the Exchange Carriers Standards Association, 5430 Grosvenor Lane, Bethesda, MD 20814-2130.

¹⁰US Regulatory Guides are available from the Superintendent of Documents, US Government Printing Office, P.O. Box 37082, Washington, DC 20013-7082, USA.

¹¹The numbers in brackets corresponds to those of the references in 1.2.

alternate mark inversion code (AMI): *See:* bipolar signal

asynchronous: A transmission process such that between any two significant instants in the same group, there is always an integral number of unit intervals. Between two significant instants located in different groups, there is not always an integral number of unit intervals. *Syn:* nonsynchronous.

available state: A state that occurs when all of the following are true:

- a) Bit-error ratio better than one in ten to the n power, for a specific number of consecutive observation periods of fixed duration.
- b) Block-error ratio better than one in ten to the n power, for a specific number of consecutive observation periods of fixed duration.
- c) A specific number of consecutive observation periods of fixed duration without a severely errored unit of time.

NOTE — the consecutive observation periods in 1., 2., and 3. above are different for each case.

bipolar code violation: A violation that occurs whenever two consecutive nonzero elements of the same polarity occur in an AMI signal. A bipolar violation is a code violation if (1) it occurs in an AMI signal, or (2) it occurs in a BnZS signal separate from a zero substitution code. Such a code violation indicates an error in transmission.

bipolar signal: A line code that employs a ternary signal to convey binary digits in which successive binary ones are represented by signal elements that are normally of alternating positive and negative polarity but equal in amplitude, and in which binary zeros are represented by signal elements that have zero amplitude. *Syn:* AMI signal.

bipolar violation (BPV): A nonzero signal element in an alternate mark inversion signal that has the same polarity as the previous nonzero element.

bit error: A bit is said to be in error when it is transferred from the source to the destination within the assigned timeslot, but the delivered bit is of a different value than that sent from the source.

bit-error ratio (BER): The ratio of the number of bit errors to the total number of bits transmitted in a given time interval. BER may be measured directly by detecting errors in a known signal, or approximated from code violations or framing bit errors. Numerical values of error ratio should be expressed in the form $n \cdot 10^{-p}$, where p is an integer greater than zero. When n is omitted, the implied value is 1.

bit rate: The number of bits transmitted per unit of time, usually expressed in bits per second (bps).

block-error ratio: The ratio of the blocks in error received in a specified period to the total number of blocks received in the same period.

block errors: A block is said to be in error when one or more bit errors occur in that block when it is transferred from the source to the destination within the timeslot assigned.

BnZS code.: A bipolar line code with n -zero substitution.

carrier group alarm (CGA): A combination of a carrier failure alarm and trunk conditioning, which is the process of disconnecting any trunk-group connections and making trunks busy.

change-of-frame alignment (COFA): A state that occurs when an off-line framer realigns the receiver to the proper frame alignment signal. *See also:* **out-of-frame (OOF) condition**

channel timeslots: A timeslot that occupies a specified position(s) in a frame and is allocated to a particular time-derived channel.

clock, reference: A clock of very high stability and accuracy that may be completely autonomous and whose frequency serves as a basis of comparison for the frequency of other clocks.

code, idle channel: A repetitive pattern (code) to identify an idle channel. In some situations this code can produce a signal on a channel.

code, peak: The codes that will produce the maximum decoder output level.

code violation: Violation of a coding rule, e.g., the AMI coding rule where the code is corrupted by bipolar violation(s).

codec: A combination of a coder and decoder operating in different directions of transmission in the same equipment.

coder offset: The difference between the code that is supposed to result from a zero-voltage input to the encoder and the code that actually occurs.

controlled slip: *See: slip, controlled*

CRC violation: If the transmitted and received CRC codes are not identical, a CRC violation has occurred, meaning one or more errors has occurred in transmission.

cyclic redundancy (check) code (CRC): Defined for some digital transmission formats (usually stated with the number of bits in the code, e.g., CRC6, CRC9, etc.). The CRC is the result of a calculation carried out on the set of transmitted bits by the transmitter. The CRC is encoded into the transmitted signal with the data. At the receiver, the calculation creating the CRC may be repeated, and the result compared to that encoded in the signal. The calculations are chosen to optimize the error detection capability.

dBDSX: A representation of signal level expressed by a measure of base-to-peak signal voltage at the digital signal crossconnect (DSX) point.

dBm: Decibel with reference to one milliwatt.

decoder: A device that performs decoding.

degraded minute: Sixty seconds of available time that has a BER worse than 10^{-6} and better than 10^{-3} . Degraded minutes are expressed as an integer.

delay, absolute: The time elapsed between transmission of a signal and reception of the same signal.

digital bit rate: The number of bits per unit of time.

digital line link (digital line path): A digital link that comprises a digital line section or a number of tandem-connected digital line sections.

digital line section: A digital section implemented on a single type of manufactured transmission medium, such as symmetric cable pair, coaxial, or fibre.

digital link: The method of digital transmission of a digital signal of specified rate between two digital distribution frames (or equivalent).

NOTES:

- 1 — a digital link comprises one or more digital sections and may include multiplexing or demultiplexing, with the rule that the digital signal exiting the link must not differ in information content from the signal entering the link.
- 2 — The term may be qualified to indicate the transmission medium used, for example, “digital satellite link.”
- 3 — The term always applies to the combination of “forward” and “return” directions of transmission, unless stated otherwise.
- 4 — The term *digital path* is sometimes used to describe one or more digital links connected in tandem, especially between equipment at which the signals of the specified rate originate and terminate.

digital milliwatt: The repetitive transmission of the following sequence of codes in a given channel will be decoded in a receiving terminal as 0 dBm0, 1000.0 Hz signal:

Digit number	1	2	3	4	5	6	7	8
	0	0	0	1	1	1	1	0
	0	0	0	0	1	0	1	1
	0	0	0	0	1	0	1	1
	0	0	0	1	1	1	1	0
	1	0	0	1	1	1	1	0
	1	0	0	0	1	0	1	1
	1	0	0	0	1	0	1	1
	1	0	0	1	1	1	1	0

digital path: The whole of the means of transmitting and receiving a digital signal of specified rate between those two digital signal crossconnect frames (DSX) (or equivalent) at which terminal equipment or switches will be connected. A digital path comprises one or more sections. *See also:* **digital link**

digital reference signal (DRS): A sequence of bits that represents a 1002 Hz to a 1020 Hz signal that, when decoded, will produce a sinusoidal signal of the same power as the digital milliwatt.

digital section: The whole of the means of digital transmission of a digital signal of specified rate between two consecutive digital signal crossconnect (DSX) frames or equivalent.

NOTES:

- 1 — A digital section forms either a part or the whole of a digital link, and includes terminating equipment at both ends, but excludes multiplexers.
- 2 — Where appropriate, the digital rate or multiplex order should qualify the title.
- 3 — The definition applies to the combination of “forward” and “return” directions of transmission, unless stated otherwise.

digital signal crossconnect frame (DSX): A device for circuit rearrangement, patching, and testing purposes. The digital signal crossconnect is designated DSX n , where n indicates the hierarchical level of the digital signal network (DSN) interconnected at that crossconnect.

encoder: A device that performs encoding.

encoding: A means of producing a unique combination of bits (a code) in response to an analog input signal.

encoding law: An algorithm for encoding, i.e., “ μ -law” or “A law.”

error, bit: *See:* **bit error**

error-free second (EFS): A one-second interval during which no error occurs.

error, logical: An error in the binary content (payload) of a digital signal, e.g., bit error.

errored second (ES): A one-second interval during which one or more errors are received.

errored second, asynchronous: A one-second interval during which one or more errors are received, which is measured by detecting errors within seconds defined by a clock that is independent of the error occurrence.

errored second, severely (SES): A second during which the error ratio is worse than a specified limit, or an OOF event occurs.

errored second, synchronous: A one-second interval during which one or more errors are received, which is measured by triggering the one-second time period on a detected error.

extended superframe format (ESF): A structure consisting of 24 DS1 frames. The frame overhead bit positions are shared between an extended superframe frame alignment signal, a cyclic redundancy check (CRC), and a data link.

f-bits: frame bits. *See:* **frame alignment signal**

frame: A cyclic set of consecutive timeslots in which the relative position of each timeslot can be identified.

frame alignment: The state in which the frame of the receiving equipment is synchronized with respect to that of the received signal.

frame alignment signal (FAS): The distinctive signal(s) inserted in every frame or once in n frames, always occupying the same relative position(s) within the frame, and used to establish and maintain frame alignment.

frame, DS1: The DS1 frame consists of 193 bit positions, the first of which is the frame overhead bit position. The remaining 192 bits are available as payload, and can be divided into 24 blocks (channels) of 8 bits each.

framing bit errors: Frame bits that are in error.

idle channel code: *See:* code, idle channel

jitter, maximum output: The peak-to-peak jitter acceptable to enable satisfactory interconnection of digital networks and equipment.

jitter, timing: Short-term deviations of the significant instants of a digital signal from their ideal positions in time.

jitter tolerance, input: The maximum level of input jitter, specified in terms of unit intervals peak to peak, that does not result in an onset of errors.

jitter transfer function: The ratio between input jitter and output jitter in specified frequency bands throughout the applicable jitter mask. The jitter transfer is controlled by the gain and cutoff frequency of the jitter transfer characteristic.

line code: An encoding technique used in digital networks for transmission on digital facilities.

load capacity (peak codes): *See:code, peak*

logic errors: *See:error, logical*

loop timing: Transmit timing derived from the received signal.

loop timing transfer function: The difference between the bit rates of the incoming and outgoing signals. The difference should be reported in parts per million.

loss of frame alignment (LOFA): Frame alignment will be considered lost when a significant ratio of frame alignment signal bits are received in error. Syn: loss of synchronization.

loss of signal (LOS): A loss of signal shall be declared when n consecutive zeros are detected on an incoming signal.

multiframe: A cyclic set of consecutive frames in which the relative position of each frame can be identified.

multiframe alignment: *See:frame alignment*

multiframe alignment signal: *See:frame alignment signal*

multiplex equipment, asynchronous: A transmission interconnection device that inter-leaves nonsynchronous low bit-rate digital signals to form a single high bit-rate digital signal. It also performs the reverse function of dividing a high bit-rate digital signal into multiple nonsynchronous low bit-rate signals. The two processes are referred to in this document as multiplexing (combining signals) and demultiplexing (separating signals). Similarly, the mechanisms used to perform these functions are referred to as multiplex equipment.

multiplex equipment, digital: The equipment for combining digital signals from one digital level to a higher digital level.

multiplex equipment, primary: The equipment for combining analog (vf) signals, or digital data signals, to a primary rate digital signal and vice versa.

nonsynchronous: *See:asynchronous*

North American digital hierarchy: See Fig 2(a). An ascending series of digital rates (1.544 Mb/s—DS1, 3.152 Mb/s—DS1C, 6.312 Mb/s—DS2, and 44.736 Mb/s—DS3) used for PCM transmission in North America.

out-of-frame condition (OOF): A state that occurs when the receive frame alignment is not consistent with the transmit system frame alignment (for the same direction of transmission). *See also:change-of-frame alignment (COFA)*

overhead bits: Bits assigned at the source that remain with the information payload until the payload reaches the sink. The overhead bits are used for functions associated with transporting the payload.

parity violation: If the received parity and the parity calculation on the data do not agree, a parity violation has occurred.

payload: The information bits (within a frame).

power level at DSX: Power in dBm of an all-ones signal measured at a digital signal crossconnect (DSX). See Tables 18 through 19. *See also:dBDSX*

pulse density violation (PDV): A violation that occurs if a signal contains more than a specified number of zeros, or the density of ones is less than specified.

pseudo-random test signal: A signal consisting of a bit sequence that approximates a random signal.

quasi-random signal (QRS): A pseudo-random test signal that has artificial constraints to limit the maximum number of zeros in the bit sequence.

recovery-of-frame alignment: The establishment of proper frame alignment of the receiver after an out-of-frame condition.

regeneration: The process of receiving and reconstructing a digital signal so that the amplitudes, waveforms, and timing of its signal elements are re-established within specified limits.

