

IEEE Standard Equipment Requirements and Measurement Techniques for Analog Transmission Parameters for Telecommunications

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Abstract: Performance requirements for test equipment that measures the analog transmission parameters of subscriber loops, message trunks, PBX trunks, and ties lines are specified. Requirements for these measurements with DS1 bit stream access are also provided. The measurement of loss, noise, and impulse noise on non-loaded cable pairs used for digital subscriber lines is addressed.

Keywords: active speech level, ADSL, amplitude jitter, digital bit stream access, digital subscriber lines, dropouts, echo canceller and suppressor disabling, echo path loss and delay, envelope delay distortion, frequency shift, gain hits, HDSL, impulse noise, intermodulation distortion, ISDN, loss, network impulse response, noise, phase hits, phase jitter, return loss, round trip delay, signal-to-total-distortion ratio, timing offset, twenty-three-tone measurements

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Introduction

(This introduction is not part of IEEE Std 743-1995, IEEE Standard Equipment Requirements and Measurement Techniques for Analog Transmission Parameters for Telecommunications.)

The provision of standard technical requirements for analog transmission measuring equipment is intended to assure comparable measurement results and proper interworking between different test equipments. It is not necessary that all the measurements specified be provided in one piece of test equipment; a manufacturer may choose to provide only a subset of the measurements specified in this standard.

The purpose of this standard is to provide technical requirements for the generation and measurement of analog transmission signals used to characterize the telecommunications network. Such characterization includes measurement of various impairments to voiceband (20–4000 Hz) signals (speech and data) at both analog and PCM access points.

This standard also provides filters for the extended frequency range of signals now carried on a traditional analog twisted copper cable pair between the central office and the customer premises. These filters permit level, noise and impulse noise measurements on lines providing digital transmission for the Integrated Services Digital Network (ISDN), the High-bit-rate Digital Subscriber Line (HDSL), and the Asymmetric Digital Subscriber Line (ADSL). The traditional C-Message weighting is intended to be used only for voice-related measurements, and a new data filter (D Filter) is intended to be used for voice-frequency digitally-related measurements. Other filters that have been replaced or deleted are listed.

The initial issue of this standard, IEEE Std 743-1984, IEEE Standard Methods and Equipment for Measuring the Transmission Characteristics of Analog Voice Frequency Circuits, was based on AT&T PUB 41009, Transmission Parameters Affecting Voiceband Data Transmission, May, 1975. As noted by the title of IEEE Std 743-1984, the measurements were largely restricted to voiceband. Telecommunications networks up to this time were composed largely of voiceband circuits on cable pairs and analog carrier on open wire, cable, or radio.

New measurement techniques and improvements to the existing standard have been added since the previous 1984 edition. Improved technology (microprocessors and digital signal processors) permits more sophisticated measurements to be incorporated into the transmission test equipment. Some of these new measurements have warranted the downgrading or even removal of measurement techniques from the previous 1984 standard.

Several new measurement techniques are included in this standard as follows:

- 23-tone system straightaway measurement of
 - a) Loss vs. frequency (attenuation distortion)
 - b) Envelope Delay Distortion (EDD)
 - c) Signal-to-Total Distortion-ratio (S/TD, including true ADPCM S/TD)
 - d) Second-order and third-order intermodulation distortion (IMD) measurements
 - e) Frequency shift and timing offset
- Round-trip envelope delay, including an extended round-trip envelope delay measurement with ranges of 120 ms and 1200 ms
- Adaptive filtering technique on a pseudo-random noise signal to provide echo path loss and echo path delay
- Network impulse response technique on a pseudo-random noise signal to measure straightaway attenuation distortion, EDD and S/TD
- Active speech level measurement

Some measurements have been removed or downgraded as follows:

- Peak-to-average ratio (P/AR) has been removed
- Four-tone intermodulation distortion (IMD) has been downgraded (replaced by the IMD measurement capabilities of the 23-tone system)

The removal or downgrading of a measurement technique does not mean the measurement can no longer be used. For example, specifying the new holding tone frequency of 1020 Hz does not mean that 1004 Hz is unsatisfactory or needs to be replaced. Crosstalk and frequency-selective level measurements have been dropped from the standard, as have the mandatory mechanical shock and vibration requirements, and classes of test instruments.

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IEEE Standard Equipment Requirements and Measurement Techniques for Analog Transmission Parameters for Telecommunications

1. Overview

This standard has introductory material (Overview, References, Definitions, etc.) followed by a synopsis of the measurement techniques in clause 3, General descriptions of measurements. The digital interface has been added since the 1984 standard, and is described in clause 4, Digitally-encoded analog measurements. Clauses 5 through 8 give specific requirements for signal generators and techniques for measurement of transients and various steady-state impairments. The filters are grouped into clause 9, followed by interface and environment requirements in clauses 10 and 11. Ten annexes cover additional information on the techniques of clauses 5 through 8.

1.1 Scope

This standard will be applicable for specifying test equipment for measuring the analog transmission parameters of subscriber loops, message network trunks, PBX trunks and tie lines, and the analog transmission parameters of digital subscriber lines. Techniques are specified for measuring parameters not previously measurable.

1.2 Purpose

The purpose of this standard is to establish measurement techniques for analog transmission parameters for telecommunications services, and to establish requirements for test equipment used at analog and digital appearances.

1.3 Application

The test equipment specified in this standard can be used to measure analog characteristics from both analog and digital access points. The newly-specified digital access permits such measurements as round-trip delay using the digital equivalent of analog signals, which permits the round-trip delay measurement without bit integrity.

1.4 Shall vs. should

The word *shall* is used to indicate mandatory requirements to be followed in order to conform to the standard and from which no deviation is permitted (*shall* equals *is required to*). The word *should* is used to indicate that among several possibilities, one is recommended and particularly suitable, without mentioning or excluding others; or that a certain course of action is preferred but not necessarily required; or that (in the negative form) a certain course of action is deprecated but not prohibited (*should* equals *is recommended that*). The word *may* is used to indicate a course of action is permissible within the limits of the standard (*may* equals *is permitted*).

1.5 Annexes

All of the annexes to this standard are informative, and are included in the standard for information purposes only.

1.6 References

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

ANSI/EIA/TIA 232-E-1991, Interface between Data Terminal Equipment and Data Circuit—Terminating Equipment Employing Serial Binary Data Interchange, July 1991.¹

ANSI/EIA/TIA 530-A-1992, High-Speed 25-Position Interface for Data Terminal Equipment and Data Circuit—Terminating Equipment, Including Alternative 26-Position Connector.

ANSI T1.102-1993, American National Standard for Telecommunications—Digital Hierarchy—Electrical Interfaces.

ANSI T1.221-1995, American National Standard for Telecommunications—Operations, Administration, Maintenance, and Provisioning (OAM&P)—In-Service, Nonintrusive Measurement Device (INMD)—Voice Service Measurements.

ANSI T1.401-1993, American National Standard for Telecommunications—Interface Between Carriers and Customer Installations—Analog Voice Grade Switched Access Lines Using Loop-Start and Ground-Start Signaling.

ANSI T1.403-1995, American National Standard for Telecommunications—Network-to-Customer Installation—DS1 Metallic Interface.

ANSI T1.413-1995, Network and Customer Installation Interfaces—Asymmetric Digital Subscriber Line (ADSL) Metallic Interfaces.

ANSI T1.506-1990, American National Standard for Telecommunications—Switched Exchange Access Network Transmission Specification.

ANSI T1.506a-1992, American National Standard for Telecommunications—Network Performance—Transmission Specifications for Switched Exchange Access Network (Absolute Round-Trip Delay) (supplement to ANSI T1.506-1990).

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

ANSI T1.601-1992, American National Standard for Telecommunications—Integrated Services Digital Network (ISDN)—Basic Access Interface for Use on Metallic Loops for Application on the Network Side of the NT (Layer 1 Specification).

CCITT Blue Book Fasc. III.4 Recommendation G.711 (11/88) Pulse Code Modulation (PCM) of Voice Frequencies.²

CCITT Blue Book Fasc. III.4 Recommendation O.6 (11/88) 1020 Hz reference test frequency.

CCITT Blue Book Fasc. III.4 Recommendation O.81 (11/88) Group-delay measuring equipment for telephone type circuits.

CCITT Blue Book Recommendation V.25 (1988), Automatic answering equipment and/or parallel automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls.

Committee T1 Technical Report 28, High-bit-rate Digital Subscriber Line (HDSL), 1994.³

IEEE Std 100-1992, The New IEEE Standard Dictionary of Electrical and Electronic Terms (ANSI).⁴

IEEE Std 455-1985 (Reaff 1992), IEEE Standard Test Procedure for Measuring Longitudinal Balance of Telephone Equipment Operating in the Voice Band (ANSI).

IEEE Std 488-1987, IEEE Standard Digital Interface for Programmable Instrumentation (ANSI).

IEEE Std 1007-1991, IEEE Standard Methods and Equipment for Measuring the Transmission Characteristics of Pulse-Code Modulation (PCM) Telecommunications Circuits and Systems (ANSI).

ITU-T Recommendation O.22 (10/93), CCITT automatic transmission measuring and signaling testing equipment ATME No. 2.

ITU-T Recommendation O.41 (01/95), Psophometer for use on telephone-type circuits, Revision 2.⁵

ITU-T Recommendation V.11 (02/94), Electrical characteristics for balanced double-current interchange circuits operating at data signal rates up to 10 Mbits/s, Revision 1.

UL 1244-1993, Electrical and electronic measuring and testing equipment.⁶

UL 1459-1987, Telephone equipment.

²CCITT publications are available from the International Telecommunications Union, Sales Section, Place des Nations, CH-1211, Genève 20, Switzerland/Suisse. They are also available in the United States from the U.S. Department of Commerce, Technology Administration, National Technical Information Service (NTIS), Springfield, VA 22161, USA.

³T1 Technical Reports are available from the Alliance for Telecommunications Industry Solutions (ATIS), 1200 G St., NW, Suite 500, Washington, DC, 20005, USA.

⁴IEEE publications are available from the Institute of Electrical and Electronics Engineers, 445 Hoes Lane, P.O. Box 1331, Piscataway, NJ 08855-1331, USA.

⁵ITU-T publications are available from the International Telecommunications Union, Place des Nations, CH-1211, Geneva 20, Switzerland. They are also available in the United States from the U.S. Department of Commerce, Technology Administration, National Technical Information Service (NTIS), Springfield, VA 22161, USA.

⁶UL publications are available from Underwriters Laboratories, Inc., 333 Pfingsten Road, Northbrook, IL 60062-2096, USA.

