TIA-912

TIA STANDARD

Telecommunications

IP Telephony Equipment

Voice Gateway Transmission Requirements

TIA-912

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TELECOMMUNICATIONS INDUSTRY ASSOCIATION

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Electronic Industries Alliance

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FOREWORD

(This foreword is not part of this standard.)

This document is a TIA/EIA Telecommunications Technical Standard produced by Subcommittee TR-41.4 of Committee TR-41. This standard was developed in accordance with TIA/EIA procedural guidelines, and represents the consensus position of the Subcommittee, which also served as the formulating group. This standard is based on TIA/EIA/TSB122-A and contributions to PN-3-3673 (to be published as ANSI/TIA/EIA-464-C) in TR-41.1.

The TR-41.4 IP Telephony Gateways Subcommittee acknowledges the contribution made by the following individuals in the development of this standard.

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There are five annexes in this standard.

Suggestions for improvement of this standard are welcome. They should be sent to:

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1 Introduction

1.1 General

Advances in digital voice communications over non-traditional channels such as the Internet and Local Area Networks have made it necessary to create a transmission standard for IP (Internet Protocol) Telephony systems. The term coined for these IP Telephony systems is “Voice Gateway”, which is the equivalent to the term PBX as used for traditional time-division-multiplexed (TDM) systems.

1.2 Purpose

This standard establishes performance and technical criteria for interfacing and connecting with the various elements of public and private telecommunications networks. Compliance with these requirements should assure quality service.

It is intended to be co-ordinated with the public network loss plan according to the principles of ANSI T1.508-1998 [Ref. 2] and to fully comply with TIA/EIA/IS-968 Standard [Ref. 3].

Voice quality-of-service issues such as the impact of transmission delay, speech compression and packet loss are addressed in TIA/EIA/TSB32-A [Ref. 4] and TIA/EIA/TSB116 [Ref. 5].

This standard was also developed in conjunction with ETSI TC STQ, who generated an equivalent half-channel loss plan standard for voice gateways that is harmonized with this standard [Ref. 15].

1.3 Categories of Performance Criteria

In accordance with EIA Engineering Publication EP-7-A, Style Manual for Standards and Publications of EIA, TIA, and JEDC, [Ref. 1], two categories of performance standards are specified, mandatory and advisory. The mandatory performance criteria are designated by the word "shall"; advisory are designated by the word "should”, "may”, or "desirable" (which are used interchangeably in this standard). The mandatory criteria generally apply to safety and protection, signaling and compatibility. They specify the absolute minimum acceptable performance levels in areas such as transmission and equipment parameters and durability.

Advisory or desirable criteria represent product goals. In some instances, these criteria are included in an effort to assure universal product compatibility even with equipment and facilities operational in statistically small quantities. In other cases, advisory criteria are presented when their attainment will enhance the general performance of the product in all its contemplated applications.

Where both a mandatory and an advisory level are specified for the same criterion, the advisory level represents a goal currently identifiable as having distinct compatibility or performance advantages, or both, toward which future designs should strive.

2 Scope

This standard covers the transmission requirements for voice gateways. For the purposes of this standard, a voice gateway is considered to be a device that performs routing functions between:

- Telephones
- Public and private network trunks
- IP based networks

Telephones considered in this standard consist of two different types:

- Analog telephones assumed to be compatible with the parameters specified in PN-3-4350.110 (to be published as ANSI/TIA/EIA-470-110-C [Ref. 6]).
- Digital telephones assumed to be compatible with the parameters specified in ANSI/TIA/EIA-810-A [Ref. 7].

2.1 Compliance Reference Point

The reference point for this standard is determined at the voice gateway interface boundaries and is not to be construed as a constraint on the internal coding or switching techniques of the voice gateway.

2.2 Compliance Interpretation

A voice gateway complies with this standard when it conforms to the requirements applicable to the interfaces with which it is equipped.

2.3 Regulatory Issues

This standard is intended to be in conformity with TIA/EIA/IS-968 regarding network harm, but is not limited to the scope of this standard. In the event that the IS-968 requirements are more stringent than those contained in this standard, the provisions of IS-968 apply.

3 References

The following standards and TSBs contain provisions, which are referenced in this document. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this document are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below. ANSI and TIA maintain registers of currently valid national standards published by them.

WARNING: This document contains references to the following works-in-progress which are subject to change:

PN-3-3673 (to be published as ANSI/TIA/EIA-464-C).

The most current version of PN-3-3673 is available in the public directory of TR-41.1 at:

http://ftp.tiaonline.org/tr-41/tr411/Public/Latest-Revision-of-PN3673/

PN-3-4350.110 (to be published as ANSI/TIA/EIA-470-110-C).