robbed bit signaling: A scheme in which the signaling bits for each channel are assigned to the least significant bit (bit 8) of frames 6 and 12 of superframe format, or frames 6, 12, 18, and 24 of extended superframe format. When a frame is used for signaling, bits 1-7 are used for channel transmission.

severely errored seconds (SES): *See: errored second, severely*

slip, controlled: The occurrence of a replication or deletion of all the information bits in a frame at the receiving terminal.

slip, uncontrolled: The loss or gain of one or more digit positions or a set of consecutive digit positions in a digital signal that is not a controlled slip of the timing processes associated with transmission or switching of the digital signal, and in which either the magnitude or the instant of that loss or gain is not controlled.

source/sink device: Source devices originate, whereas sink devices terminate signals. Examples of source/sink devices include channel banks, and digital crossconnect systems.

stuffing bits: Bits inserted into a frame to compensate for timing differences in constituent lower rate signals.

superframe (SF): A structure that consists of 12 DS1 frames (2316 bits). The DS1 frame comprises 193 bit positions, the first of which is the frame overhead bit position. Frame overhead bit positions are used for terminal frame (F_t) and signaling frame (F_s) alignment only.

synchronous: A mode of transmission in which the sending and receiving terminal equipment are operating continuously at the same rate and are maintained in a desired phase relationship by an appropriate means.

timeslot: Any cyclic time interval that can be recognized and defined uniquely.

timing jitter: *See: jitter, timing*

trunk conditioning: *See: carrier group alarm*

unavailable seconds (UAS): The time interval in seconds, starting with the first of ten or more consecutive SES and ending at the beginning of ten consecutive non-SES. *See: unavailability*

unavailability: A state of nonservice that occurs when one or more of the following is true:

- 1) A bit-error ratio worse than one in ten to the n power for a specific number of consecutive observation periods of fixed duration has occurred.
- 2) A block-error ratio worse than one in ten to the n power, for a specific number of consecutive observation periods of fixed duration, has occurred.
- 3) More than a specific number of consecutive severely errored units of time has occurred.
- 4) A LOS event is detected.

unit interval: The nominal difference in time between consecutive significant instants of an isochronous signal.

wander: The long-term variations of the significant instants of a digital signal from their ideal position in time. Long term implies that these variations are of low frequency, less than 10 Hz.

yellow alarm: *See: alarm, yellow*

zero-byte timeslot interchange (ZBTSI): A method of coding in which a variable address code is exchanged for any zero octet. The address information describes where, in the serial bitstream, zero octets originally occurred. It is a five-step process where data enters a buffer, zero octets are identified and removed, the nonzero bytes move to fill in the gaps, the first gap is identified, and a transparent flag bit is set in front of the message to indicate that one or more bytes originally contained zeros.

2.2 Voice Frequency Analog Terms and Definitions

absolute delay: The interval of time between the transmission of a signal and the reception of the same signal at a different point in the circuit.

attenuation distortion (frequency response): The relative attenuation at any frequency with respect to that at a reference frequency.

crosstalk: Undesired energy appearing in one path as a result of coupling from another path. Crosstalk is further classified as near-end, far-end, intelligible, or unintelligible.

distortion, single frequency: The production of frequency components other than the applied single frequency; these components may not necessarily be harmonic components.

distortion, total: The ratio in dB of the energy of a fundamental test signal, to all the other energy appearing in the band of interest.

distortion, total harmonic: The ratio in dB of the energy of a fundamental test frequency to the energy of its harmonics appearing in the band of interest.

echo return loss: *See: return loss, echo*

envelope delay: The propagation time between two points for the envelope of a wave.

envelope delay distortion (EDD): The difference between the envelope delay at one frequency and the envelope delay at a reference frequency, which is usually taken as the frequency of minimum envelope delay.

frequency response: *See: attenuation distortion*

gain/loss: The logarithmic ratio (expressed in decibels) of output power to input power.

gain stability: The ability of a device to maintain its gain within specified limits, under specified environmental and power variations.

holding tone: A test tone in the range of 1002 to 1020 Hz, having specific requirements as specified in IEEE Std 743-1984 [13]. The level of the holding tone is specified as part of the test requirement. The tone is used to measure analog circuit impairments.

insertion loss: *See: loss, insertion*

intermodulation distortion: Nonlinear distortion of a system or transducer characterized by the appearance of frequencies at the output equal to the sums and differences of integral multiples of the two or more component frequencies present at the input. Harmonic components also present are not included as part of the intermodulation distortion. For voice frequency telephony in North America, intermodulation requirements and measurements are based on the four-tone method in accordance with IEEE Std 743-1984 [13].

level, power dBm: Decibels relative to one milliwatt. This is the customary unit worldwide for measurement of telecommunications signal power. Zero dBm equals one milliwatt.

level, tracking: Deviation of the gain or loss as the input level to a codec is varied. Usually the reference level is 0 dBm0 at 1004 Hz.

longitudinal balance: The ratio of the disturbing longitudinal voltage V_s and the resulting metallic voltage V_m of the network under test, expressed in decibels.

noise, dBrn: Decibels relative to one picowatt (0 dBrn equals -90 dBm) reference noise level. This is the customary North American unit for measurement of noise power on telecommunication circuits.

noise, dBrnC: Decibels relative to one picowatt reference noise level, measured with C-message filter weighting.

NOTE — there are several other filter weightings besides C-message.

noise, idle channel: Noise present on a channel with the distant end terminated, and no input signal.

noise, impulse: Noise characterized by transient disturbances. Impulses exceeding a specified threshold are counted for a specific duration with an impulse noise counter designed in accordance with IEEE Std 743-1984 [13].

noise with tone, dBrnC-notched: The measure of noise on a channel when a holding tone is present. A very narrow band elimination filter is used with a C-message filter to significantly reduce the level of the holding tone, and pass the noise in the remaining part of the channel.

quantizing noise (distortion): The impairment introduced during the process of digitally encoding and decoding an analog signal.

return loss: A measure of power reflected back to the originating end of a channel due to impedance mismatch. This measurement is also used to characterize equipment impedance accuracy vs. a test impedance.

return loss, echo (ERL): A frequency weighted average, over the middle of the voice band, of the return losses $RL(f)$ at any point in a channel, with the output of the channel terminated with a specified standard impedance. The weighting is given in IEEE Std 743-1984 [13]. The 3 dB bandwidth of the weighting is 560 Hz to 1965 Hz.

return loss, singing (SRL): The minimum of SRL-low (SRL-L) and SRL-high (SRL-H). SRL-low is the frequency weighted average of the return losses in a low-frequency band (with 3 dB bandwidth from 260 Hz to 500 Hz). SRL-high is the frequency weighted average of the return loss in a high-frequency band (with 3 dB bandwidth from 2200 Hz to 3400 Hz). The weightings are given in IEEE Std 743-1984 [13].

tracking: *See: level, tracking*

transmission level point (TLP): For a particular point in a transmission system, the design signal level in dB relative to the level at the zero TLP reference point.

3. General Overview of Standard

The broadest application of this document is the testing of the link, section, and multiplex equipment that comprise a PCM digital network. The methods given apply to end-to-end performance testing of digital links and the subsystems of those links. Emphasis is also given to the testing of multiplex equipment including primary multiplex equipment containing analog-digital conversion devices. This standard is divided into three subject areas: testing configurations, test parameters, and test equipment functions. The coverage of a complete subject is distributed through these three divisions. For instance, someone wants to make a complete set of tests on a DM12 multiplexer, which combines the DS1 rate to DS2. First, he would look at Tables 10, 11, and 12 for a list of possible operational tests on the multiplex equipment. Then he would refer to the parameter section for detailed explanations of the measurement algorithms, objectives, scope, and signal types. Next, he would refer to the test equipment configurations section for descriptions of test arrangements—where to place signal generators, signal receivers, etc. Finally, he would refer to the test equipment functions for information detailing the capabilities of the signal generators and signal analysis equipment. The parameter section describes the mathematics of measurement, the configuration section describes the topology, the function section describes the tools used for measurements.

3.1 Testing Configurations

The two basic areas of concern in this standard are covered in a simple form, illustrated in a topological view in Fig 1, showing how the individual elements of a Digital Link are partitioned as subsystems. Tests on subsystems are specified at the crossconnect terminals DSX n .

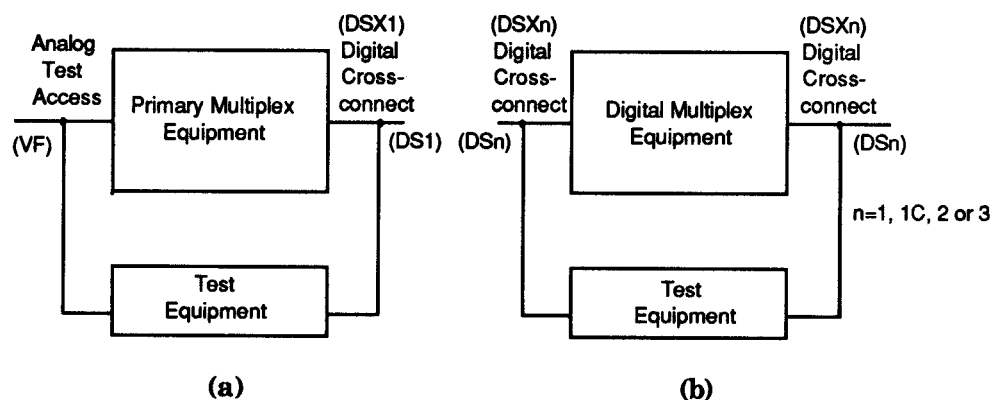


Figure 1 — Digital Link Subsystems

3.1.1 Multiplex Equipment Configurations

Digital multiplex equipment test configurations involve testing between digital test access at both sides [see Fig 1(b)]. Digital test access is usually at digital crossconnect frames while analog access is at either two-wire or four-wire test boards or through switching machine test access ports.

Primary multiplex equipment test configurations involve testing between an analog access at one side and a digital access at the other [see Fig 1(a)]. Drop-and-insert methods cover operational tests where the equipment is still in service.

3.1.2 Link/Section Configuration

Digital links and sections are tested by inserting test equipment at the crossconnects on either side of the link/section. Drop and insert methods are required for operational tests where the equipment is still in service.

3.2 Test Objectives

There are four types of test objectives defined in this standard: Certification, Installation, Pre-service, and Operational. These objectives call for very precise and comprehensive tests at the certification level, and less comprehensive and precise at the operational level. Instruments that conform to this standard will be tailored to the test objectives application (as defined above). The standard will provide the expected range and accuracy requirements of the parameters to be tested; however, the user must specify the actual pass-fail limits for each application.

3.3 Basic Measurements

3.3.1 Analog Measurements

Analog measurement methods needed to measure primary multiplex equipment are given in this standard, but the standards for analog test equipment are given in IEEE Std 743-1984 [13]. The primary multiplex equipment tests measure performance of the A-D, D-A, filters, and terminating networks. In most cases, either the stimulus or response is on the digital side of the multiplex equipment, in the form of an encoded analog signal. The digital data must be analyzed to calculate the equivalent signal values. For example, loss analog to digital: insert analog milliwatt into the primary mux, extract the digital data samples from the timeslot channel under test and numerically process the data through a root-mean-square (rms) algorithm. The dBm0 units are compared to the input power obtaining loss.

3.3.2 Digital Measurements

The digital measurements in this standard are applied to digital paths and sections as well as to multiplex equipment. The measurements could be divided into two basic groups, those that are purely digital and those that are digital interfaces. Digital signals are a special class of analog signals—a voltage with a specified amplitude and rise-fall times occurring within a certain time window is judged to be a one, a zero, or an error. All measurements that are made on the resulting one or zero data are purely digital. Examples include bit-error ratio, error-free seconds, time to red alarm, etc.

Measurements involving decisions as to whether a signal is a one or zero would be interface tests. Examples include pulse shape, jitter, impedance, clock timing, etc. Typically a path or section is tested with a purely digital test for an overall performance evaluation and then interface tests might follow for troubleshooting or verification.