2. Definitions and abbreviations

2.1 Definitions

This clause contains those definitions relating to the measurement of analog and digitally-encoded analog signals, which may or may not be listed in IEEE Std 100-1992 dictionary. The basic source for many definitions is ANSI T1.506-1990 and ANSI T1.506a-1992, covering switched access services, and Bellcore TR-NWT-000335 [B1]⁷, covering special access services.

2.1.1 absolute delay: The interval of time between the transmission of a signal and the reception of the same signal (or its associated signal, if in a different domain) at a different point in a circuit. [IEEE Std 100-1992] (*Syn.*: transmission delay or propagation delay.)

2.1.2 active speech level: A period of time during which speech spurt intervals are followed by speech pause intervals. See 8.9.

2.1.3 attenuation distortion: The change in attenuation at any frequency with respect to that at a reference frequency. [IEEE Std 100-1992]

2.1.4 C-Message weighting: A frequency-weighting characteristic used for measurement of noise in voice-frequency communications circuits and designed to weight noise frequencies in proportion to their perceived annoyance effect in telephone service. The C-Message weighting is used to evaluate noise to the human ear, using a 500-type telephone set.

2.1.5 C-Message noise: The noise on an idle channel or circuit, i.e., a channel or circuit with a termination and no signal (holding tone) at the transmitting end, measured through a C-Message weighting. The noise is expressed in dBnC.

2.1.6 C-Notched noise: The noise power on a channel with a holding tone (signal) at the transmit end, measured through a C-Message weighting and a 1010 Hz notch filter in tandem.

2.1.7 D Filter: A 300 Hz to 3400 Hz bandpass filter used for measuring noise, impulse noise, or data modem signal power. Noise measured through the D-Notched filter is used to evaluate its effect on the performance of a data modem.

2.1.8 D noise: The noise on an idle channel or circuit, i.e., a channel or circuit with a termination and no signal (holding tone) at the transmitting end, measured through a D Filter. The noise is expressed in dBnD.

2.1.9 D-Notched noise: The noise power on a channel with a holding tone (signal) at the transmit end, measured through a D Filter and a 1010 Hz notch filter in tandem. May also be measured as a Signal-to-D Notched-Noise ratio (S/DNN) in decibels (dB).

⁷The numbers in brackets correspond to the bibliographic items listed in annex J.

2.1.10 digital milliwatt: The repetitive transmission of a sequence of codes in a given channel, which will be decoded in a receiving terminal as a 0 dBm, 1000.0 Hz signal (see table 1 in 4.2.2.2). [IEEE Std 100-1992]

The digit sequence was established by the CCITT (now ITU-T) in CCITT Blue Book Recommendation G.711 (1988). The digital milliwatt, which has no quantizing noise, is generally not transmitted over a telecommunication facility because it can cause false framing on D4-formatted DS1 facilities.

2.1.11 Digital Reference Sequence (DRS): A 797-byte representation, in A-law or μ -law coding (as listed in annex B), of one of three levels (0, -10, or -13 dBm) of a 1013.8-Hz sine wave. With digital bit integrity, the 797-byte sequence may also be used for bit error ratio testing through the network.

2.1.12 E Filter: A 1 kHz to 50 kHz bandpass filter used for measuring the power of a digital data signal, noise, or impulse noise on an ISDN basic access digital subscriber line.

2.1.13 echo path delay (EPD): The time difference between an incident signal and the returned reflection of that incident signal. There can be multiple echo paths with different delays.

2.1.14 echo path loss (EPL): The difference in dB between an incident signal and the returned reflection of that signal. There can be multiple echo paths with different losses.

2.1.15 echo return loss (ERL): The frequency weighted average of the return losses over the middle of the voice band, with the far end terminated with a specified impedance. The 3 dB down frequencies of the weighting are 560 Hz to 1965 Hz (see also singing return loss).

2.1.16 echo canceler disabling tone: A 1350 ms, 2100-Hz tone sequence, which includes two phase reversals. It is used to disable echo cancelers and echo suppressors. (See 4.3.)

2.1.17 envelope delay: The time that the envelope of a modulated signal takes to pass from one point in a network (or transmission system) to a second point in the network. Envelope delay is often defined as the rate of change, with angular frequency, of the phase shift between two points in a network. [IEEE Std 100-1992] (The ITU-T term *group delay* has the same definition).

Examples of envelope delay are as follows:

- a) The time it takes the envelope of a carrier frequency amplitude modulated by 83 1/3 Hz to pass between two points in a network. (See 8.1.)
- b) The rate of change, with angular frequency, of the phase shift between adjacent pairs of tones (frequency difference of 156.25 Hz) in the 23-tone test signal. (See 8.6.3.)
- c) The rate of change, with angular frequency, of the phase shift measured between frequencies spaced 31.25 Hz apart in the Network Impulse Response test signal. (See 8.8.)

2.1.18 envelope delay, round trip: The sum of the outgoing envelope delay and return path envelope delay, where provision has been made at the far end of the circuit to either loopback or remodulate the envelope delay test signal back to the transmitting measuring set.

2.1.19 envelope delay distortion (EDD): The difference between the envelope delay at a test frequency and the envelope delay at a reference frequency.

2.1.20 equal level echo path loss (ELEPL): The measure of echo path loss at a four-wire interface that is corrected by the difference in dB, between the transmit and receive TLPs.

2.1.21 extended round trip envelope delay: The round-trip envelope delay of a network as measured by a small shift in envelope modulation frequency to extend the range of measurement beyond the normal range limit of $1/f_{\text{modul}}$.

2.1.22 frequency shift: A condition on a connection where all the frequencies in a signal are shifted by the same amount.

2.1.23 F Filter: A 5 kHz to 245 kHz bandpass filter used for measuring the power of a High-bit-rate Digital Subscriber Line (HDSL) signal, noise, or impulse noise on an HDSL.

2.1.24 G Filter: A 20 kHz to 1100 kHz band pass filter used for measuring the power of an Asymmetric Digital Subscriber Line (ADSL) signal, noise, or impulse noise on an ADSL.

2.1.25 holding tone: The tone, near 1 kHz, transmitted over a telecommunication circuit for performing noise-with-tone, jitter, and transient impairment measurements.

2.1.26 impulse noise: Any burst of noise that produces a voltage exceeding the rms value of the background or quantizing noise by more than 12 dB. [IEEE Std 100-1992]

2.1.27 intermodulation distortion (IMD): 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 two or more of the input frequencies. IMD is the ratio, in dB, of the test signal to specific spurious output signals generated by the nonlinearity.

2.1.28 level: The level in dBm of a signal or tone into a resistive load equal to the designated impedance at the point of measurement. Levels measured into complex impedances, as specified by ITU-T Recommendation O.41, clause 3.13, are not recognized by this standard.

2.1.29 longitudinal balance: The electrical symmetry of the equipment on two wires comprising a pair with respect to ground. The term is used to express the degree of susceptibility to longitudinal interference. [IEEE Std 100-1992]

2.1.30 phase jitter: Any deviation or fluctuation of the zero crossings of a holding tone or combination of tones from their nominal position in time.

2.1.31 quantizing noise: The noise introduced during the process of digitally encoding an analog signal. [IEEE Std 100-1992]

2.1.32 round-trip delay: The sum of the absolute delays on an outgoing path and return path. Different methods of measuring round-trip delay may produce somewhat different results.

2.1.33 signal-to-noise ratio (S/N): The ratio in dB of the power of a single- or multiple-tone test signal to the power of the background noise (after a D Filter) not related to the application of the test signal, plus all spurious signals resulting from the application of the test signal, except for 2nd-order and 3rd-order intermodulation (IMD) distortion (see 8.6.6).

2.1.34 signal-to-total-distortion ratio (S/TD): The ratio in dB of the power of a single or multiple-tone test signal to the power of all spurious signals (after a D Filter) resulting from the application of the test signal plus the power from background noise.

2.1.35 singing return loss: The return loss of a circuit measured with two separately transmitted signals with a flat spectral distribution between 3 dB frequencies of 260 Hz and 500 Hz (SRL low) and 2200 Hz and 3400 Hz (SRL high). The lower of the two return losses (SRL low or SRL high) will be the best measure of the margin of the circuit against singing.

2.1.36 straightaway: A one-way measurement requiring no path or connection in the opposite direction.

2.1.37 three-tone slope: A measure of attenuation distortion at 404 Hz and 2804 Hz relative to the loss at 1020 Hz (or 1004 Hz). Three-tone slope is specified for many telecommunication services.

2.1.38 timing offset: The difference between two physical units' fundamental clock sources; those sources being the timing basis from which signals and sampling are derived and analyzed (usually expressed proportionally in parts per million). Timing offset will cause a uniform percentage change in signal frequencies.

2.1.39 tone-to-C-Notched noise ratio: The ratio in dB of the incoming holding tone power to the C-Notched noise power at the point of measurement. The incoming holding tone power is reduced in power by at least 50 dB by the 1010 Hz Notch.

2.1.40 tone-to-D-Notched noise ratio: The ratio in dB of the incoming holding tone power to the D-Notched noise power at the point of measurement. The incoming holding tone power is reduced in power by at least 50 dB by the 1010 Hz Notch.

2.1.41 total distortion: The summation of noise and distortion resulting from the application of a test signal (quantizing noise, phase jitter, intermodulation distortion, etc.), and the noise not related to the application of a test signal (background noise).

2.1.42 transmission level point (TLP): a point in a transmission system at which the ratio is specified in dB of the power of a test signal at that point to the power of a signal at a reference point. The reference level point, called the zero transmission level point (0 TLP), is an arbitrary established point relative to which transmission levels at all other points are specified. A signal level of X dBm at the 0 TLP is designated X dBm₀.

2.1.43 1 kHz envelope delay: The envelope delay at a carrier frequency of 1020 Hz.

2.2 Abbreviations and acronyms

The following abbreviations and acronyms, commonly used in the communications field, are used in this standard. Some are defined in 2.1. Listed with each abbreviation and acronym is the subclause in the remainder of this standard where it is introduced.