The most current version of PN-3-4350.110 is available in the public directory of TR-41.3.5 at:

http://ftp.tiaonline.org/tr-41/tr4135/Public/Latest-Revision-of-PN4350-110/

[1] EIA Engineering Publication EP-7-A, Style Manual for Standards and Publications of EIA, TIA, and JEDC

[2] ANSI T1.508 (1998), *Loss Plan for Evolving Digital Networks*.

[3] TIA/EIA/IS-968, *Connection of Terminal Equipment to the Telephone Network.*

[4] TIA/EIA/TSB32-A (December 1998), *Overall Transmission Plan Aspects for Telephony in a Private Network.*

[5] TIA/EIA/TSB116 (March 2001), *Voice Quality Recommendations for IP Telephony*.

[6] PN-3-4350.110 (to be published as ANSI/TIA/EIA-470-110-C), *Handset Acoustic Performance Requirements for Analog Terminal Equipment Connected to the Public Switched Telephone Network*.

[7] ANSI/TIA/EIA-810-A (December 2000), *Transmission Requirements for Narrowband Voice over IP and Voice over PCM Digital Wireline Telephones.*

[8] ITU-T Recommendation G.711 (11/88), *Pulse Code Modulation (PCM) of Voice Frequencies*.

[9] PN-3-3673 (to be published as ANSI/TIA/EIA-464-C), *Requirements for PBX switching Equipment.*

[10] ITU-T Recommendation G.122 *Influence Of National Systems on Stability and Talker Echo in International Connections*

[11] ANSI/IEEE Standard 743-1984 *Standard Methods and Equipment for Measuring the Transmission Characteristics of Analog Voice Frequency Circuits*

[12] ANSI/IEEE Standard 455-1985 *Standard Test Procedure for Measurement of Longitudinal Balance of Telephone Equipment in the Voice Band*

[13] TIA/EIA TSB31-B *Part 68 Rationale and Measurement Guidelines*

[14] ITU-T Recommendation G.107 (12/98) and (05/00), *The E-Model, A Computational Model for use in Transmission Planning*

[15] ETSI ES 202 020 *Harmonized Pan-European/North-American loss and level plan for voice gateways to IP based networks*

4 Definitions, Abbreviations and Acronyms

For the purposes of this standard, the following definitions, abbreviations and acronyms apply.

4.1 Voice Gateway Definition

A device which routes packetized voice from one end-point to another, and provides other voice related functions that a data gateway would not provide.

Its function is analogous to a PBX in that it provides connectivity between customer premise voice terminals. It may provide interfaces to analog and digital (TDM and IP) voice terminals, and access to both public and private WANs and public and private switched telephone networks.

4.2 Insertion Loss Definition

The insertion loss of a voice gateway connection is defined as the 1 kHz level difference between the power delivered from a source connected across an input port to the power delivered to a measuring instrument connected across an output port.

Both the signal source and the measurement instrument are assumed to have an impedance of 600 Ω at 1 kHz.

The insertion loss values are expressed as an absolute loss in dB between interface ports.

4.3 Sound Pressure Level Definition

Sound pressure level is a value expressed as a ratio of the pressure of a sound to a reference pressure. The following sound level units are used in this standard:

dBPa: The sound pressure level, in decibels of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure of this sound to the reference pressure of 1 Pascal (Pa). Note: 1 Pa = 1 N/m^2.

dBSPL: The sound pressure level, in decibels of a sound is 20 times the logarithm to the base 10 of the ratio of the pressure of this sound to the reference pressure of 2×10^{-5} N/m^2 (0 dBPa = 94 dBSPL).

4.4 Loudness Rating Definitions

4.4.1 Loudness Rating

Loudness ratings are a function of the acoustic/electrical conversion characteristics of the originating and terminating equipment (typically telephones). These ratings are determined by measuring the conversion characteristics over the telephony frequency band and by applying a weighting factor for each third octave band.

These loudness ratings are defined as the Send Loudness Rating (SLR) and Receive Loudness Rating (RLR), and the sum of these ratings (plus any circuit gain or loss) is defined as the Overall Loudness Rating (OLR).

The following convention is used in this standard when referring to loudness ratings:

- The Send Loudness Rating (SLR) and Receive Loudness Rating (RLR) are collectively referred to as the Loudness Rating (LR).
- The loudness ratings are given in the order SLR and RLR, i.e. a digital telephone with an SLR of 8 dB and RLR of 2 dB would be designated as having an LR of 8 and 2.