3.4 Test Equipment Functions

The test equipment functions have three main areas of concern: generators, analyzers, and interface circuits. The generators must supply all the test patterns, whether analog or digital. The analyzers must accept analog or digital data as required, and convert received data into the forms required for output display. The interface circuits supply the capability to extract data from serial bitstreams, inject data into bitstreams, provide clocks, extract clocks, etc. All the requirements imposed on a digital crossconnect interface apply to the interface for the test instrument.

4. Standards for Digital Multiplex Equipment Testing and Test Equipment

4.1 Introduction

This section provides test methods and test equipment criteria for testing digital multiplex equipment. This part of the section presents the objectives, scope, and definitions for digital multiplex testing and test equipment. Section 4.2, Test Parameters, details the parameters to be measured for digital multiplex equipment. Section 4.3, Test Equipment Functions, details the test equipment functions needed to test digital multiplex equipment. Section 4.4, Testing Configurations, is to be used in testing digital multiplex equipment. Details are restricted to equipment used in the North American digital hierarchy. The levels in the North American digital network that are covered by this standard include DS1 (1.544 Mb/s), DS1C (3.152 Mb/s), DS2 (6.312 Mb/s), and DS3 (44.736 Mb/s).¹²

NOTE — Some details on equipment for the CCITT hierarchy may be found in [20] and [23].

4.1.1 Objectives

4.1.1.1 General Objectives

The objective of this section is to provide complete information on the test equipment and methods for testing digital multiplex equipment including the following:

- 1) Primary multiplex equipment that converts between analog and 1.544 Mb/s digital signals (DS1) that contain the PCM coded samples.
- 2) Digital multiplex equipment that converts among the 1.544 (DS1), 3.152 (DS1C), 6.312 (DS2), and 44.736 (DS3) Mb/s levels of the digital hierarchy.

¹²Some standards state 1.544 Mb/s as 1544 kb/s and vice versa. Based upon common usage in North America, 1.544 Mb/s is used throughout this document.

4.1.1.2 Test Objectives

Digital multiplex equipment test objectives define the purpose of the tests and select the parameters whose measurements will provide information that will indicate satisfaction with that purpose.

Test objectives are the following:

- 1) *Certification.* A comprehensive set of tests performed, normally by the manufacturer, to assure that the product delivered to the contractor meets the criteria established and provides the performance, reliability, and quality specified. These are comprehensive measurements and require the highest degree of accuracy.
- 2) *Installation.* A set of tests, normally performed by the supplier, installer, or user, designed to show that the certified product has been accommodated to the operational environment and that the equipment installed can be accepted for use in specific applications. These tests are less comprehensive than certification tests.
- 3) *Pre-service.* Tests and adjustments that are usually performed by the user to verify that the product is ready for use in conjunction with other equipment in its operational environment. These tests will be less comprehensive than certification testing.
- 4) *Operational.* Tests performed by the user on a scheduled or trouble basis. These tests are used in the maintenance, trouble isolation, or restoration of either the product or the network to its required operational status. Testing in this environment is different from certification testing. Pass/fail tests should be used for frequently needed tests, and where practical should be on an in-service basis. Occasionally trouble isolation may require sophisticated test equipment with capabilities similar to those required for certification testing.

4.1.2 Scope of Multiplex Section

Standards in this section apply to digital multiplex equipment intended for telephony applications using the North American digital hierarchy [see Fig 2(a)]. This document is limited to levels DS1, DS1C, DS2, and DS3 in the hierarchy [see Fig 2(b)]. It includes primary multiplex equipment that connects voiceband distribution frames and 1.544 Mb/s DS1 crossconnects. It also includes asynchronous digital multiplex equipment connected on both sides to a distribution frame using North American DS1, DS1C, DS2, or DS3 signals. Synchronous multiplex equipment and equipment offering other than 64 kb/s coded voiceband channels (e.g., 32 kb/s voiceband coding, 256 kb/s program, or 64 kb/s data) are not covered in this document. Signaling parameter measurement is outside the scope of this document. Although information on tests among analog and digital ports are included, details for exclusively analog methods and test equipment are given in IEEE Std 743-1984 [13].

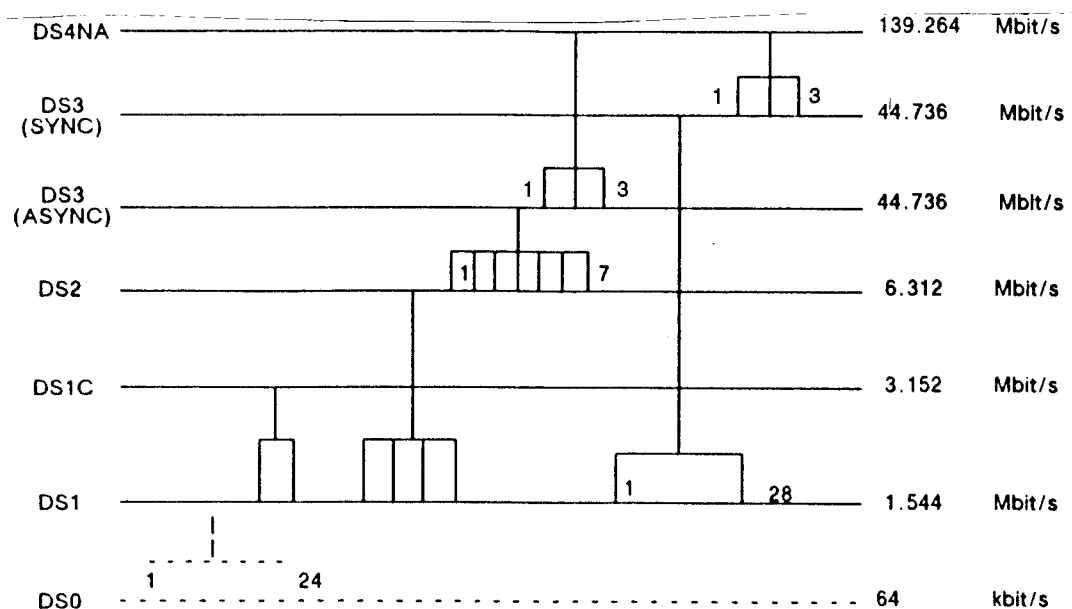
4.2 Test Parameters

This section lists, describes, and gives typical measured values for the test parameters used in measuring digital multiplex equipment.

4.2.1 Performance Parameter Description

The multiplex equipment test parameters are defined with respect to the following:

- 1) Test objectives
- 2) Test scope
- 3) Signal types
- 4) Multiplex types (primary or higher order)



NOTE: The DS0 rate is not explicitly defined as a physical interface in this standard, but it is an important electrical component of the DS1 and higher rate interface formats.

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**Figure 2—Digital System
(a) Hierarchy**

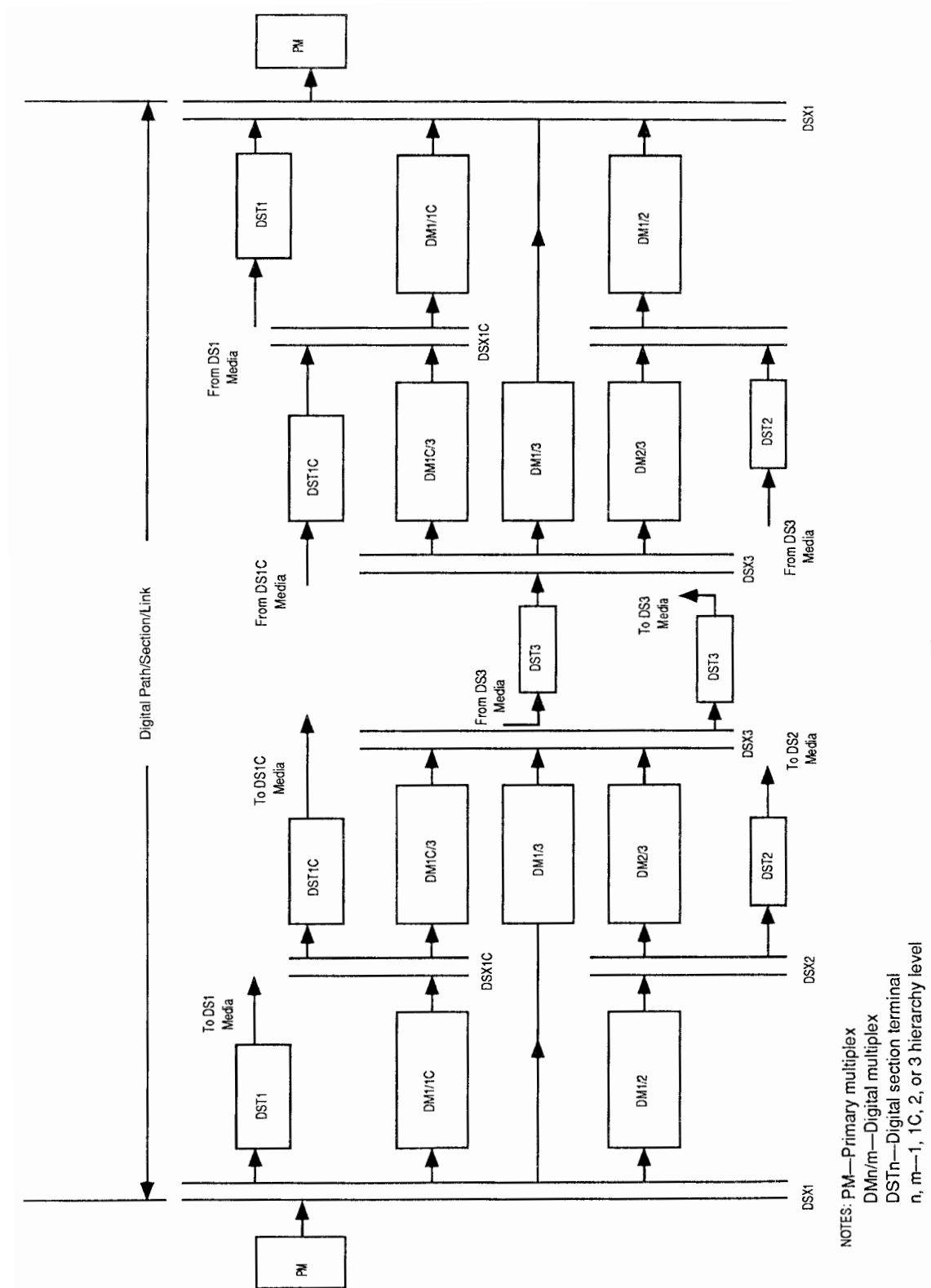


Figure 2 — Digital System
(b) Components

A multiplex equipment test parameter provides information concerning multiplex characteristics in accordance with the test objectives and the test scope. The characteristics are specific to the levels of the digital hierarchy spanned by the multiplex equipment types and are identifiable by the signal transformations provided by the multiplex equipment. The parameters in general fall into specific groups depending on whether they are characteristics of the multiplex equipment considered in isolation or interfacing with other elements in the network.

4.2.1.1 Test Objectives

The test objectives define the purpose of the test and list the parameters whose measurement will provide information that will indicate satisfaction with the purpose. The test objectives are Certification, Installation, Pre-service, and Operational.

4.2.1.2 Test Scope

The scope of the test defines the limits within which the test will be performed. This involves designation of directly measurable or derived parameters, the accuracy with which they are measured, and the range of values through which they are measured.

4.2.1.2.1 Instrument Accuracy

The general guideline for the accuracy of a measurement is that the overall error in the measurement should be less than 10% of the tolerance of the quantity to be measured. For situations where no tolerance is specified, e.g., $S/N \geq 35$ dB, the guideline is that the overall error in the measurement should be less than 10% of the minimum quantity to be measured or of the minimum specified limit, whichever is appropriate.