ADPCM	Adaptive differential pulse code modulation (8.2.1)
A/D	Analog-to-digital converter (8.7.3.3)
ADSL	Asymmetric Digital Subscriber Line (3.2.2)
AFIR	Adaptive finite impulse response filter (3.4.8)
CRC	Cyclic redundancy check (4.2.2.5)
D/A	Digital-to-analog converter (8.7.3.3)
dBm	Decibels relative to 1 mW (6.1)
dBm ₀	Decibels relative to 1 mW, referred to a zero transmission level point (6.1)
dB _{rn}	Decibels relative to 1 pW reference noise level (6.2)
dB _{rnC}	Decibels relative to 1 pW reference noise level, measured with a C-Message or C-Notch frequency weighting (6.2)
dB _{rnD}	Decibels relative to 1 pW reference noise level, measured with a D Filter or D-Notch Filter (6.2)
DFT	Discrete Fourier transform (8.6.1)
DSL	Digital subscriber line (3.2.2)
DSP	Digital signal processor (8.7.3.7)
DRS	Digital reference sequence for a 1013.8-Hz sine wave (5.3.3)
EDD	Envelope delay distortion (3.4.1)

ELEPL	Equal level echo path loss (8.3)
EPD	Echo path delay (3.4.7)
EPL	Echo path loss (3.4.7)
ERL	Echo return loss (8.3)
ESF	Extended superframe format (4.2.2.2)
HDSL	High-bit-rate Digital Subscriber Line (3.2.2)
IMD	Intermodulation distortion (3.4.5)
ISDN	Integrated Services Digital Network (3.2.2)
ITU-T	International Telecommunications Union—Telecommunications Standardization Sector (the former name is the CCITT) (3.2.2)
LMS	Least mean squares algorithm (8.7.2)
LSB	Least significant bit (4.2.2.2)
NIR	Network impulse response (3.4.8)
P/AR	Peak-to-average ratio (3.4.10)
PCM	Pulse code modulation (4.2.1)
RL	Return loss (8.3)
S/N	Signal-to-noise ratio (4.2.2.4)
SRL	Singing return loss (8.3)
S/TD	Signal-to-total-distortion ratio (3.2.3)
TLP	Transmission level point (4.2.2.2)
T/TD	Tone-to-total-distortion ratio (3.2.3)
2nd IMD	Second-order intermodulation distortion (8.6.5)
3rd IMD	Third-order intermodulation distortion (8.6.5)

3. General descriptions of measurements

3.1 Digitally-encoded measurements

The scope of this standard includes measurement of analog signals that have been sampled, quantized, and encoded in digital bit streams. Measurements of such signals will be accomplished by mathematically processing the train of sampled signal values represented by the digital bit stream. The formats and characteristics of the digital signals treated by this standard are specified in 4.2.

3.2 Steady-state measurements using a tone or quiet termination

3.2.1 Level and attenuation distortion

This is a measurement of signal power at an access point of nominal impedance in a circuit. The impedance may range from 100 Ω to 900 Ω , with 600 Ω most commonly used. The expected signal levels at the access points usually range from -40 dBm to $+15$ dBm.

The frequency range of interest for normal speech and voiceband data services may extend from below 60 Hz to above 4 kHz. Wide-band data circuits further extend this range. The requirements for level measurement with a sinusoidal tone are given in 6.1. The requirements for frequency response (or attenuation distortion) measurements with a 23-tone test signal are given in 8.6.2.

3.2.2 Noise

Noise measurements can be made with the channel or cable pair terminated in its nominal impedance and no test signal on the channel. These noise measurements are made using one of several frequency weighting networks/filters (see clause 9). These include:

- a) *C-Message*. A frequency weighting that weights the noise according to its perceived annoyance to a typical listener, of standard telephone service.
- b) *C-Notch*. C-Message weighting with the addition of a narrow stop-band or notch filter centered at 1010 Hz. This measurement is used to make a dynamic evaluation of the effects of noise on voice-grade services.
- c) *3.4 kHz Flat*. When used on voice frequency circuits, this filter permits the investigation of the presence of low-frequency noise (power induction, etc.).
- d) *D Filter*. A filter composed of a 300 Hz high-pass filter and a 3.4 kHz low-pass filter connected in tandem. The 300 Hz high-pass filter may be used separately as a low-frequency (power line) hum protection filter. The 3.4 kHz low pass filter may be used for all functions where the 3 kHz Flat filter had been used previously. The 3.4 kHz low-pass filter is designated as 3.4 kHz Flat.

NOTE—The analog and digital pole and zero specifications are provided in clause 9 to assure proper transient performance.

- e) *15 kHz Flat*. A filter used when making unweighted measurements of noise on program circuits. It is a 15 kHz low-pass filter of Butterworth shape rolling off at 12 dB per octave. It is not ordinarily used on voice message circuits.
- f) *Psophometric*. Psophometric weighting is specified by the ITU-T in ITU Recommendation O.41 (01/95) for use in measuring noise on telephone circuits in countries that recognize the ITU-T Recommendations. It differs from the C-Message weighting.
- g) *E Filter*. A filter with a 1–50 kHz pass band used for testing the Integrated Services Digital Network (ISDN) digital subscriber line (DSL). The E Filter analog pole-zero specifications assure repeatable transient measurements for different test sets.
- h) *F Filter*. A filter with a 5–245 kHz pass band for testing the High-bit-rate Digital Subscriber Line (HDSL). The F Filter analog pole-zero specifications assure repeatable transient measurements for different test sets.
- i) *G Filter*. A filter with a 20–1100 kHz pass band for testing the Asymmetric Digital Subscriber Line (ADSL). The G Filter analog pole-zero specifications assure repeatable transient measurements for different test sets.

3.2.3 Signal-to-total-distortion ratio (S/TD)

A holding tone, nominally 1 kHz (see 5.3), may be transmitted, then notched out before the noise and total distortion is measured. Alternatively, a multiple-tone signal may be transmitted (see 8.6.4 or 8.8.3), and the total distortion determined by means of a DFT algorithm. The holding tone or multiple tones provide a signal in the channel so that harmonic distortion, quantizing noise, phase jitter, and amplitude jitter become part of the total distortion measurement. If a holding tone is transmitted, the resultant noise measurement after the holding tone has been notched out by the 1010 Hz Notch filter (see 9.3) is technically a tone-to-total-distortion (T/TD) measurement or notched noise measurement (after weighting the total distortion with a D Filter or a C-Message weighting). S/TD measurements give a measure of the noise encountered by a continuous data signal or the noise a listener would hear during a speech burst. S/TD is the ratio of the received power of the holding tone or multiple-tone signal to the remainder of the energy (after weighting the total distortion with a D Filter or a C-Message weighting).

3.2.4 Phase jitter

Phase jitter is the summation of incidental phase modulation and the effects of interference and noise. The phase modulation component of phase jitter typically results from unwanted phase modulation on carrier supplies in frequency division multiplex terminals. The modulation frequencies are usually low, consisting of ringing frequency, power-line frequencies, and their harmonics. Phase jitter is normally measured by examining phase disturbances on a test tone. See 6.3 for requirements.

3.2.5 Amplitude jitter

Amplitude jitter is the summation of incidental amplitude modulation and the effects of interference and noise. Amplitude jitter is usually measured by examining amplitude disturbances on a test tone. See 6.4 for requirements.

3.2.6 Frequency

The measurement of the frequency of a transmitted or received signal. It is usually made while setting up a test signal or in conjunction with the measurement of other parameters such as level or envelope delay distortion.

3.3 Transient measurements using a tone or quiet termination

The general requirements for transient measurements are in 7.2.

3.3.1 Impulse noise

Noise bursts or spikes (impulses) that are much higher than normal peaks of the background noise or noise-with-tone are considered impulse noise. An impulse noise measurement is made by counting the number of impulses exceeding a preset threshold. Often it is desirable to measure the amplitude distribution of the noise impulses. This is done by counting the impulses exceeding three different thresholds. Impulse noise impulses may have extremely sharp rise times and may cause filters to ring. To avoid counting the oscillations due to the ringing, the counter(s) is blanked for a period of time after a threshold has been exceeded. See 7.3 for requirements.

3.3.2 Phase and gain hits

A rapid change in phase or gain of a signal is called a phase hit or a gain hit. These hits are measured by monitoring a received tone (holding tone) and counting hits that exceed selectable thresholds for specified durations. See 7.4 and 7.5 for requirements.

3.3.3 Dropouts

A 12 dB or greater reduction in the holding tone level as measured at the start of the measurement interval is defined as a dropout. See 7.6 for requirements.

3.4 Steady-state measurements using multiple-tone signals

3.4.1 Envelope delay distortion (EDD)

Envelope delay distortion (EDD) is the measurement of the phase linearity of a channel. Envelope delay, or the derivative of phase with respect to frequency, is measured because of the difficulty in establishing a phase reference and also because some channels, including analog carrier systems, have a time-varying zero-frequency phase intercept.

Envelope delay distortion is related to the differences in transmission time for the various voiceband frequencies over a given line. Such differences in delay will produce intersymbol interference in many data signals (see 8.1).

3.4.2 Round-trip envelope delay (round-trip delay)

Round-trip envelope delay is the envelope delay measurement at a single frequency, with the back-to-back envelope delay set to 0.00 ms. Because of the 83.33 Hz modulation frequency for envelope delay measurements, round-trip delays from 0.0 to 12.0 ms will be displayed with modulo 12 arithmetic. (See 8.2.)

3.4.3 Extended envelope delay

Shifting the envelope delay modulation frequency of 83.33 Hz by 8.333 Hz permits unambiguous resolution of the round-trip envelope delay to $1/8.333 \text{ Hz} = 120 \text{ ms}$ by determining the 12 ms increment in which the more accurate 0–12 ms normal round-trip envelope delay lies. If the delay is known to be greater than this 120 ms (domestic) range, then a second range-extending measurement can be made. The envelope delay modulation frequency of 83.33 Hz is shifted by 0.8333 Hz to permit unambiguous resolution of the true round-trip envelope delay to $1/0.8333 \text{ Hz} = 1200 \text{ ms}$ by determining the 120 ms increment in which the 0–120 ms domestic round-trip envelope delay lies. (See 8.3.)

3.4.4 Return loss

A ratio of the transmitted power to the power reflected back to the originating end of a channel. Return loss measurements are most important for mixed 4-wire/2-wire systems providing two-way transmission. Such systems are subject to line echoes. If the return loss is low, large talker or listener echoes, or both, may occur creating what is referred to as the rain barrel effect in voice communications and may also cause mutilation of some data signals. (See 8.4.)

3.4.5 Intermodulation distortion (IMD)

The four-tone IMD test technique from IEEE Std 743-1984 has been replaced by the 23-tone IMD technique. (See 8.5.)

3.4.6 23-tone measurements

The 23-tone test signal consists of 23 equally-spaced tones from 203.125 Hz to 3640.625 Hz with known phase relationships. Since the phases of the 23 tones are known, straightaway envelope delay distortion measurements may be made at 22 frequencies. At the same time, the frequency response (or attenuation distortion) at the 23 transmitted frequencies may be measured along with other measurements. (See 8.6.)

3.4.7 Echo Path Loss and (round trip) Echo Path Delay

A measure of power reflected back, and the time it takes for that power to be returned to an observation point of a connection due to one or more 4-wire to 2-wire impedance mismatches along the connection.