4.2.1.2.2 Range

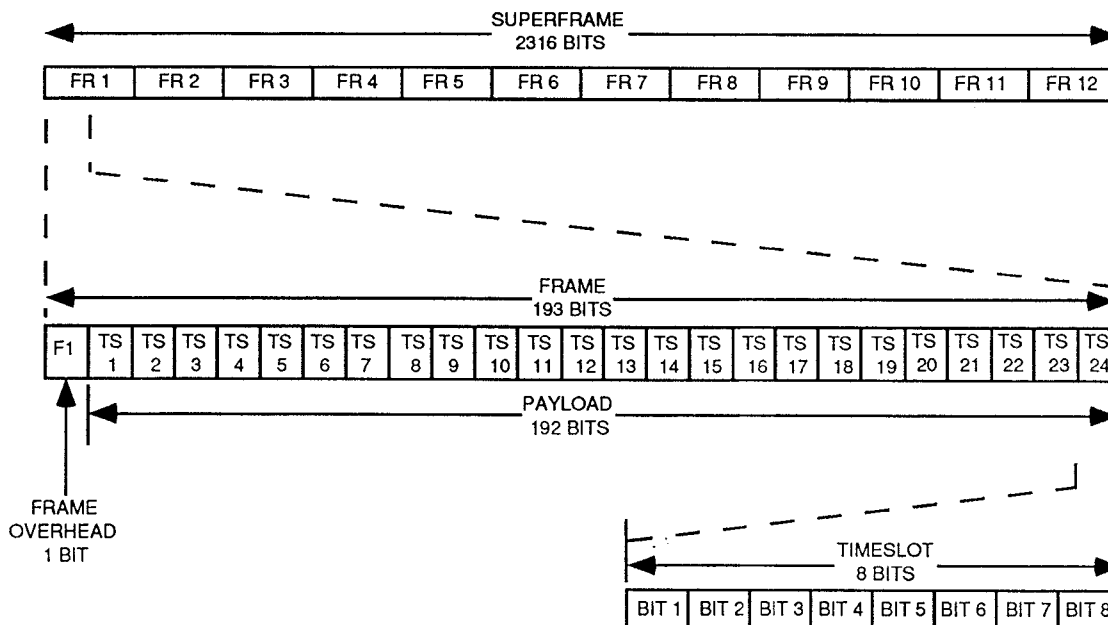
The range of the parameter indicates a significant set of measurements required to define the product characteristics of interest. This section gives the range of variation of each parameter of the test objectives.

4.2.1.3 Signal Types

Digital signals include DS1, DS1C, DS2, and DS3. Descriptions of these signal formats are contained in this section. (See also ANSI T1.102-1987 [2].)

4.2.1.3.1 DS1 Signal Format

- 1) *Frame organization.* The conventional DS1 frame consists of twenty-four 8-bit words and one framing bit for a total of 193 bits per frame (see Fig 3). The nominal bit rate of the outgoing DS1 signal is 1.544 Mb/s, and the sampling rate for a message channel is 8000 Hz. The eighth bit of each PCM word is used for signaling every sixth frame.
 - a) *Superframe format.* As shown in Fig 3 and Table 1, the framing bit is time shared to identify terminal framing (F_T) and signaling framing (F_S). Terminal framing identifies the location of timeslot one, and signaling framing identifies those frames that have the A and B bit signaling. In the SF-type framing format, the framing pattern is generated by interleaving two signals, the terminal frame 101010 pattern and the signaling frame 001110 pattern, into a 12-frame superframe.
 - b) *Extended superframe format.* The extended superframe format (ESF) shown in Fig 4 and Table 2 shows the F-bit assignments of the ESF. Interleaving (time multiplexing) of signals for the extended superframe format occurs in a similar fashion as for the superframe except that three signals are combined into a 24-frame extended superframe. The three signals are as follows:
 - 2 kb/s framing pattern sequence 001011
 - 2 kb/s cyclic redundancy check (CRC6)
 - 4 kb/s facility data link



SUPERFRAME FRAME OVERHEAD BIT SEQUENCE
 TWELVE FRAME OVERHEAD BITS OCCUPY SEQUENTIAL FRAME OVERHEAD BIT POSITIONS AS FOLLOWS:

F1, S1,
 F2, S2,
 F3, S3,
 F4, S4,
 F5, S5,
 F6, S6

Channel number/timeslot assignments are included in Table 3.

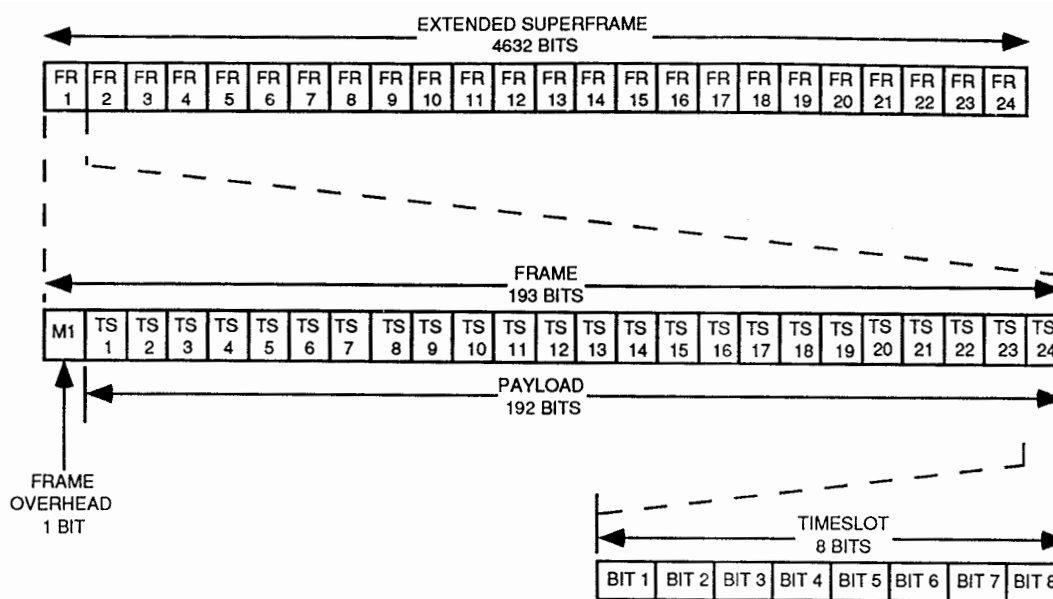
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Figure 3—Superframe Structure

Table 1—Assignment of Bits in DS1 Superframe (SF) Signals

Frame Number	F-bit		Information		Signaling Channel
	Terminal Framing F_t	Signaling Framing F_s	Coding Bits	Signaling Bits	
1	1	—	1–8	—	
2	—	0	1–8	—	
3	0	—	1–8	—	
4	—	0	1–8	—	
5	1	—	1–8	—	
6	—	1	1–7	8	A
7	0	—	1–8	—	
8	—	1	1–8	—	
9	1	—	1–8	—	
10	—	1	1–8	—	
11	0	—	1–8	—	
12	—	0	1–7	8	B

NOTE — The sequence shown above is respective; 6 and 12 above are denoted signaling frames.



EXTENDED SUPERFRAME FRAME OVERHEAD BIT SEQUENCE
 TWENTY-FOUR FRAME OVERHEAD BITS OCCUPY SEQUENTIAL FRAME OVERHEAD BIT POSITIONS AS FOLLOWS:

M1, C1, M2, F1
 M3, C2, M4, F2
 M5, C3, M6, F3
 M7, C4, M8, F4
 M9, C5, M10, F5
 M11, C8, M12, F6

Channel number/timeslot assignments are included in Table 3.

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Figure 4—Extended Superframe Structure

Table 2—Assignment of Bits in DSI Extended Superframe (ESF) Signals

ESF		F-bits			Bit Use in Each Channel Timeslot		Signaling Bit-Use Option			
Frame No.	Bit No.	FPS	FDL	CRC	Traffic	Signaling	T	2	4	16
1	0	—	m	—	Bits 1–8					
2	193	—	—	C1	Bits 1–8					
3	386	—	m	—	Bits 1–8					
4	579	0	—	—	Bits 1–8					
5	772	—	m	—	Bits 1–8					
6	965	—	—	C2	Bits 1–7	Bit 8	—	A	A	A
7	1158	—	m	—	Bits 1–8					
8	1351	0	—	—	Bits 1–8					
9	1544	—	m	—	Bits 1–8					
10	1737	—	—	C3	Bits 1–8					
11	1930	—	m	—	Bits 1–8					
12	2123	1	—	—	Bits 1–7	Bit 8	—	A	B	B
13	2316	—	m	—	Bits 1–8					
14	2509	—	—	C4	Bits 1–8					
15	2702	—	m	—	Bits 1–8					
16	2895	0	—	—	Bits 1–8					
17	3088	—	m	—	Bits 1–8					
18	3281	—	—	C5	Bits 1–7	Bit 8	—	A	A	C
19	3474	—	m	—	Bits 1–8					
20	3667	1	—	—	Bits 1–8					
21	3860	—	m	—	Bits 1–8					
22	4053	—	—	C6	Bits 1–8					
23	4246	—	m	—	Bits 1–8					
24	4439	1	—	—	Bits 1–7	Bit 8	—	A	B	D

FPS —Framing pattern sequence (...001011...)

FDL—4kb/s facility data link (Message bits m)

CRC—CRC-6 cyclic redundancy CHECK (bits C1-C6)

Option T—Transparent (bit 8 for traffic)

Option 2—Two-state signaling (channel A)

Option 4—Four-state signaling (channels A and B)

Option 16—Sixteen-state signaling (channels A, B, C, and D)

- 2) *Channel Numbering Identification.* There are three existing channel numbering sequences in use (see Table 3). Sequence one, the sequential numbering scheme, is preferred. Means for accommodating pseudo-random or interleaved channel sequencing are desirable.

4.2.1.3.2 DS1C Mode 2 Signal Format

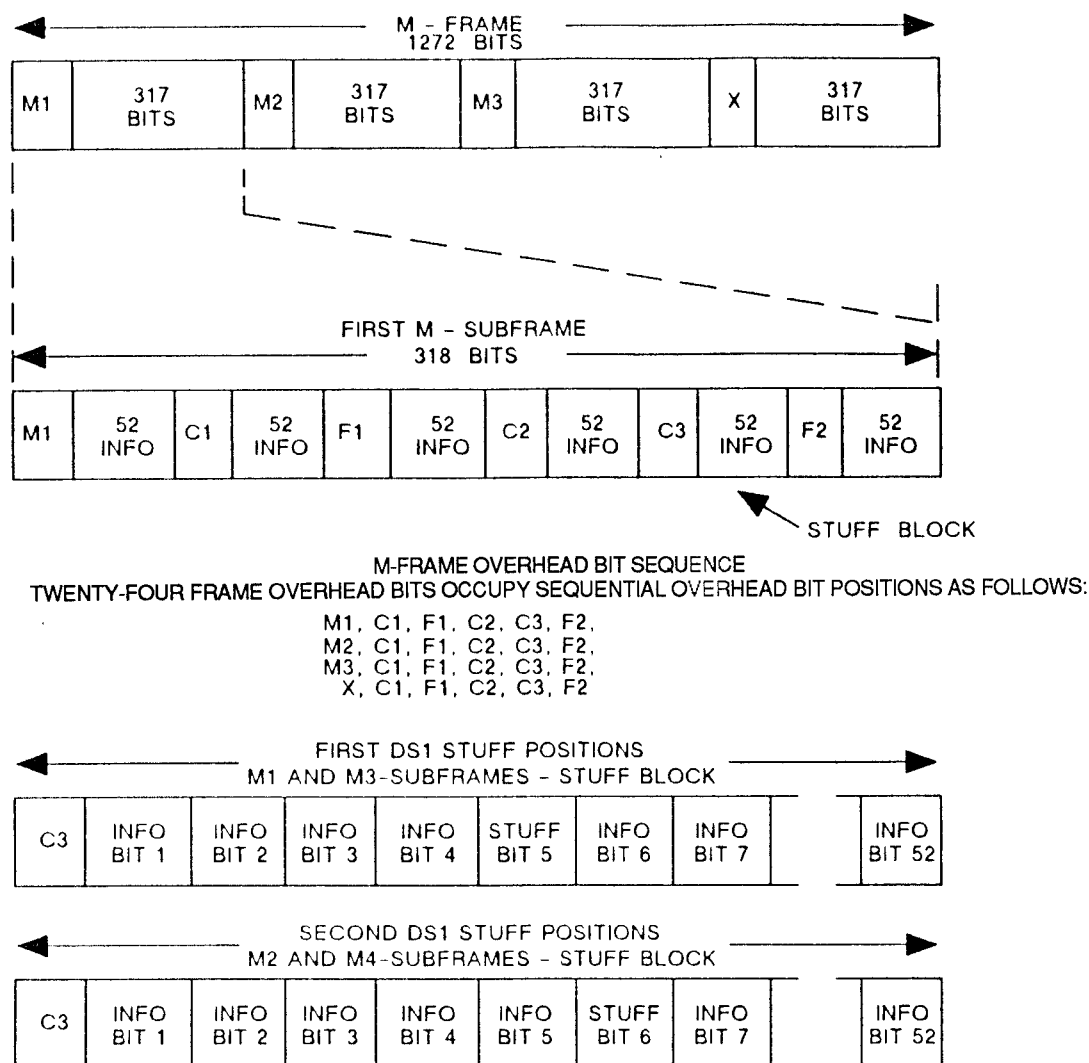
The DS1C Mode 2 signal format is shown in Fig 5.

4.2.1.3.3 DS2 Signal Format

The DS2 signal format is shown in Fig 6.

Table 3—Channel Number/Timeslot Assignments

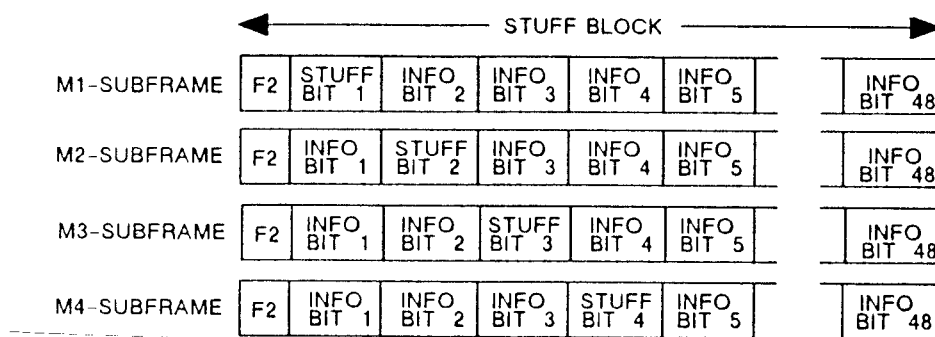
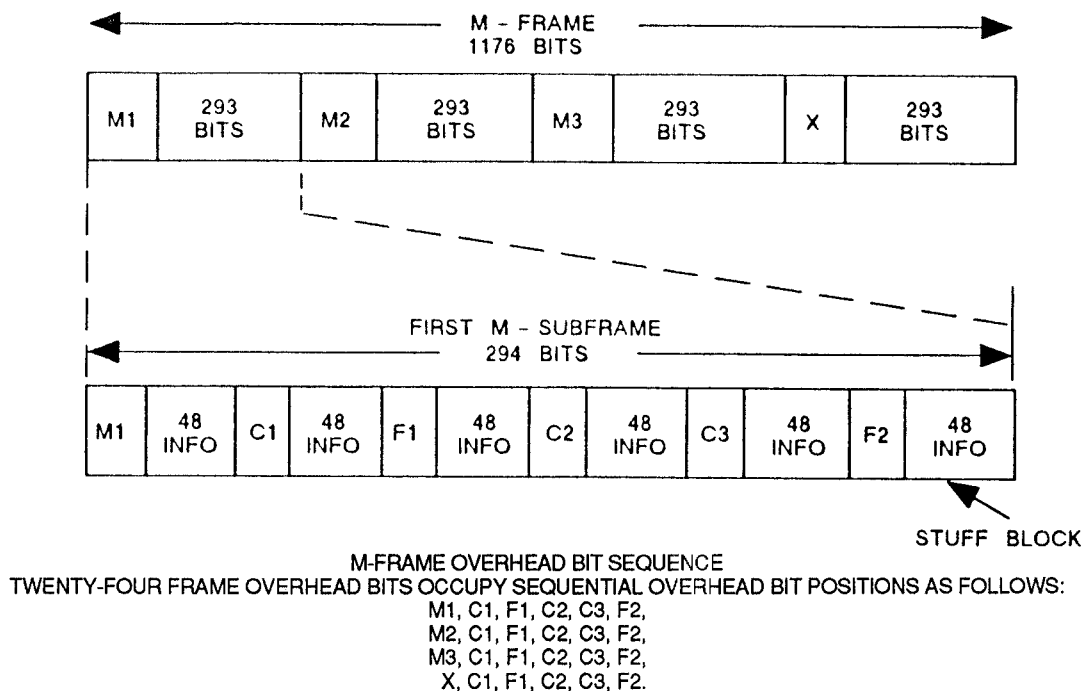
Timeslots	Basic D-Bank Sequential Channel Assignments (Preferred)	D-Bank Channel Sequencing for Operating with Interleaved Channel Number Assignments	D-Bank Channel Sequencing for Operating with Pseudo- random Channel Number Assignments
1	1	1	12
2	2	13	13
3	3	2	1
4	4	14	17
5	5	3	5
6	6	15	21
7	7	4	9
8	8	16	15
9	9	5	3
10	10	17	19
11	11	6	7
12	12	18	23
13	13	7	11
14	14	19	14
15	15	8	2
16	16	20	18
17	17	9	6
18	18	21	22
19	19	10	10
20	20	22	16
21	21	11	4
22	22	23	20
23	23	12	8
24	24	24	24



- NOTES:
- (1) The M-frame alignment signal is M1 = 0, M2 = 1, and M3 = 1.
 - (2) The M-subframe alignment signal is F1 = 0 and F2 = 1.
 - (3) The X-bit is an alarm service digit.
 - (4) The C-bits are the M-subframe stuff indicators.
C1 = C2 = C3 = 1 indicates stuffing was done.
C1 = C2 = C3 = 0 indicates no stuffing.
 - (5) The first DS1 is stuffed in M-subframes 1 and 3.
 - (6) The second DS1 is stuffed in M-subframes 2 and 4.

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Figure 5—DS1C Frame Structure



- NOTES: (1) The M-frame alignment signal is M1 = 0, M2 = 1, and M3 = 1.
 (2) The M-subframe alignment signal is F1 = 0 and F2 = 1.
 (3) The C1, C2, and C3 bit positions are available for application-specific use.
 (4) In the M12 multiplex application, stuffing for 1th DS1 channel occurs in M-subframe 1, in the 1th information bit of the last block. The C-bits of that M-subframe are set to C1 = C2 = C3 = 1 if stuffing occurs, or C1 = C2 = C3 = 0 if no stuffing occurs.

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Figure 6—DS2 Frame Structure

4.2.1.3.4 DS3 Signal Format

The DS3 signal format described here and shown in Fig 7 is required to provide compatibility of present generation transmission facilities and higher-order multiplex equipment between carriers. Future digital network interworking may require standardization of other signal formats, possibly at the expense of established supervisory routines for present facilities.

The DS3 signal is partitioned into M-frames of 4760 bits each. The M-frames are divided into seven M-subframes, each having 680 bits. Each subframe is further divided into eight blocks of 85 bits with 84 of the 85 bits available for information and one used for control.

The frame alignment signal (F bits 1001...), shown in Fig 7, is used to identify all control bit positions. The multiframe alignment signal (M-bits 010...) is used to locate all seven subframes.

X-bits occupy the bit positions at the beginning of the first and second subframes. In any one M-frame, the two X-bits must be identical, i.e., either 11 or 00. The X-bits may be used by a DS3 source for asynchronous low-speed signaling. If so, the source shall not change the state of the X-bits more than once every second, since transmission equipment may also use the X-bits but at higher rates. If not used, the X-bits shall be set to 11.

The bit position at the beginning of the third and fourth subframes is a P-bit that contains parity information. DS3 sources must count parity over the 4704 information bits following the first X-bit in an M-frame and insert the resulting parity information in the P-bit positions of the following frame. PP=11 if the digital sum of all information bits is 1, and PP=00 if the digital sum of all information bits is 0.

The remaining control bit positions (C-bits on Fig 7) are reserved for use by DS3 sources. Any errors made in these bit positions will not be detected by checking parity. In multiplex formats these bits are labeled C with subscripts i and j, where i is the subframe and j gives the relative position of the C-bit in the subframe.

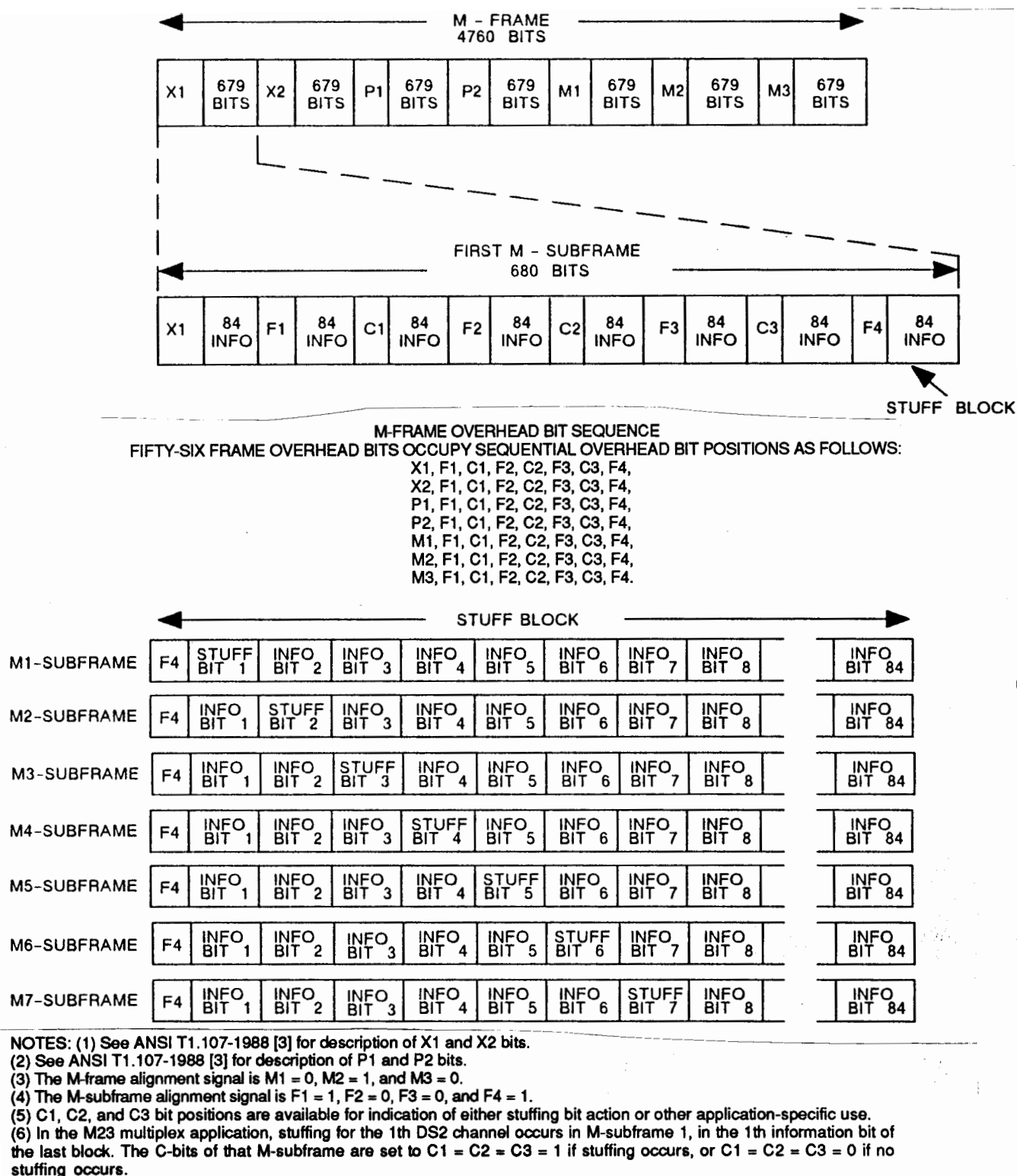


Figure 7—DS3 Frame Structure

4.2.1.4 Multiplex Equipment Type

Parameters to be measured for multiplex equipment characteristics depend on the digital signal levels they transform in the digital hierarchy. These characteristics may be classified as belonging to primary multiplex equipment (digital channel banks), or higher-order digital multiplex equipment. Higher-order multiplex equipment includes DM1-1C, which interleaves two DS1s into a DS1C; DM1-2, which interleaves four DS1s into a DS2; DM1-3, which interleaves 28 DS1s into a DS3; DM1C-3, which interleaves 14 DS1C into a DS3; and DM2-3, which interleaves seven DS2s into a DS3 [see Fig 2(b)].