Echo path loss (EPL) and Echo path delay (EPD) are important measurements for the isolation of problems caused by one or more 4-wire to 2-wire impedance mismatches along a communications path by determining the power returned and the distance away (in time) from the observation point for each echo generation point. High-speed voiceband modems employing echo cancellation techniques are particularly sensitive to multiple echoes. In addition, EPL and EPD in a weighted combination with end-to-end loss and noise can be used to assess the voice quality on a “grade-of-service” basis for that connection. (See 8.7.)

3.4.8 Network impulse response measurement system

A method for determining the S/TD utilizing estimates of the Network Impulse Response (NIR) is developed around the principles of Adaptive Finite Impulse Response (AFIR) filters. This technique effectively models the linear characteristics of a transmission channel. Knowledge of the channel impulse response provides a means to not only determine the Signal-to-Total Distortion Ratio of the channel, but also yields the straight away attenuation distortion and group delay characteristics of the channel under test. (See 8.8.)

3.4.9 Active speech level

Active speech level is an objective measure of the average speech power (rms level) of the active portion of the speech. The active part of the speech includes those short periods of low energy that a listener perceives as part of the speech, but excludes pauses. (See 8.9.)

3.4.10 Peak-to-average ratio

P/AR is a measure of intersymbol interference in voiceband data signals. P/AR was designed to evaluate the simultaneous effects of envelope delay distortion, bandwidth reduction, and poor return loss (gain and phase ripples) on high-speed data transmission, but is unable to separate these effects. The 23-tone measurement capability of 8.6.3 of this standard and the echo path loss capability of 8.7 of this standard make P/AR unnecessary, as explained in annex E.

4. Digitally-encoded analog measurements

4.1 Introduction

Requirements and discussions are grouped according to the type of transmitted test signal as well as by the type of measurements made, e.g., a sinusoidal tone, a quiet termination, transients, and multiple-tone test signals for the measurement of various transmission parameters.

4.2 Digitally-encoded analog signals

This standard provides for measurement of certain transmission characteristics and impairments of analog signals that have been embedded in digital bit streams by sampling, quantizing, and encoding.

4.2.1 Characteristics specific to digitally encoded analog signals

Analog signals are band limited to frequencies below one half the sampling rate to minimize aliasing. Noise floors, signal-to-noise ratios, and signal ranges of digitally encoded analog signals are restricted by the quantizing noise associated with the development of the PCM signal. Such restrictions are treated individually in appropriate clauses of the standard.

Certain parameters specified for analog implementations require modification in order to meet the constraints imposed by the digitization of the analog signal. The specification of the C-Message filter, for example, includes frequencies beyond half the sampling rate and the filter will be subject to aliasing of its response if implemented digitally as defined in a system with an 8000 per second sampling rate.

4.2.2 DS0 signal format

A DS0 signal is a 64 kbit/s digital signal formed by sampling and encoding an analog signal which, has been band-limited to between 0 and 4 kHz, using a sampling rate of 8000 samples per second. The DS0 signal conforms to the requirements of ANSI T1.403-1995, and IEEE Std 1007-1991.

4.2.2.1 Transfer function

The transfer function of the DS0 encoder shall conform to a piece wise linear approximation to a $\mu=255$ logarithmic companding law, as described in tables 4 and 5 of IEEE Std 1007-1991.

Note that the piece wise linear approximation referenced above results in a logically inverted sign-magnitude representation of the encoded level so that the magnitude bits of the highest representable level are 0000000. The same seven bits of the corresponding lowest level are 1111111. The codes for positive levels are prefixed by a sign bit with a logic value one and negative code levels are prefixed with a zero. The logical inversion described above provides a high ones density for low-level signals or for idle channels, a desirable condition for timing recovery from digital signals traversing a repeatered network.

4.2.2.2 Transmission level point and the digital milliwatt

The zero transmission level point (0 TLP) is the level produced by the digital milliwatt in 600 Ω as defined in Appendix A of IEEE Std 1007-1991. The digital milliwatt is transmitted by repetition of the eight-frame pattern shown in table 1 at the 8000-Hz sampling rate.

Table 1—The digital milliwatt

Bit →	± Bit position								Voltage in V
	1	2	3	4	5	6	7	8	
Frame									
1	0	0	0	1	1	1	1	0	−0.427
2	0	0	0	0	1	0	1	1	−1.009
3	0	0	0	0	1	0	1	1	−1.009
4	0	0	0	1	1	1	1	0	−0.427
5	1	0	0	1	1	1	1	0	0.427
6	1	0	0	0	1	0	1	1	1.009
7	1	0	0	0	1	0	1	1	1.009
8	1	0	0	1	1	1	1	0	0.427

The digital milliwatt is not used on D4-formatted facilities because its bit patterns in bit positions 4, 6, and 8 mimic the framing bit pattern used for resynchronization of DS1 facilities. This false framing problem does not exist on DS1 facilities with the extended superframe format (ESF). A sinusoid 3.17 dB higher in amplitude than the digital milliwatt is the highest level signal that can be encoded without waveform clipping distortion.

4.2.2.3 Robbed-bit signaling

DS1-type terminal equipment commonly uses a technique called “robbed-bit signaling” to multiplex both digitally-encoded analog information and a low-speed signaling channel into the same DS0 channel octet. The effect on the digitally-encoded analog signal is that the least significant bit of the octet in every sixth frame is “robbed” to form the signaling channel, thus reducing the accuracy of the DS0 representation of the encoded analog from 8 bits to 7 bits in every sixth frame. The measuring techniques described in this standard are applicable to DS0 signals transporting analog information with or without robbed-bit signaling. Transmission impairments associated with robbed-bit signaling are discussed in [B6] and [B7] of the bibliography in annex J.

4.2.2.4 Effects of μ -law quantization

An ideal digital encoding of analog signals for telecommunications would be logarithmic, which would permit a constant Signal-to-Noise ratio (S/N) for a broad range of input signal levels. This ideal logarithmic characteristic is approximated in the μ -law encoding by eight piece-wise linear segments for positive voltages, and eight piece-wise linear segments for negative voltages (with the two linear segments at 0 V joined in a common segment). The encoding in the middle segment around 0 V is linear, so the S/N tracks the voltage level in that segment. The resulting 15-segment μ -law S/N curves have S/N ratios, which improve from 30 dB at -45 dBm input levels to ≈ 36 dB at -10 dBm for both the sine wave and 23-tone waveforms. For a sine wave input signal, a sharp break in the S/N occurs at $+3.17$ dBm (1.578 V in 600 Ω), which is the highest encodable 8-bit level. Since the sine wave spends most of the time near its peak value, the break in the S/N is very abrupt as the sine wave level is increased. For two sine waves of equal power, the clipping would occur at ≈ 0 dBm. For a Gaussian noise driven signal (or 23-tone signal), with its higher peak-to-rms ratio, the clipping starts at -5 dBm, with a less abrupt loss in the S/N. Voice is often approximated with an even more sharply peaked Laplacian signal, which has a more rounded S/N corner at -9 dBm.

The most significant bit of the 8-bit, μ -law word is the *sign* bit. The next three bits code the segment number. The four least significant bits are coded into $2^4 = 16$ linear steps for each segment. This logarithmic approximation is made by coding each successive segment with a linear step size twice that of the previous segment. Thus, raising the peak value of a sine wave slightly from the highest linear step in one segment to the first step in the next segment will cause the step size (in Volts) to double. This doubling in step size causes the S/N to dip somewhat at each segment transition, as the signal power is raised. It is this same phenomenon that causes a μ -law encoded sine wave to have more amplitude jitter than a Gaussian noise signal of equal power, and slightly less phase jitter than the Gaussian noise signal of equal power. Harmonic distortion of a sine wave in a μ -law encoder is typically down more than 60 dB.

4.2.2.5 Controlled frame slips

In the U.S., the 8-bit samples of 24 DS0 channels are combined in a digital channel bank with a single framing bit to form a $1/8000$ Hz = 125 ms frame. A hierarchy of stable clocks and derived clocks permit synchronization of the DS1 bit streams into the end offices. However, if the clock of a digital bank feeding the $[8(24) + 1]/8000 = 1.544$ Mb/s DS1 bit stream to a digital switch, digital cross-connect system, or echo canceler (such devices have a common classification of source-sink) is not synchronized (loop timed) to the clock of the source-sink, then a periodic controlled “frame slip” occurs. The frame slip is either a repetition or deletion of one 125 ms frame, depending on which clock is faster. Between the frame slips, the digital switch will gradually fill (or empty) a 193-bit buffer until it reaches overflow (or underflow), and a slip is required. The digital switch forces the slip on the *incoming* side of the digital switch, as shown in figure 1, and then correctly frames all the output PCM bit streams. This correct output framing effectively blocks all downstream *digital* detection of the slip based on frame structure. Even the newer ESF, with its additional performance monitoring through the cyclic redundancy checks (CRC), does not identify slips. The frame slips occur on the *input* side of the digital switch and the ESF CRCs are computed on the *output* side of the switch, so a frame slip will not produce a CRC error.

If a 1020 Hz holding tone were applied to the connection during the controlled frame slips, the slip itself would appear to be a positive or negative phase jump of $125(10^{-6})(1020)360^\circ = 45.9^\circ$. This indelible phase signature on the holding tone could be used to detect the slip as a phase hit greater than 40° (see 7.4.2). The phase of the holding tone would appear to roll uniformly during the period between the slips, and is the equivalent of a timing offset.