4.2.2 Primary Multiplex Equipment Voiceband Parameters

This section contains description, accuracy, and range information for primary multiplex equipment voiceband performance parameters (see IEEE Std 743-1984 [13] and [14]).

4.2.2.1 Level at Input and Output

This section treats 2- and 4-wire levels and office cable compensation (see Appendix A).

4.2.2.1.1 Four-Wire Analog Ports

The nominal 4-wire levels, when measured at an external access point of primary multiplex equipment, should be -16 dB transmission level point (TLP) transmit and +7 dB TLP receive. Actual input level (ranging from -16 to -17.5 dB) is measured by applying an analog signal to the input and verifying that the digital output is equivalent in power to the digital milliwatt. In out-of-service tests, actual output level (ranging from +7 to +8.5 dB) is measured by applying a digital reference signal or a digital milliwatt and observing the analog output power. Most level measurements are made with an analog impedance of 600 Ω .

In special cases, 4-wire input levels in the range from -17.5 dB to +7.0 dB TLP and output levels in the range -16 to +8.5 dB TLP may occur; also 150 and 1200 Ω impedance may be encountered. Other special cases may involve even greater variation in level and impedance.

4.2.2.1.2 Two-Wire Analog Ports

Input level is measured by applying a 0 dBm0 analog signal to the input and verifying that the digital output is equivalent in power to the digital milliwatt. Output level is measured by applying a digital reference signal or a digital milliwatt and observing the analog output power. Level measurements are to be made with an analog impedance of 600 Ω or 900 Ω , as appropriate. The 2-wire levels at the primary multiplex input should be at least -13 to 0 dB TLP; at the primary multiplex output they should be at least 0 to -6 dB TLP. Other special cases may involve even greater variation in level and impedance.

4.2.2.1.3 Office Cabling Compensation

Loss compensation is usually provided for a range of office equipment and cabling loss variation of at least from 0 to 1.5 dB, monotonically adjustable in 0.1 dB steps (i.e., as the adjusted loss value increases in 0.1 dB steps, the actual loss must always increase). Capacitance compensation is usually provided for a range from 0 to at least 0.126 μ F monotonically adjustable in nominal 0.002 μ F steps. The office cabling compensation should be considered in determining analog port levels.

4.2.2.2 Gain

Gain is the ratio of output to input power, usually expressed in decibels; measurements are usually in volts, amperes, or milliwatts.

There are three aspects of gain to be considered:

- 1) The coding and compression law ($\mu = 255 - \text{law}$)

- 2) The variation of the ratio of the decoded signal to the signal prior to encoding (variation of gain with input level)
- 3) The variation of gain with input frequency (attenuation distortion)

When input analog signals are converted into digital signals, which are then converted back to analog signals, the decoded signals include quantizing noise, which degrades the signal quality. To control this degradation and for the more efficient use of the capacity of the PCM system, a nonlinear companding encoder is used. The encoder characteristics reduce the quantizing noise by quantizing finely at small signal amplitudes and grossly at large amplitudes. This is a nonlinear process that involves compression (lawful variation of the gain) of the signal and quantization within the encoder and a corresponding expansion (lawful inverse variation of the gain) at the decoder. Although the compression and expansion characteristics are nonlinear, the ratio of the received to the transmitted signal should display a linear transfer characteristic. If compressor characteristics at the transmit side and expander characteristics at the receive side do not exactly compensate for one another, a nonlinear transfer characteristic results.

4.2.2.2.1 CODEC Transfer Characteristic

The relationship between encoder decision levels and decoder output levels is defined by specifying the coder/decoder (codec) transfer characteristic.

Parameter Characteristics. The compression law should be a piecewise linear approximation to a $\mu=255$ law that is given by the following:

$$\frac{|y|}{\mu} = \frac{\ln(1 + \mu \cdot |x|)}{\ln(1 + \mu)}, \mu=255,$$

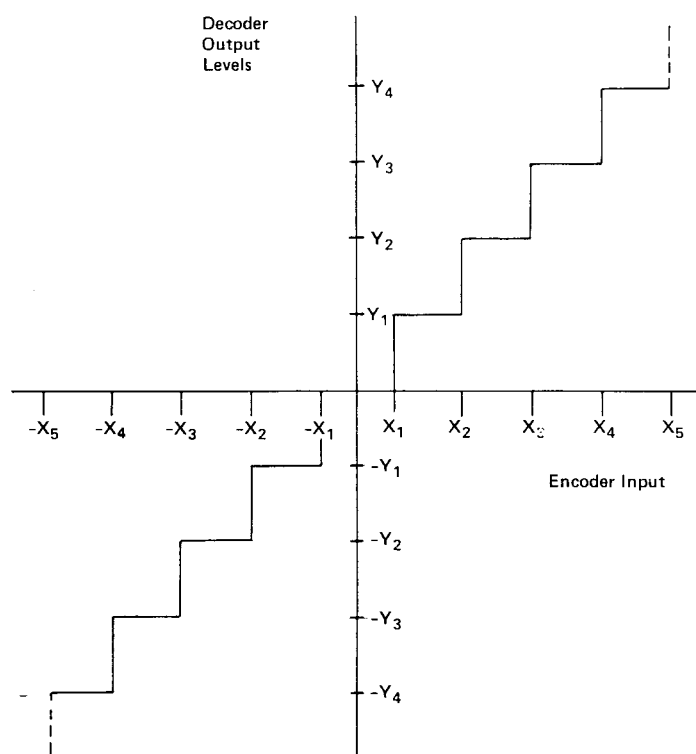
where x and y are respectively the compressor input and output, both normalized to unity. As an option to minimize distortion, the transfer characteristic for the abbreviated 7-bit words (used during signaling frames) should differ from that for the normal 8-bit PCM word. Figure 8 shows the transfer characteristic for normal 8-bit PCM words. The output levels (Yn 's) are always midway between the decision levels (Xn 's) (this can be termed a "decision level assignment" codec). The output levels are given by the following:

$$Y(O) = X(O)$$

$$Y(n) = \frac{X(n) + X(n+1)}{2}, n = 1, 2, \dots, 127.$$

Figure 9 shows the optional transfer characteristic for abbreviated 7-bit PCM words in the signaling frame. Except for the level closest to zero, the output levels ($Z(n)$'s) are also midway between the decision level ($X(n)$'s). These output levels are given by the following:

$$Z(n) = X(2n-1), n = 1, 2, \dots, 64.$$

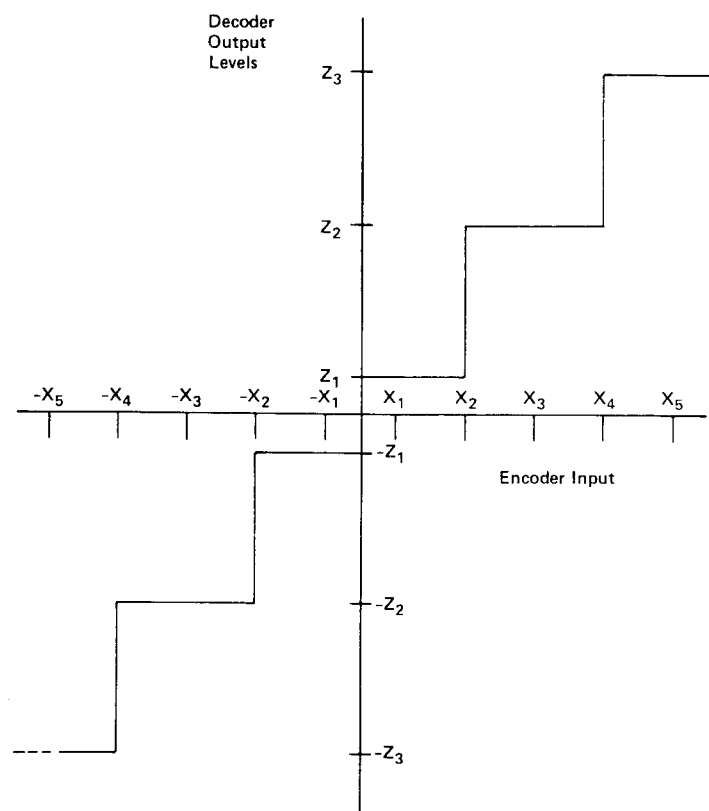


$$Y_0 = X_0 = 0$$

$$Y_n = \frac{X_n + X_{n+1}}{2}; n = 1, 2, 3, \dots, 127$$

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Figure 8—CODEC Transfer Characteristics for Eight-Bit Words



$$Z_n = X_{2n-1}; n = 1, 2, 3, \dots, 64$$

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Figure 9—CODEC Transfer Characteristics for Seven-Bit (Signaling) Frame

Decision Levels: Table 4 lists the relative magnitudes of the decision levels. The decision levels are placed symmetrically about zero. The maximum decision level magnitude in Table 4 is normalized to 8159 so that all magnitudes may be represented by integer values. The magnitude of the peak value of a +3 dBm0, single-frequency signal is different from that of the largest decision level. This results from the definition of the digital milliwatt as 0 dBm0.

Code Assignments: The assignments of PCM code words and decoder output levels are given in Table 5 in terms of coding ranges (decision levels). The code assignment provides a sign bit and ones-complement magnitude coding, resulting in a high ones-density in the output data stream for encoding of speech. The W-bits defined in Table 5 (PCM words column) are required to guarantee that the output data stream never includes more than 15 consecutive zeros. This is related to T1 line clock recovery.

Table 4—Encoder Decision Levels

Level Number n	Level Magnitude* $ X_n $
0	0
$1 \leq n \leq 16$	$2n-1$
$17 \leq n \leq 32$	$4n-33$
$33 \leq n \leq 48$	$8n-161$
$65 \leq n \leq 80$	$32n-1569$
$81 \leq n \leq 96$	$64n-4129$
$97 \leq n \leq 112$	$128n-10\,273$
$113 \leq n \leq 28$	$256n-24\,609$

*Normalized to lie in the range of 0 to 8159 units.

4.2.2.2.2 Variation of Gain with Input Level (“tracking”)

Variation of gain with input level is the deviation in gain of the output from the gain of a 1 kHz input signal at 0 dBm0 when the input level is varied over an appropriate range; this is also called tracking.

With inputs from –37 dBm0 to +3 dBm0, the maximum deviation should be 0.5 dB. With inputs from –50 dBm0 to –37 dBm0, the maximum deviation should be 1 Db.

4.2.2.3 Gain Stability

Gain stability is the deviation with time of the gain (or loss) at 1 kHz, of a 0 dBm0 level signal.

The gain (or loss) should be measured at a frequency between 1002 and 1020 Hz, 0.0 dBm0, from channel bank VF input to VF output. This loss should be the reference loss with respect to which deviations are taken. Measurements are to be made with analog impedances of 600 or 900 Ω , as appropriate. When a sine wave at a nominal 1 kHz and a level of 0 dBm0 is applied to any voice-frequency input, the gain deviation measured at the corresponding voice-frequency output should not vary by more than ± 0.2 dB during any 10 min interval of typical operation, nor by more than ± 0.5 dB during any one year under the permitted variations in the power supply voltage and temperature.

4.2.2.4 Attenuation/Frequency Distortion (“Frequency Response”)

Attenuation (or amplitude) frequency distortion occurs when an analog signal is attenuated differently as a function of frequency over the channel band of interest.

The separate receive and transmit section attenuation versus frequency boundaries for 4-wire and 2-wire channels are shown in Figs 10 and 11, respectively. The overall requirement is the sum of the transmit and receive allocations. The loss measurements are made relative to a nominal 1 kHz and with 0 dBm0 input.

NOTE — Some standards specify –10 dBm0 input.