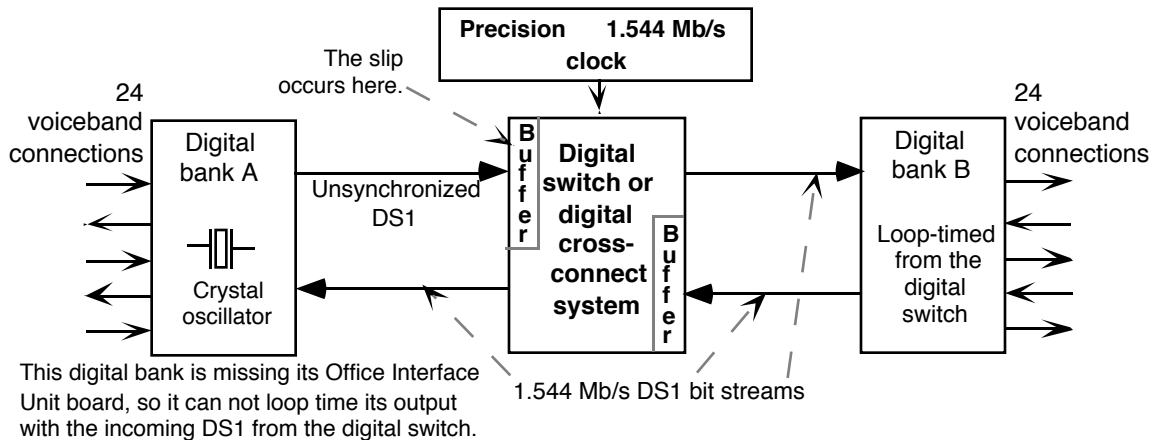


Figure 1—Controlled frame slip at digital switch

4.2.3 DS1 or DS3 digital bit stream access

Digital bit stream access for the IEEE Std 743 test set, which has DS0 transmit and receive capability, can be provided by incorporating a suitable DS1 digital drop and insert capability. This combined DS1 drop and insert test set provides for DS1 digital bit stream access, DS0 bit stream transmit and receive capability, synchronization, and other capabilities as detailed in IEEE Std 1007-1991. The normal DS1 bit stream is a bipolar, alternate mark inversion, return-to-zero format, with a 3-V peak-to-zero amplitude. The balanced impedance for the DS1 bit stream is $100\ \Omega \pm 10\%$.

The combined DS1 drop and insert function and the IEEE Std 743 test set function are shown in figure 2. A test set manufacturer could also provide an entire DS3-DS1-DS0 drop and insert capability in the same set with the IEEE Std 743 test circuitry. IEEE Std 1007 provides the requirements for the DS1 and DS3 digital access.

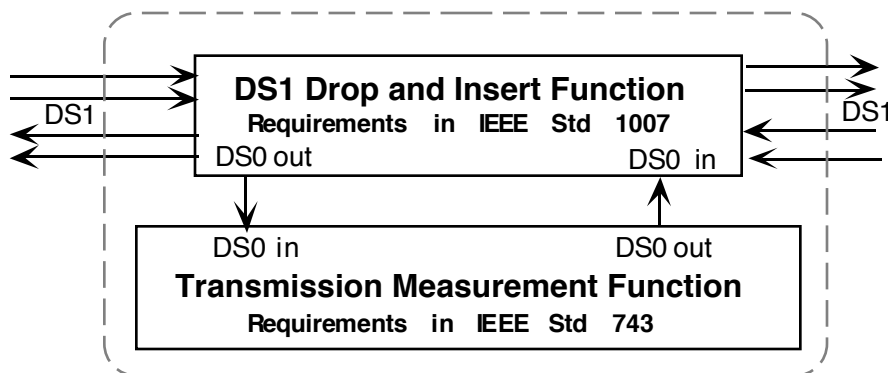


Figure 2—Test set with combined DS1 bit stream access and IEEE Std 743 measurement capabilities

Not shown in the figure are the possible *analog* input and output leads of the IEEE Std 743 test set. Some DS1 drop and insert test sets also provide for a DS0 bit clock and a DS0 byte clock in addition to the DS0 input and DS0 output leads. An IEEE Std 743 test set with DS0 transmit and receive capabilities, but no DS1 access capabilities, could perform IEEE Std 743 tests with such a DS1 drop and insert set as detailed in annex I.

4.2.4 Digital output to digital input test

For DS0 digital signal transmission, equivalent analog test signals shall be encoded by computations from the $\mu=255$ transfer function (see 4.2.2.1), and free of any constraints of analog filters or output transformers. The Digital Reference Sequence (DRS) of 5.3.3 is an example of a digital encoding of a sinusoidal holding tone (at 1013.8 Hz). Decoding or encoding of a received DS0 signal shall be done by calculation from the $\mu=255$ transfer function, and measurement algorithms shall permit displayed results free of analog anti-aliasing filter and transformer input constraints. Thus, for transmitted digital-equivalent 0 dBm tones between 20 Hz and 3990 Hz from a transmitter, the displayed received level shall be within 0.2 dB of that computed for the DRS tone (see 5.3.3).

4.3 Echo canceler disabling

Echo cancelers (and echo suppressors) may be disabled for a measurement sequence by sending a 2100 Hz tone for a minimum of 1350 ms with phase reversals, as described in ITU-T Recommendation O.22 (10/93). To avoid enabling an echo canceler once it has been disabled, there should be no quiet period during the measurement sequences longer than the echo canceler re-enabling interval, as described in the CCITT Blue Book Recommendation V.25 (1988).

4.3.1 Echo canceler disabling tone sequence

A 2100 Hz (± 8 Hz) tone at a level of -12 dBm0 (± 1 dB) should be transmitted for a period of 450 ms (± 25 ms) followed immediately by a 450 ms (± 25 ms) period of the tone with a 180° ($\pm 5^\circ$) phase reversal, followed immediately with a 450 ms (± 25 ms) period of the tone with its original phase.

4.3.2 Echo control devices re-enabling interval

Echo control devices may be re-enabled after the echo control disabling tone sequence (see 4.3.1) has been sent if there is a quiet interval in both directions of transmission for at least 75 ms (± 20 ms). Echo control devices will be re-enabled if there is a quiet interval in both directions of transmission for at least 200 ms (± 20 ms).

5. Sinusoidal signal generators

5.1 Tone generators, common requirements

The variable-frequency generator is distinguished from the fixed-frequency holding tone generator. There are certain common requirements for both generators.

5.1.1 Output level

The required output level range is $+10$ dBm to -40 dBm into an appropriate (see 10.1) resistive termination. The output level setability over this range shall be 0.1 dB. At a test frequency of 1 kHz, the output level shall be accurate to ± 0.1 dB between 0 dBm and -19 dBm and accurate to ± 0.2 dB, elsewhere. A signal-off capa-

bility, while maintaining a constant output impedance, is desirable to provide a quiet termination for noise measurement. Flatness with frequency is covered in 5.2.3.

For generators whose output is a PCM encoded signal, the required output level range shall be + 3 dBm to -40 dBm. This level shall be verified using rms detection calibrated to the 1013.8 Hz digital reference sequence (see 5.3.3). The requirements for setability and accuracy shall be the same as stated above. The digital idle (quiet termination) code shall be seven 1's followed by a 0 in the eighth bit (least significant bit) position (see [B2], Section 6.2). The all 1's code is reserved for the Alarm Indication Signal code.

5.1.2 Frequency stability

The frequency should not deviate by more than 0.1% in any 1 h period after an initial warm-up of 5 min while still meeting the other requirements for generator frequency accuracy.

5.1.3 Level stability

The level should not vary by more than 0.05 dB in any 1 h period after an initial warm-up of 5 min.

5.1.4 Total distortion

The total distortion including harmonics, noise, and spurious tones, as measured on a distortion analyzer that eliminates only the fundamental tone, shall be at least 40 dB below the output level over the range of +10 dBm to -40 dBm.

5.1.5 Background noise

If generator levels below -40 dBm are possible, the distortion requirements in 5.1.4 shall be changed consistent with a noise floor of -80 dBm.

5.1.6 Level display

The set shall display the output level which would be dissipated in an appropriate (see 10.1.1) resistive termination. The displayed level shall not be affected by any impedance connected.

5.2 Variable frequency generators

5.2.1 Frequency accuracy

The accuracy requirement for the generator frequency is ± 1 Hz from 50 Hz to 9999 Hz (or to the highest advertised frequency below 9999 Hz) and accurate to ± 10 Hz from 10 kHz to 100 kHz. For oscillators with output frequencies above 100 kHz, the accuracy shall be ± 100 ppm.

Adjustment resolution is related to the frequency range as shown in table 2.

Table 2—Frequency adjustability vs. frequency range

Frequency range (kHz)	Adjustable in increments of not more than (Hz)
< 4	4
4 to 10	10
10 to 100	100
> 100	1000

5.2.2 Frequency range

The generator shall have a frequency range of at least 50 Hz to 3900 Hz.

5.2.3 Flatness

The properly terminated output of the generator shall be flat to within ± 0.1 dB (or less) over the frequency range of 200 Hz to 15 kHz (or to the top of the frequency range, if less than 15 kHz). As the frequency is varied at any output level setting, the maximum and minimum level indications shall not differ by more than 0.2 dB. For frequencies above 15 kHz, a total level excursion of 0.5 dB or less shall be met.

5.3 Holding tone generator

The holding tone generator finds most frequent use in the measurement of tone- or signal-to-total-distortion ratio (see 6.2.13), impulse noise with tone, phase hits, gain hits, and dropouts (see 7.6), phase jitter (see 6.3), and amplitude jitter (see 6.4). Tone sources of 1 kHz should meet these requirements.

5.3.1 Frequency range

Acceptable frequencies may range from 1002 Hz to 1022 Hz. Care should be exercised that the tone frequency does not drift below 1002 Hz as a result of the permissible frequency drift. The recommended holding tone frequencies are in the range of 1020 Hz ± 2 , -7 Hz (see CCITT Blue Book Recommendation O.6 [11/88]).

5.3.2 Spurious noise

Any generator used as a source of holding tone shall have:

- 0.2° phase jitter or less
- 0.2% amplitude jitter or less
- No impulse noise at a level 20 dB below the power of the tone
- No phase hits greater than 3°
- No gain hits greater than 0.5 dB
- A S/TD of at least 50 dB

All of these tests may be performed using equipment defined in this standard, with d) and e) requiring the addition of a controlled amount of phase or amplitude modulation, or both.

5.3.3 Digital Reference Sequence (DRS)

A 797-byte DRS defines a 1013.8 Hz sine wave for analog testing and provides a specific digital sequence for bit error ratio testing. The DRS frequency to ten figures is $101(8000/797) = 1013.801757$ Hz. The DRS encoding formulas are given in table 3, where the constant $K = 2\pi(101/797)$ and N is the byte number for an 8000 Hz sample rate.

Table 3—Formulas for μ -law ($\mu=255$) and A-law ($A=87.6$) encoding of the 1013.8-Hz DRS

Companding law	Tone level	Formula for $V(N)$
μ	0 dBm	$8159(10^{-3.17/20})\sin(K[N-1])$
μ	−10 dBm	$8159(10^{-13.17/20})\sin(K[N-1])$
μ	−13 dBm	$8159(10^{-16.17/20})\sin(K[N-1])$
A	0 dBm	$4096(10^{-3.14/20})\sin(K[N-1])$
A	−10 dBm	$4096(10^{-13.14/20})\sin(K[N-1])$
A	−13 dBm	$4096(10^{-16.14/20})\sin(K[N-1])$

The 797 codes generated from the 0 dBm, μ -law ($\mu=255$) DRS formula of table 3 are listed in annex B for verification purposes.

6. Steady-state measurements

6.1 Level and attenuation distortion

The received level is the power dissipated in a an appropriate resistive termination by the received tone. The received level is expressed in dBm (dB relative to 1 mW) as follows:

$$\text{Received level in dBm} = 10 \log (10^3 P)$$

where

P is the power in watts of the received signal delivered to a standard impedance i.e., 1 mW = 0 dBm

For level measurements made directly from a PCM-encoded signal, the received level is expressed in dBm (dB relative to the rms level of the digital reference sequence).

The loss (more correctly the transducer loss) of the circuit is expressed in dB. It is the difference between the power available from the source that is driving the circuit under test, (the power the source would deliver to a standard matched impedance [e.g., 600 Ω or 900 Ω], also called the transmitted level), and the received level.

The transmission level at any point in a transmission circuit or system is the ratio, expressed in dB, of the power of a signal at that point to the power of the same signal at a reference point called the zero transmission level point (0 TLP). Stated another way, the transmission level at any point on a circuit is defined as the design gain or loss, expressed in dB, between that point and an arbitrary point called the zero transmission level point (0 TLP). By design, certain points in telecommunications systems have fixed TLPs. Transmission level points for voice frequency and data circuits at analog points in telecommunications systems typically range from +13 TLP to −19 TLP.

Table 4 shows the relationship between TLPs and power level in dBm. The TLP will also offset the expected value for noise measurements. The term dBm₀ (dBm at 0 TLP) is used to express the test signal power at any point on a circuit referred to 0 TLP.

Table 4—Measured power levels in dBm for test signals at various TLPs

Nominal test signal power in dBm	Transmission level points (TLP) in dB							
	+13	+7	0	−2	−3	−4	−16	−19
+7	+20 dBm	+14	+7	+5	+4	+3	−9	−12
0	+13 dBm	+7	0	−2	−3	−4	−16	−19
−10	+3 dBm	−3	−10	−12	−13	−14	−26	−29
−13	0 dBm	−6	−13	−15	−16	−17	−29	−32
−16	−3 dBm	−9	−16	−18	−19	−20	−32	−35

Linearity or tracking refers to loss measurements made at a variety of transmitted levels.

6.1.1 Gain slope

The measurement of loss or received level vs. frequency while the transmitted level is held constant is referred to as frequency response or attenuation distortion. The specific term gain-slope refers to the measurement of received level at 400 (404) Hz, 1000 (1004 or 1013.8) Hz, and 2800 (2804) Hz and calculation of the difference between levels at 404 Hz and 1004 Hz for 404 Hz gain-slope and the difference between levels at 2804 Hz and 1004 Hz for 2804 Hz gain-slope. Conventionally, + means more loss and − means less loss than the 1004 Hz value. The nominal 4 Hz offset prevents beating problems in level measurements on PCM systems as a result of the test frequencies being small submultiples of the 8 kHz sampling rate.

6.1.2 Level measurement

6.1.2.1 Range

At an analog interface, the measurement range shall be from +24.4 dBm_{100Ω} to −49.9 dBm_{100Ω}; from +23.1 dBm_{135Ω} to −49.9 dBm_{135Ω}; from +16.6 dBm_{600Ω} to −49.9 dBm_{600Ω}; and from +14.9 dBm_{900Ω} to −49.9 dBm_{900Ω}.

At a digital interface, the measurement range of the analog equivalent of PCM encoded signals shall be from +3 dBm_{600Ω} to −40 dBm_{600Ω}.

6.1.2.2 Accuracy

The level measuring set shall meet the accuracy requirements shown in table 5 for its specified frequency range, over the specified level ranges specified in 6.1.2.1.

Table 5—Level measuring accuracy requirements

Frequency range (Hz)	Accuracy (dB)
20 to 200	± 0.5
200 to 15 000	± 0.2
15 000 to $f_{\max}$	± 0.5
1002 to 1020	± 0.1 (0 to -19 dBm)

$f_{\max}$ = specified maximum frequency of the level measurement

For auto-ranging detectors, the accuracy shall be checked near auto-ranging points as the input signal is being lowered and raised through the auto-range points.

For measurements made directly from the PCM-encoded signal, the accuracy shall be as follows:

200 Hz to 3900 Hz: ± 0.2 dB (+ 3 dBm to -40 dBm)

1002 Hz to 1022 Hz: ± 0.1 dB (0 to -19 dBm)

6.1.2.3 Bridging

The test set shall clearly indicate the impedance for which the bridging indication is calibrated. In addition to meeting the bridging impedance requirements of 10.2, the level displayed on the set shall be compensated to be within 0.1 dB of that which would have been observed if the bridging impedance were infinite. All other requirements of 6.1 apply to the set in the bridging mode.

6.1.2.4 Level of data signals

With appropriate filtering (D Filter), voiceband data modem signal power can be measured with the level detector. The E Filter may be used for level measurement of ISDN line termination signals or noise on an ISDN subscriber line. The activating signals for the ISDN digital subscriber line (DSL) network terminations (which must be provided either in the IEEE Std 743 test set or by external equipment) are listed in annex F. The F Filter may be used for level measurement of the line termination for High-bit-rate Digital Subscriber Lines (HDSL). The G Filter may be used for level measurement of the line termination for Asymmetrical Digital Subscriber Lines (ADSL). Requirements for these filters may be found in clause 9.

6.1.2.5 Detector

An average detector shall be available for level detection, since it is 2 dB to 3 dB less sensitive to interfering noise than rms detection. The displayed indication of the average detector shall be internally adjusted to read the rms power of the sine wave. True rms level detection shall also be provided by selection, and is the preferred type of detector for non-sinusoidal signals or when the waveform of the signal being measured is unknown. There will be less than 0.05 dB difference in measured sine wave level between the two detectors if the T/TD is at least 19 dB.

An average detector will read 0.5 dB lower than a true rms detector on a 50% amplitude modulated test signal (EDD line signal; see 8.1.1.1). An average detector will read 1 dB lower than a true rms detector on a 23-tone test signal (see 8.6). Annex G shows the measurement differences between average and rms detectors for various well-defined waveforms.

6.1.2.6 Display response time

For a suddenly applied 1 kHz sine wave, the display should be within ± 0.1 dB of the final indication within 3 s. This requirement shall be met for a suddenly-applied signal consisting of a 0 dBm 1 kHz tone with a 50 V dc bias. Further, the display should not fluctuate by more than 0.1 dB when measuring a constant amplitude 20 Hz sine wave over a range of + 3 dBm to -40 dBm.

6.1.2.7 Crosstalk

Care shall be taken in test sets that contain a generator and a level detector to ensure that the generator of the test set does not crosstalk into the level detector at a level that degrades level measurement accuracy beyond the limits stated in 6.1.2.2.

6.2 Noise and signal-to-total distortion ratio

This subclause covers the requirements for the measurement of noise and the requirements for a test set that measures S/TD ratio automatically (see 6.2.13).

Noise power is measured in dBm referenced to 1 pW, as given by the formula

$$N \text{ (in dBm)} = 10 \log (P \times 10^{12})$$

where

- N is displayed noise indication
- P is received noise power (W)

For example, if $P = 1 \mu\text{W}$, then $N = 60$ dBm (-30 dBm).

The unit symbol dBmC (dBm C-Message) is used for noise measurement with C-Message (see 9.2) or C-Notch weighting (see 9.3). Analog channel measurements of digitally-encoded (PCM) systems require the use of a holding tone in order to exercise the quantizer in its normal operating region.

Noise may be measured either with or without a holding tone present. If no holding tone is present, the background noise, or idle channel noise, is measured to a far end termination, which normally consists of a nominal 600 Ω to 900 Ω resistive termination. The rms noise power is measured through an appropriate weighting network. If a holding tone is used, the noise-with-tone measurement is made with a generator having a 600 Ω or 900 Ω output impedance, an output frequency between 1002 Hz and 1022 Hz, and an output level appropriate for the system under test. The rms noise power is normally measured with a weighting network after rejection of the holding tone with a sharp notch filter (notched noise measurement). The displayed indication for notched noise is the summation of noise on the circuit that is not a function of the holding tone plus any harmonic distortion or quantizing noise that is a result of the holding tone. Amplitude or phase modulation sidebands may fall outside of the notch filter rejection band and thus contribute to the indication.

Noise-to-ground measurements permit evaluation of the longitudinal (common mode) voltage present on physical plant. There is no need to make noise-to-ground measurements with a longitudinal holding tone or notch filter present. Noise-to-ground measurements are not normally made where a low impedance to ground is present.

6.2.1 Accuracy

The noise measurement shall be accurate to ± 1 dB. The noise-to-ground measurement shall be accurate to ± 1.5 dB. These accuracies shall be verified for both sinusoidal and random noise inputs. The accuracy of a C-Notch noise measurement when a holding tone is present is covered in 6.1.2.2.

6.2.2 Filters

The D, E, F, and G Filters are also used for the measurement of background noise and impulse noise. These filters are defined in clause 9. The phase response of these filters, which is set by defining the type of filter, is specified to assure repeatable impulse noise measurements between test sets by different manufacturers. For measuring noise interference to voice, the C-Message filter is used. When a 1 kHz holding tone is used on a circuit, a 50 dB notch filter is used to remove its energy for the measurement of signal-to-noise-ratio.

6.2.3 Ranges

The required noise measurement ranges are shown in table 6.

Table 6—Noise measurement ranges

Filter	Noise (dBrn _{100; 135; 600; 900})	Impedance in Ω	Noise-to-ground (dBrn ₆₀₀)	Noise (dBrn ₆₀₀)
	Analog interface		Analog interface	Digital interface
C-Message weighting	10 to 90	600/900	40 to 130	22 to 90
C-Notched	10 to 90	600/900	Not applicable	22 to 90
3.4 kHz Flat	10 to 90	600/900	40 to 130	22 to 90
3.4 kHz Flat—notched	10 to 90	600/900	Not applicable	22 to 90
D Filter	10 to 90	600/900	40 to 130	22 to 90
D-Notched	10 to 90	600/900	Not applicable	22 to 90
15 kHz flat	10 to 90	600/900	40 to 130	Not applicable
E Filter	10 to 90	135	40 to 130	Not applicable
F Filter	20 to 90	135	50 to 130	Not applicable
G Filter	30 to 90	100	60 to 130	Not applicable

See table 11 for the corresponding impulse noise threshold ranges for the various filters.

6.2.4 Noise-to-ground input configuration

There shall be no dc path to ground for a 200 V dc longitudinal signal of either polarity. The noise-to-ground input impedance configuration shall be as shown in figure 3.