Table 5—Assignment of Transmitted Codes and Decoded Levels

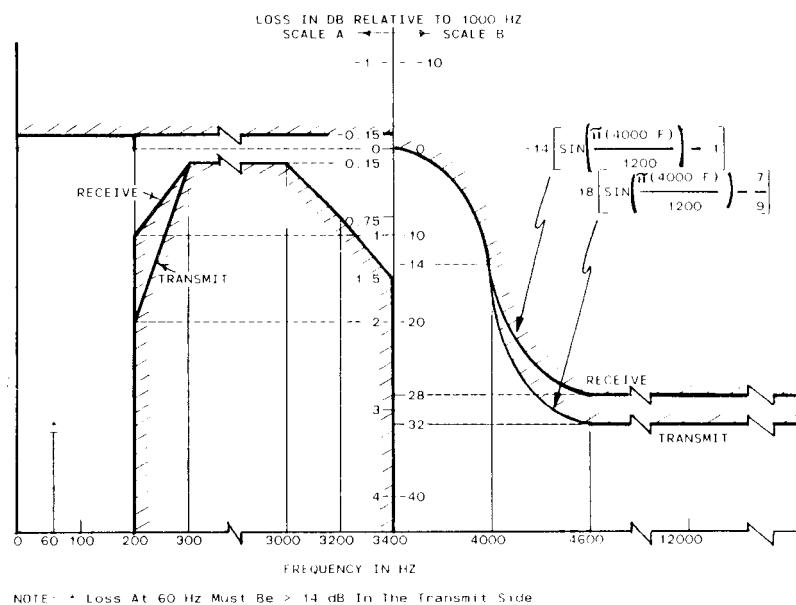
CODING RANGES	PCM WORDS** 1 2 3 4 5 6 7 8	DECODER LEVELS	
		INFORMATION FRAME	SIGNALING FRAME
X_{127}, X_{128}	1 0 0 0 0 0 0 0	Y_{127}	Z_{64}
X_{126}, X_{127}	1 0 0 0 0 0 0 1	Y_{126}	
...	...*	...	...
X_{n+1}, X_{n+2}	$(255-(n-1)) \text{Modulo } 2$	Y_{n+1}	$Z_{(n+2)/2}$
X_n, X_{n+1}	$(255-n) \text{Modulo } 2$	Y_n	
...	...*	...	...
X_3, X_4	1 1 1 1 1 1 0 0	Y_3	Z_2
X_2, X_3	1 1 1 1 1 1 0 1	Y_2	
X_1, X_2	1 1 1 1 1 1 1 0	Y_1	
X_0, X_1	1 1 1 1 1 1 1 1	Y_0	Z_1
$X_0, -X_1$	0 1 1 1 1 1 1 1	Y_0	$-Z_1$
$-X_1, -X_2$	0 1 1 1 1 1 1 0	$-Y_1$	
$-X_2, -X_3$	0 1 1 1 1 1 0 1	$-Y_2$	
$-X_3, -X_4$	0 1 1 1 1 1 0 0	$-Y_3$	$-Z_2$
...	...*	...	...
$-X_n, -X_{n+1}$	$(127-n) \text{Modulo } 2$	$-Y_n$	$-Z_{(n+2)/2}$
$-X_{n+1}, -X_{n+2}$	$(127-(n+1)) \text{Modulo } 2$	$-Y_{n+1}$	
...	...*	...	...
$-X_{124}, -X_{125}$	0 0 0 0 0 0 1 1	$-Y_{124}$	$-Z_{63}$
$-X_{125}, -X_{126}$	0 0 0 0 0 0 1 0	$-Y_{125}$	
$-X_{126}, -X_{127}$	0 0 0 0 0 0 W_1 1	$-Y_{126}$	
$-X_{127}, -X_{128}$	0 0 0 0 0 0 W_2 0	$-Y_{125}$	$-Z_{64} \text{ OR } -Z_{63}$

DEFINITION OF W-BITS					
INFORMATION FRAME		SIGNALING FRAME			
W ₁	W ₂	SIG. BIT	W ₁	W ₂	DECODER LEVEL
0	1	1	0	0	Z ₆₄
		0	1	1	Z ₆₃

NOTE — *Codes Change In Normal Binary Sequence.

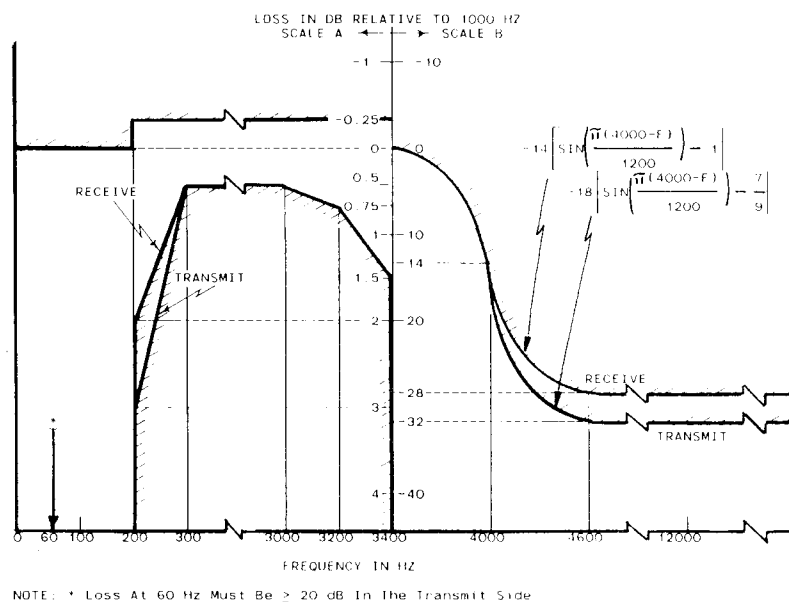
NOTE — **During Signaling Frame, The Eighth Bit Carries Signaling Information And The PCM Word Consists Of Bits 1 to 7.

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Figure 10—Four-Wire Frequency Response Boundaries



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Figure 11—Two-Wire Frequency Response Boundaries

4.2.2.5 Return Loss

Return loss (see Appendix B) is measured for 4-wire and 2-wire ports.

- 1) For 4-wire impedance (control of the deviation between the receive or transmit 4-wire test access port impedances for 150, 600, and 1200 Ω), the return loss should be measured against 150, 600, and 1200 Ω . The return loss measured at 1 kHz should be 28 dB minimum. The return loss measured at any frequency between 300 and 3000 Hz should be 23 dB minimum.
- 2) For 2-wire impedance, the return loss should be measured against 900 Ω or 600 Ω , as appropriate, in series with 2.16 μF . A loop resistance of 25 Ω should be inserted between the channel unit and the return loss measuring set. The far end shall be connected to a 4-wire channel unit with 600 Ω terminations. The return loss should be as follows:
Echo (ERL): 28 dB min.
Singing return loss (SRL Hi or Low): 20 dB min.

4.2.2.6 Trans-hybrid Loss

Trans-hybrid loss, also called terminal balance return loss, is a measure of how much isolation the hybrid is providing from the transmit to the receive side with the 2-wire port terminated as it would be in operation (see [14]).

Trans-hybrid loss should be as follows:

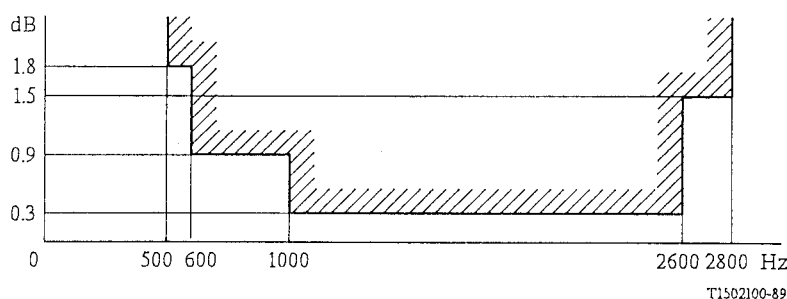
Echo (ERL): 34 dB min

Singing return loss (SRL Hi or Low): 20 dB min

4.2.2.7 Envelope Delay and Envelope Delay Distortion

Differences in velocity of propagation of different frequencies in the signal passband of a channel cause relative delay between the frequencies in the signal at the channel output. Envelope (or group) delay distortion is the derivative of channel phase response with respect to frequency. Ideally, the phase response should be linear, indicating that all frequencies are equally delayed (see IEEE Std 743-1984 [13] for amplitude-modulated test signal).

- 1) Envelope delay (analog in to analog out) at the frequency of minimum envelope delay is measured by the shift in the time of the envelope of the amplitude-modulated test signal. The envelope delay should not exceed 600 μs .
- 2) The minimum value of envelope delay is taken as the reference for the envelope delay distortion.
- 3) Envelope delay distortion with frequency should lie within the limits shown in the template of Fig 12.
- 4) The objectives above should be met with an input power level of -13 dBm .



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Figure 12—Envelope Dealy Boundaries

4.2.2.8 Noise

Noise includes idle channel noise and impulse noise.

4.2.2.8.1 Idle Channel Noise

The noise present in a communication channel when no signals are applied is idle channel noise. The conditions and terminations must be stated for the value to be significant. Noise in voice frequency channels is measured as weighted or flat. Weighted noise is measured with a C-message weighting network (see IEEE Std 743-1984 [137]) that is included in the meter circuit to measure the disturbing effect as it would appear to the telephone user. The noise should be no more than 20 dBrnC0 (objective 17 dBrnC0) for the transmit or receive section of the primary multiplex. (See 4.2.2.8.3 for noise measurements on companded and PCM systems.)

4.2.2.8.2 Impulse Noise

Impulse noise should be measured with and without a 1004 Hz (1002 Hz to 1020 Hz) holding tone. When measured with a holding tone, the impulse noise requirements are referenced to the received holding tone level.

- 1) *Impulse noise without a holding tone.* Impulse noise is measured with any assortment of channel units in the primary multiplex equipment. Measurements are made to the transmit section, receive section, or overall equipment, and impulse noise should be measured with the channel under test terminated and all other channels activated by signaling in either direction. With C-message weighting employed the impulse noise should satisfy the following:

Noise Measuring Set Threshold Level (dBrnC0)	Maximum Impulse Counts in 30 min
41	10
51	1
58	0.1

¹³The full text of the Blue Book, from which this figure is extracted, may be ordered from the International Telecommunication Union, Sales Section, Place des Nations, CH 1211 Geneva, Switzerland.

- 2) *Impulse noise with a 1004 Hz holding tone.* Tests needed with –13 and –23 dBm0, 1004 Hz holding tones looking for zero counts in 5.2 min at thresholds 9 dB below the received holding tone level.

4.2.2.8.3 Noise with Holding Tone

Noise with holding tone is a measurement of the total C-notched noise at the output (including distortion components) with a –13 dBm0, 1002 Hz to 1020 Hz signal applied at the input. The performance is controlled by signal-to-distortion requirements. This is the required method for performing noise measurements on companded or PCM systems.

4.2.2.9 Discrimination Against Out-of-Band Input Signals

Discrimination against out-of-band input signals is a measure of the effectiveness of the transmit filter in removing undesirable signal components.

The parameter values for discrimination of out-of-band input signals are given under single frequency distortion.

4.2.2.10 Spurious In-band Output Signals

Spurious in-band output signals are the ratio of the selectively received in-band image signals at the output port to a 0 dBm0 sine wave 700-1100 Hz applied to the input port. The received signal frequency must not be the same as the input signal.

The objectives for spurious in-band output signals are given under single frequency distortion.

4.2.2.11 Spurious Out-of-band Output Signals

Spurious out-of-band output signals are the ratio of the selectively received out-of-band image signals at the output port to a 0 dBm0 sine wave 700-1100 Hz applied to the input port.

The objectives for spurious out-of-band signals are given under single frequency distortion.

4.2.2.12 Longitudinal Balance

Longitudinal balance is a measure of the effect on the metallic (2- or 4-wire balanced transmission) of longitudinal disturbances. Measurements use the methods of IEEE Std 455-1985 [11] and are indicated as longitudinal-to-metallic balance (balance L-M).