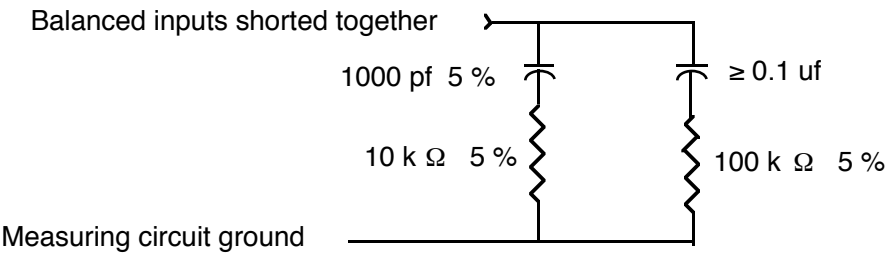


Figure 3—Input impedance for noise-to-ground measurement

6.2.5 Detector

The detector circuit shall measure the true rms value of the noise. An approximate, or full-wave quasi-rms detector circuit should be used if its output does not differ from a true rms detector by more than ±0.5 dB for the following input signals:

- a) Random noise
- b) Sine wave
- c) Two non-harmonically related sine waves of equal level, and at least 100 Hz apart.
- d) Gated bursts of a 1.8 kHz sine wave, at a 50 Hz rate, with 20% of the cycle at full amplitude and 80% of the cycle down 8.4 dB from the full amplitude. An rms detector would indicate a drop in level of 5.0 dB compared to the full amplitude sine wave for this case. The 8.4 dB drop in level should be chosen to avoid auto-ranging points.

6.2.6 Measurement averaging time

The response time for the detector and indicating means should meet the following limits below. With the application of bursts of 1.8 kHz tone to the input of the set, gated at a duty cycle of 50%, with half the cycle at full amplitude and the other half down 8.4 dB from full amplitude, the meter or digital display device shall indicate a variation as shown below. The levels should be chosen to avoid auto-ranging points.

Gated burst tests	
Gating frequency (Hz)	Peak-to-peak indicator variation (dB)
10	≤1
2	≥3

For a set having a digital display, it is permissible to adjust the total input power with a 1 dB vernier control to a point where the digital display does not change so as to pass the *less than* 1 dB requirement.

6.2.7 Damped response

If a set has a DAMPED display capability, then in addition to the requirements of 6.2.6, which shall be met in the NORMAL mode, the following requirement should be met with DAMPED mode: The damping factor requirement is determined by measuring the gating frequency as in 6.2.6, which just causes a 3 dB spread in the noise indications in first the NORMAL mode and then in the DAMPED mode. The ratio of these gating frequencies is the damping factor, which should lie between 2.4 and 4.8. For digital displays, the refresh rate should be slowed by a factor between 2.4 and 4.8.

6.2.8 Display response time

For a suddenly applied random noise signal 20 dB_{rn} or greater, the display should be within ± 1 dB of the final indication within 3 s. This requirement shall be met with a 50 V dc bias on the random noise signal.

6.2.9 Loss of holding tone

A nominal 1 kHz holding tone is necessary to provide more meaningful measurements of noise on certain facilities. For a noise-with-tone measurement, there should be an unmistakable indication if the holding tone level drops below -40 dBm or if the frequency is outside the range of 1002 Hz to 1022 Hz.

6.2.10 Holding tone level

The set shall accommodate received holding tone levels from 0 dBm to -40 dBm.

6.2.11 Crest factor

To not significantly impair the measurement of noise level, the set shall not compress signals that are at least 8 dB above the highest permissible displayed indication by more than 0.5 dB.

6.2.12 Turnover

With a 3.4 kHz Flat weighting filter in place, a rectangular waveform of 300 Hz should be applied to the input with a 20% duty cycle, and the noise indication should be noted. The input leads should be inverted. The new noise indication shall be within 1 dB of the first indication.

6.2.13 Signal-to-total-distortion ratio

6.2.13.1 Display

If the set measures S/TD ratio, then the ratio shall be displayed in decibels in a manner so that it is clearly distinguished from a noise measurement. The level and the noise shall be monitored separately and continually during the measurement period. Each displayed measurement shall be the result of a new measurement of the level (as per 6.1.2) and the C-Notched noise. The requirements of 6.2.1, 6.2.2, 6.2.9, 6.2.10, and 6.2.11 apply to the noise measurement. The requirements of 6.2.5, 6.2.6, and 6.2.10 apply, with the addition that a holding tone shall be added at the input to produce an in-range S/TD ratio. The set should continuously display the S/TD ratio in dB and the received sinusoidal tone level in dBm. The 23-tone signal permits measurement of signal-to-noise ratio as a separate measurement from S/TD ratio, which includes the effect of intermodulation distortion (see 8.6.5).

6.2.13.2 Range and accuracy

The set shall measure S/TD ratio over the range of 10 dB to 45 dB. The accuracy shall be ± 1 dB over the 10 dB to 40 dB range, and ± 2 dB or better over the 41 dB to 45 dB range.

6.2.13.3 Display response time

For a suddenly-applied random noise signal with holding tone, the indication should be within ± 1 dB of the final indication within 4 s.

6.3 Phase jitter

Phase jitter measurements indicate the cumulative effect of incidental phase modulation and additive tones or noise on the phase of a holding tone (see 5.3 for holding-tone requirements). The peak-to-peak deviations in phase of the received signal are detected after band limiting to reduce the effect of additive noise.

The most commonly found single-frequency components of phase jitter are 20 Hz (ringing current), 60 Hz (commercial power) and the second through fifth harmonics of these. Since the peak phase deviation caused by ac components of phase modulation rarely exceeds 0.2 radians (low-index phase modulation), only one pair of significant sidebands is produced for each sinusoidal component. Hence, a bandwidth of approximately 600 Hz centered about a carrier near 1 kHz suffices to recover the major suspected sinusoidal phase modulation without incurring large amounts of uncorrelated interference.

Since noise can cause what would appear to be a significant amount of phase jitter, a S/TD measurement should always be made in conjunction with phase jitter measurements. Also, because quantizing noise can cause a significant phase jitter indication, care is needed in the choice of the carrier frequency and in the filtering to suppress the effect of noise on the measurement.

Incorrect phase jitter indications (lower than true value) will result if a precise 1000 Hz tone (not a holding tone) is employed on a PCM system with an 8 kHz sampling rate. Instances of phase jitter in the region from 4 Hz to 20 Hz have been noted on some facilities. Since some data sets are affected by this low-frequency jitter, there may be a need for an option that permits testing in this range with equal frequency weighting in the total band of 4 Hz to 300 Hz.

6.3.1 Accuracy

The displayed peak-to-peak indication shall be accurate to $\pm 5\%$ of the measured value $\pm 0.2^\circ$.

6.3.2 Level range

The set shall accommodate the input signal levels below, and should provide the means to indicate when the received signal is outside of these ranges:

- a) Analog interface: -40 dBm to $+10$ dBm.
- b) Digital interface: -40 dBm to $+3$ dBm.

6.3.3 Frequency range

The set shall accommodate input frequencies from 1002 Hz to 1022 Hz. The set should give some indication when the signal is outside this range.

6.3.4 Display range

The set shall be capable of displaying phase jitter indications from 0.0° to at least 25.0° peak-to-peak.

6.3.5 Noise rejection

A band-limited white noise signal down 30 dB from a 1 kHz sine-wave carrier shall indicate less than 4° peak-to-peak jitter. For purposes of this test, the band-limited white noise signal shall be flat to within ± 1 dB from 250 Hz to 3.5 kHz with asymptotic slopes of -12 dB per octave. When a limiter is used in the detection process, this requirement typically dictates a bandpass filter in front of the limiter. For quantizing noise from a PCM system, the nonlinearities resulting from approximation with linear segments to the ideal logarithmic encoding characteristic will result in phase jitter approximately 20% lower than for white noise with the same signal-to-noise ratio.

6.3.6 Frequency weighting

After demodulation, the signal shall be weighted so that jitter components in the selected band are given equal weighting. The weighting characteristics may be measured by applying a pure 1 kHz tone at a 0 dBm level as the carrier and a second pure tone 20.0 dB lower in level as a source of phase jitter.

For purposes of this test, a pure tone is defined as a signal source that meets the requirements of a holding tone generator as specified in 5.3. Tables 7 and 8 give the jitter indication limits as a function of the difference in frequency between the two tones.

Table 7—Frequency weighting band of 20 Hz to 300 Hz

Difference frequency (Hz)	Phase jitter degrees Peak-to-peak (p-p)
2	< 1.0
5	< 3.0
10	< 8.0
20–240	10.8–12.2
300	10.0–12.2
500	< 3.0

Table 8—Frequency weighting band of 4 Hz to 300 Hz

Difference frequency (Hz)	Phase jitter degrees Peak-to-peak (p-p)
0.4	< 1.0
1	< 3.0
2	< 8.0
4	9.2–12.1
8–240	10.8–12.2
300	10.0–12.2
500	< 3.0

6.3.7 Single-frequency-interference

Apply a 1 kHz tone to the input of the set at a 0 dBm level. Add in a second tone down 20 dB in level and vary its frequency. For frequencies below 300 Hz, the displayed indication shall be less than 1°p-p. For frequencies between 1.5 kHz and 4 kHz, the displayed indication shall be less than 3°p-p. For frequencies above 4 kHz, the displayed indication shall be less than 0.5°p-p.

6.3.8 Level-to-phase conversion

With the test setup as in the previous test and the second tone at 1.1 kHz, change the composite level over the range of +10 dBm to −40 dBm without a change in the ratio of the test-tone and the second tone amplitudes. The spread of the indications shall not exceed 0.7°p-p.

6.3.9 Amplitude-to-phase conversion

Apply a 10% amplitude-modulated 1 kHz carrier to the input of the set at a 0 dBm level. As the modulation frequency is varied from 2 Hz to 900 Hz, the displayed jitter indication shall be less than 1°p-p.

6.3.10 Measurement averaging time

The detector and indicating circuits for the 20 Hz to 300 Hz frequency weighting should have response averaging time characteristics as follows:

Apply a gated sine-wave phase modulation on a steady 1 kHz carrier to the test set. The modulation signal shall be gated at a duty cycle of 50%. The indicator shall show a variation as shown below:

Gating frequency (Hz)	Indicator variation
5.0 Hz	$2(\max - \min)/(\max + \min) < 10\%$
1.0 Hz	$2(\max - \min)/(\max + \min) > 40\%$

6.3.11 Detector

6.3.11.1 Peak-to-peak

The detector shall be a true peak-to-peak detector. A half-wave peak or two-times peak detector shall not be used.

6.3.11.2 Response to white noise

The detector shall measure white noise at the 2.58 sigma (99%) point.

6.3.11.3 Two-tone response

A test for the quality of peak detection is as follows:

- Apply a 0 dBm, 1000 Hz carrier tone to the input of the set together with a second tone at 1100 Hz, which is adjusted in level to produce a phase-jitter indication of exactly 10.0°.
- Remove the 1100 Hz tone and add a tone at 1170 Hz with another generator that is adjusted in level to produce a phase-jitter indication of exactly 10.0°.

- c) Add the 1100 Hz tone, at its previously adjusted level, to the carrier plus the 1170 Hz tone. The phase-jitter indication shall lie between 18.5° and 19.8° .

6.3.12 Time to display correct indication

When the 20 Hz to 300 Hz weighting is used, the display should be within 0.7° p-p of the final indication in 6.3.6 at a difference frequency of 100 Hz, within 4 s of application of the complete test signal. When the 4 Hz to 300 Hz weighting is used, the time to display the correct indication should be within 25 s of application of the complete test signal.

6.3.13 Demodulated carrier

If the set employs a phase locked loop, the demodulated carrier from the phase locked loop should be made available for measurement of frequency shift over the facility. This carrier is relatively free from the interfering effects of noise on the facility.

6.3.14 Demodulated phase jitter

The test signal, after phase demodulation and frequency weighting, should be made available on an external jack to permit analysis of the jitter signal.

6.4 Amplitude jitter

Amplitude jitter measurements indicate the cumulative effect of incidental amplitude modulation and additive tones or noise on the envelope of a holding tone (see 5.3). Peak-to-peak deviations in the envelope of the received signal are detected after band limiting to reduce the effect of additive noise.

The most commonly found single-frequency components of amplitude jitter are 60 Hz (commercial power) and its second through fifth harmonics. A bandwidth of approximately 600 Hz centered about a carrier near 1 kHz suffices to recover the major suspected amplitude jitter without incurring large amounts of uncorrelated interference.

Since group delay distortion and frequency response of a channel can cause amplitude jitter to be created from phase jitter, and vice versa, it is recommended that amplitude jitter be measured in conjunction with phase jitter. Comparison of amplitude jitter measurements with phase jitter measurements can be an effective tool in the diagnosis of channel impairments.

Since noise can cause what would appear to be a significant amount of amplitude jitter, a T/TD measurement always should be made in conjunction with amplitude jitter measurements. Also, because quantizing noise can cause a significant jitter, care shall be exercised in the choice of the carrier frequency and in the filtering to suppress the effect of noise on the measurement.

Incorrect amplitude jitter indications (lower than true value) will result if a 1000 Hz carrier is employed on a PCM system with an 8 kHz sampling rate. Instances of amplitude jitter in the region from 4 Hz to 20 Hz have been noted on some facilities. Since some data sets may be affected by this low-frequency jitter, there may be a need for an option that permits testing in this range with equal frequency weighting in the total band of 4 Hz to 300 Hz.

6.4.1 Accuracy

The displayed indication shall be accurate to $\pm 5\%$ of the measured value $\pm 0.2\%$ peak.

6.4.2 Level range

The set shall accommodate the input signal levels below, and should provide the means to indicate when the received signal is outside of the measurement ranges.

- a) Analog interface: -40 dBm to $+10$ dBm.
- b) Digital interface: -40 dBm to $+3$ dBm.

6.4.3 Frequency range

The set shall accommodate input frequencies from 1002 Hz to 1022 Hz. The set should give some indication when the signal is outside of this range.

6.4.4 Display range

The set should be capable of displaying amplitude jitter indications from 0.0% peak to at least 25% peak.

6.4.5 Noise rejection

A band-limited white noise signal down 30 dB from a 1 kHz sine-wave carrier shall indicate less than 4% peak jitter. For this test, the band-limited white noise signal shall be flat to within ± 1 dB from 250 Hz to 3.5 kHz with asymptotic slopes of -12 dB per octave. For quantizing noise from a PCM system, the nonlinearities resulting from approximation with linear segments to the ideal logarithmic encoding characteristic will result in amplitude jitter approximately 10% higher than for white noise with the same signal-to-noise ratio.

6.4.6 Frequency weighting

After demodulation, the signal should be weighted so that jitter components in the selected band are given full weighting. The weighting characteristics may be measured by applying a pure 1 kHz tone at a 0 dBm level as the carrier and a second pure tone, 20.0 dB lower in level, as a source of amplitude jitter. For this test, a pure tone is defined as a signal source that meets the requirements of a holding tone generator as specified in 5.3. Tables 9 and 10 give the jitter indication limits as a function of the difference frequency between the two tones.

Table 9— Frequency weighting band of 20 Hz to 300 Hz

Difference frequency (Hz)	Amplitude jitter (% peak)
2	< 0.9
5	< 2.6
10	< 7.0
20–240	9.4–10.6
300	8.7–10.6
500	< 2.6

Table 10—Frequency weighting band of 4 Hz to 300 Hz

Difference frequency (Hz)	Amplitude jitter % peak
0.4	
< 0.9	
1	< 2.6
2	< 7.0
4	8.0–10.6
8–240	9.4–10.6
300	8.7–10.6
500	< 2.6

6.4.7 Single-frequency interference

Apply a 1 kHz tone to the input of the set at a 0 dBm level. Add in a second tone down 20 dB in level and vary its frequency. For frequencies below 300 Hz, the displayed indication shall be less than 0.9% peak, and for frequencies between 1.5 kHz and 4 kHz, the display indication shall be less than 2.6% peak. For frequencies above 4 kHz, the displayed indication shall be less than 0.5% peak.

6.4.8 Level-to-amplitude jitter conversion

With the test setup as in the previous test and the second tone at 1.1 kHz, change the composite level over the range of +10 dBm to –40 dBm without a change in the ratio of the test-tone and the second tone amplitudes. The spread of the indications shall not exceed 0.6% peak.

6.4.9 Phase-to-amplitude conversion

Apply an 11.5° peak-to-peak phase modulated 1 kHz carrier to the input of the set at a 0 dBm level. As the modulation frequency is varied from 2 Hz to 900 Hz, the displayed jitter indication shall be less than 0.9% peak.

6.4.10 Measurement averaging time

The detector and indicating circuits for the 20 Hz to 300 Hz frequency weighting should have response averaging time characteristics as follows:

Apply a gated sine-wave amplitude modulation on a steady 1 kHz carrier to the test set. The modulation signal should be gated at a duty cycle of 50%. The indicator should show a variation as follows:

Gating frequency (Hz)	Indicator variation
5.0	$2(\max - \min)/(\max + \min) < 10\%$ peak
1.0	$2(\max - \min)/(\max + \min) > 40\%$ peak

6.4.11 Detector

6.4.11.1 Peak-to-peak

The detector shall be a true peak-to-peak detector. The half-wave peak or two-times peak detector shall not be used.

6.4.11.2 Response to white noise

The detector shall measure white noise at the 2.58 sigma (99%) point.

6.4.11.3 Two-tone response

A test for the quality of peak detection is as follows:

- a) Apply a 0 dBm, 1000 Hz carrier tone to the input of the set together with a second tone at 1100 Hz, which is adjusted in level to produce an amplitude jitter indication of exactly 10.0%.
- b) Remove the tone at 1100 Hz and add a tone at 1170 Hz with another generator that is adjusted in level to produce an amplitude jitter indication of exactly 10.0%.
- c) Add the tone at 1100 Hz at its previously adjusted level, to the carrier plus the tone at 1170 Hz. The jitter indication shall lie between 18.5% and 19.8%.

6.4.12 Time to display correct indication

When the 20 Hz to 300 Hz weighting is used, the display should be within 0.6% peak of the final indication in 6.4.6 at a difference frequency of 100 Hz, within 4 s of application of the complete test signal. When the 4 Hz to 300 Hz weighting is used, the time to display the correct indication shall be within 25 s of application of the complete test signal.

6.4.13 Demodulated amplitude jitter

The test signal, after amplitude demodulation and frequency weighting, should be made available at an external jack to permit analysis of the jitter signal.

6.5 Frequency measurement

The frequency measurement capability of an instrument shall meet the following specifications. Requirements for an optional *frequency shift* measurement capability are added in 8.6.2.9 where they differ from the frequency measurement capability.

6.5.1 Level range

The set should have means to indicate when the received signal level is outside of the ranges below:

- a) Analog interface: -50 dBm to +10 dBm.
- b) Digital interface: -40 dBm to +3 dBm.

6.5.2 Frequency range

If the frequency measuring set is part of a larger test set, it shall cover the frequency range of that set.

6.5.3 Accuracy

The frequency measuring set shall be accurate to ± 1 Hz for frequencies up to 10 kHz. It shall be accurate to ± 10 Hz for frequencies above 10 kHz and up to 100 kHz. It shall display a frequency up to 9999 Hz with a resolution of 1 Hz.

6.5.4 Update rate

The frequency measuring set should display a new measurement at the rate of once per second. If the frequency measuring set is used to display the frequency of a manually-tuned generator to 1 Hz, it should display a new measurement at a rate of at least 4 times per second.

6.5.5 Interference and noise

The frequency measuring set shall meet the accuracy requirements in the presence of single-frequency interference or noise with a signal-to-interference or S/TD ratio of 20 dB. The single-frequency interference requirement shall be met for any interference frequency within the measurement range of the set. The noise requirement shall be met for band-limited white noise as follows:

- a) Voiceband instruments—noise bandwidth as defined in 8.1.2.8.
- b) Wideband instruments—noise bandwidth extending from 50 Hz to the highest frequency that the set can measure.

6.5.6 Display response time

The response time of the measuring set display is a function of the required accuracy.

- a) For a suddenly applied 1 kHz sine wave with a 20 dB signal-to-white noise (50 Hz to 4000 Hz) ratio, the display should be within the stated accuracy within
 - 1) 30 s for frequency shift measurement,
 - 2) 3 s for frequency measurement.
- b) For an instantaneous 100 Hz frequency shift of a sinusoidal tone, the display should be within the stated accuracy within
 - 1) 15 s for the frequency shift measurement,
 - 2) 2 s for frequency measurement.

7. Transient measurements

7.1 Description

This clause describes the measurement of two classes of transient phenomena in telecommunications circuits—impulse noise and rapid changes in phase or gain (including sudden loss of signal). Impulse noise is a large excursion of an input voltage signal level that is not correlated with the transmitted signal (if any) and is higher than the normal peaks of message circuit noise. A rapid change in phase is called a phase hit. A rapid change in received signal amplitude is called a gain hit. The phase or gain may stay at its new value or return to the original value some time later. A dropout occurs when the loss of the circuit increases by 12 dB or more compared to that at the start of the measuring interval.