According to the methods of IEEE Std 455-1985 four discrete frequencies, in the range from 60 to 3400 Hz, are deliberately offset from the dominant power harmonics to avoid inadvertent masking of performance by power system induction. A longitudinal source with voltage V_s is applied at these frequencies to an interface for a particular type of connection. The resulting metallic voltage, V_m , is measured at both interfaces. The longitudinal source is then to be applied at the other interface, and both metallic voltages again measured. For any given path of a connection path type, the longitudinal-to-metallic mode is then defined as follows:

$$\text{Balance (L – M)} = 20 \log \frac{V_s}{V_m},$$

where V_m is the largest of the four metallic voltages measured for the connection.

With longitudinal source voltage V_s applied at the interface, the resulting metallic voltage V_m should satisfy the following:

Test Signal Frequency Hz	Minimum 20 log 2-wire	V_s/V_m 4-wire (–16 TLP)
200	58	74
500	58	74
1000	58	74
3000	53	69

With the above test arrangement and with the primary multiplex connected to a far-end primary multiplex, the metallic voltage at the far-end primary multiplex output (when referred to the input TLP) should satisfy the same objectives as either V_m or E_m .

4.2.2.13 Distortion

Distortion is the difference between the wave shapes of an input signal and the output signal it produces. The distortion is the result of nonlinear processes in the transmission of the signal. Two types of nonlinear processes occur: intentional from quantization of analog signals; and unintentional as occurs with the use of nonlinear elements in transmission equipment. The former gives rise to quantization distortion; the latter to single frequency and intermodulation distortion generally referred to as nonlinear distortion. Nonlinear distortion measurements are made usually in terms of intermodulation distortion measurements.

4.2.2.13.1 Signal to Distortion (Total Distortion)

Total distortion is the sum of all in-band distortion measured relative to input signal power. In the process of PCM, the continuous range of values of an analog signal is divided into non-overlapping, but not necessarily equal subranges, and a discrete value of the output is uniquely assigned to each subrange. Whenever the input value falls within a given subrange, the digital output has the corresponding discrete value. The result is a stepped waveform approximation to the input analog waveform. The difference between these waveforms is called quantization distortion and the undesirable random signal caused by the error of approximation is called quantization noise. The parameter measuring the quality of a PCM process is called the signal-to-distortion ratio, defined as the logarithmic ratio of the signal power to the quantization noise power.

Measurement of this ratio includes measurement of other distortion factors, which should be small with respect to the quantization distortion, hence the term "total distortion."

The signal to total distortion (C-notched noise) should meet the following requirements:

Input Level	Minimum S/D (dB)	
	Overall	Receive or Transmit
0 to –30 dBm0	33	35
–40 dBm0	27	29
–45 dBm0	22	25

The input frequency of the holding tone must be in the range of 1004 to 1020 Hz, and, as stated in IEEE Std 743-1984 [13], the total distortion of the holding tone should be down 50 dB in the band of 200 Hz to 4000 Hz.

In some cases it may be desirable to meet the objectives of the above table with input frequencies of 304 and 2996 Hz when measured by notching out the 304 or 2996 Hz signal and using C-message weighting to measure the total distortion power.

The signal to distortion for the receive section is measured with test equipment that provides a digital input to the received section and is capable of measuring the analog output. The signal to distortion for the transmit section is measured with test equipment that provides analog input to the transmit section and is capable of computing or measuring the digital output.

4.2.2.13.2 Single Frequency Distortion

Single frequency distortion is noise resulting at the output of a nonlinear transmission element when a single frequency sinusoid is applied at the input. The output yields a distortion of the pure sinusoid in that the resulting waveform consists of components other than the applied frequency. These additional components are called the single frequency noise.

The overall requirement with a 0 dBm0 sinusoidal input is as follows:

Input	Output
Swept from 50 to 12 kHz:	–28 dBm0 or less at any other frequency
Selected tone (between 1004 and 1020 Hz):	–140 dBm0 or less at any other frequency 50 Hz to 4 kHz

With inputs from 50 Hz to 3.4 kHz, the –28 dBm0 requirements depends on the characteristics of the receive filter. With inputs from 4.6 to 12 kHz, the performance depends on the characteristics of the transmit filter and the requirement is –32 dBm0. In the band from 3.4 to 4.6 kHz, the performance depends on both receive and transmit filters. The filter characteristics shown in Figs 10 and 11 are designed to provide satisfactory performance.

4.2.2.13.3 Intermodulation Distortion

Intermodulation distortion resulting from nonlinearities is measured with the 4-tone method.

Intermodulation distortion should be small enough so that with an input power of –13 dBm0, the second-order intermodulation distortion ratio is no less than 50 dB and the third-order intermodulation distortion ratio is no less than 54 dB (see IEEE Std 743-1984 [13]).

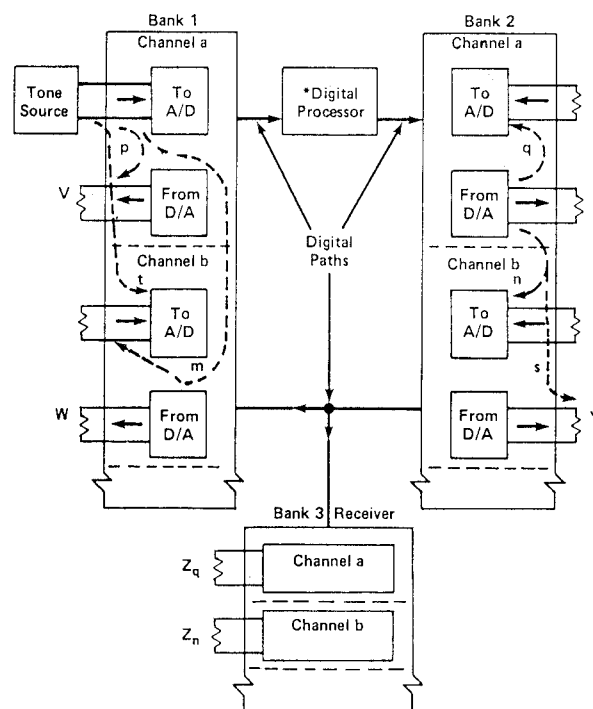
4.2.2.14 Insertion Loss

Insertion loss is the difference between input power and output power for any input signal and is usually measured with a 1002 to 1020 Hz input signal at 0 dBm0.

4.2.2.15 Voice Frequency Crosstalk

Crosstalk is an unwanted coupling of signal power from one signal path to another. Crosstalk includes near-end crosstalk, where an outgoing wanted signal appears as an incurring unwanted signal on the same or another channel (if the unwanted signal is on the incoming path of the same channel as the outgoing wanted signal, the coupling is called intrachannel crosstalk); and far-end crosstalk, where an outgoing wanted signal appears as an outgoing unwanted signal on another channel (see Fig 13).

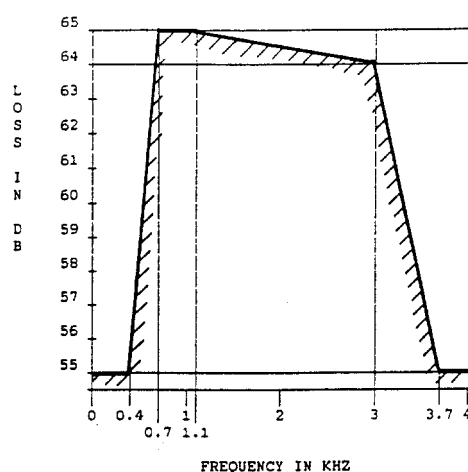
The primary multiplex equipment should meet the crosstalk requirements with any of the channels tested as the disturbing channel and any other channel tested (near or far end) as the disturbed channel. Four-wire intrachannel crosstalk should also meet the requirements. With a 0 dBm0 signal being transmitted on the disturbing channel terminated in its nominal impedance, the minimum allowable equal level crosstalk coupling loss for the overall circuit (transmit plus receive portion contributions) measured with a frequency selective meter should meet the bounds in Fig 14. For the transmit or receive portion alone the loss should meet a bound, on an equal level basis, at least 6 dB greater than that shown in Fig 14.



- Notes: * Used To Force All Ones In Receiver For Disturbing Channel.
1. Channel "a" Is The Disturbing Channel.
 2. Channel "b" Is The Disturbed Channel.
 3. Terminate Disturbed Channels In Appropriate Impedances.
 4. To Measure Z_q Or Z_n , Connect Digital Line To Bank 3 Receiver (See Text).

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Figure 13—Crosstalk Measurement Connections



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Figure 14—Minimum Equal Level Coupling

4.2.2.16 Load Capacity

The load capacity of a terminal is the maximum signal power it can handle without distorting the output or having other bad effects on the terminal. For primary multiplex equipment the load capacity is determined by the definition of the digital milliwatt and the companding law. The resulting overload is slightly greater than +3 dBm0 (+3.17 dBm0).

The performance is controlled by signal-to-distortion requirements.

4.2.2.17 Coder Offset

Coder offset is the difference between the code that is supposed to result from a zero-volt input to the encoder and the code that occurs.

For operating channels, this is controlled by total distortion requirements. When the analog input of an equipped channel in the primary multiplex equipment is terminated in an appropriate idle channel impedance (with no speech or other signal energy present), the coder offset for that channel should have a magnitude no greater than step 16 of the nonlinear coding characteristic (see Fig 8). When a channel position in the channel bank is unequipped, the coder offset for that channel must have a magnitude no greater than step 48 of the nonlinear coding characteristic.

4.2.2.18 Peak Codes

Peak codes are determined by the definition of the digital milliwatt and the companding law. The performance is controlled by signal-to-distortion requirements.

4.2.2.19 Signaling

Signaling information between network elements controls such things as routing of connections through the network, alerting the called end of connections, and control of billing for calls. This information may be contained in designated

signaling bits in the DS1 bitstream or may be separately routed through common (refer to [24]) control signaling systems such as CCITT Signaling System No. 7. Terminals that do analog to digital conversion may also have to do signaling conversion between dc or ac signaling states on the metallic connections to the terminal and the built-in signaling bits. There are many different signaling states that need to be converted depending on the external equipment connecting to the digital terminal, including dial pulsing, E&M lead signaling, and foreign exchange loop start and ground start signaling. The ability to extract and insert signaling bits in the DS1 bitstream is required. Signaling parameter measurement is outside the scope of this document.

4.2.2.19.1 Signaling Freezing

On detection of loss of frame, the on-hook and off-hook signaling states presented to the channel units should be maintained in the state that existed before detection of the condition. This is called signaling freezing and requires buffering of the signaling information.

The probability of freezing in the state for a 50% ones-density out-of-frame condition should be no more than 5%. The signaling states must not be changed until the terminal regains synchronization or declares a carrier failure alarm.

4.2.3 Primary Multiplex Equipment: DS1 Parameters

4.2.3.1 Performance Parameters

4.2.3.1.1 Return Loss

The definition of return loss is given in Appendix B and is measured at the DS1 port of the primary multiplex equipment.

The return loss should be a minimum of 26 dB.

4.2.3.1.2 Error Measurements

The bit-error ratio (BER) is the fundamental measure of digital performance in that is the fraction of bits that are not received correctly. No error detection scheme (short of a bit-by-bit comparison) can detect all bit errors, but all signal formats allow for an approximation to be made. Detection methods for estimating bit-error ratio in use include bipolar violations, frame alignment signal errors, and cyclic redundancy check (CRC) violations.

Capability may be provided within primary multiplex equipment to monitor in service for bit-error performance, out of frame (OOF) or change of frame alignment count (COFA), and CGA count. When an out-of-frame condition occurs, bit-error monitoring should be stopped. When framing is resumed, bit-error monitoring should be restarted. Capabilities may be provided within the primary multiplex equipment for options to monitor (see Section 2. for definition):

- 1) Bit-error count for a selectable or open time interval
- 2) Bipolar violation ratio—bit-error ratio estimated from bipolar violations
- 3) Framing bit-error count for a selectable or open time interval
- 4) Bit-error ratio estimated from framing bit errors
- 5) CRC violation count for a selectable or open time interval
- 6) Bit-error ratio estimated from CRC
- 7) Errored second count for a selectable or open time interval
- 8) Error-free second count for a selectable or open time interval
- 9) Severely errored second count for a selectable or open time interval
- 10) Unavailable seconds for a selectable or open time interval could be expressed as a count or a percentage
- 11) Degraded minute count for a selectable or open time interval