4.4.2 Equivalent Loudness Rating

For the purpose of loss planning it is necessary to know the equivalent loudness rating (ELR) of an analog voice gateway port (it can be assumed that a digital port would have a LR of 8 and 2). The ELR of a port is the SLR or RLR of the terminal connected to that port, plus any gain or loss in the connection between the terminal and the port.

For example; an analog telephone with an LR of 8 and –6, connected to a voice gateway via a 2-wire loop with 3 dB loss in each direction, would have an ELR of 11 and –3 as shown in Figure 1.

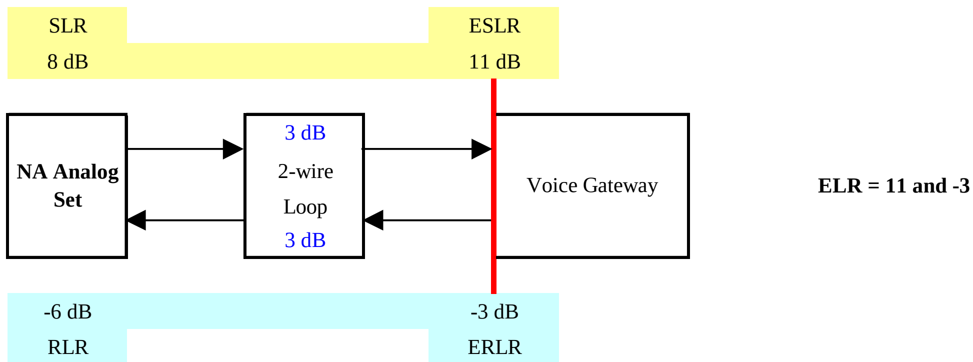


Figure 1 – Equivalent Loudness Rating Example

4.4.3 IP Send Loudness Rating (iSLR)

The concept of equivalent loudness ratings can apply at any point in the connection path. A special case is the voice gateway-to-IP network connection point, as this is the reference point for all IP transmission levels. At this point the ESLR is defined as the iSLR (IP SLR).

See Annex B for further information on loudness ratings.

4.5 Port Definitions

Figure 2 illustrates the connection types, and corresponding voice gateway port definitions, for a typical voice gateway application. There may be other applications not covered by this diagram, but sufficient information is provided for users to determine the applicable connection types for their particular application, and the corresponding port losses to be applied.

Users should be aware that the port definition might only apply to the voice gateway end of the connection, as other entities could have a different port definition for the same connection. For example, the voice gateway FXO connection is defined as an analog access line by the central office.

Note: Connections such as DAL, FXO, FXD, and ATT are generically referred to as trunks. This term refers to connections between central offices in the PSTN, and analogously is used to describe connections between central offices, PBXs, and voice gateways. It should also be noted that from a central office and T1.508 perspective, these voice gateway trunks are called lines.

4.6 Port Descriptions

4.6.1 ONS – On Premise Station

An ONS interface is used for standard analog telephones, representative of 2500-type telephones, located on the same premises as the voice gateway, and is the direct equivalent of the PBX ONS connection. The connection loss from the station to the voice gateway is typically low. The term FXS is sometimes used in place of ONS.

4.6.2 OPS – Off Premise Station

An OPS interface is used for standard analog telephones, representative of 2500-type telephones, not located on the same premises as the voice gateway, and is the direct equivalent of the PBX OPS connection. The connection loss from the station to the voice gateway is typically significant. This port is also used for analog two-wire connections to remote voice gateways, PBXs and Key Systems, via a local central office. The term FXS is sometimes used in place of OPS.

4.6.3 DGS – Digital Station

A DGS interface is used for all digital telephones conforming to ANSI/TIA/EIA-810-A.

Note: This includes digital telephones based on both TDM and packet transmission.

4.6.4 WAN – Wide Area Network

A WAN interface connects to IP-based wide area networks. The transmission path within the WAN is entirely digital. Note: If the WAN connection provides tandem access to the PSTN via another voice gateway, DAL must be used instead of WAN to avoid violating TIA/EIA/IS-968, Section 4.5.2.5.1.

Note: The term WAN is used in the same context as PSTN, in that it represents connections between geographically separated voice gateways. It should be noted that WAN and LAN are synonymous from a transmission perspective.

4.6.5 DAL – Digital Access Line

A DAL interface connects to all digital network connections, except for IP-based wide area network connections, and it is the direct equivalent of the PBX DAL interface. It should be noted that although the connection to the public switched network may be digital, there is no guarantee that the end-to-end connection will remain digital. Therefore the loss inserted by the gateway on connections between DAL and other gateway interfaces is subject to the limits specified in TIA/EIA/IS-968, Section 4.5.2.5.1.

4.6.6 FXO – Foreign Exchange Office

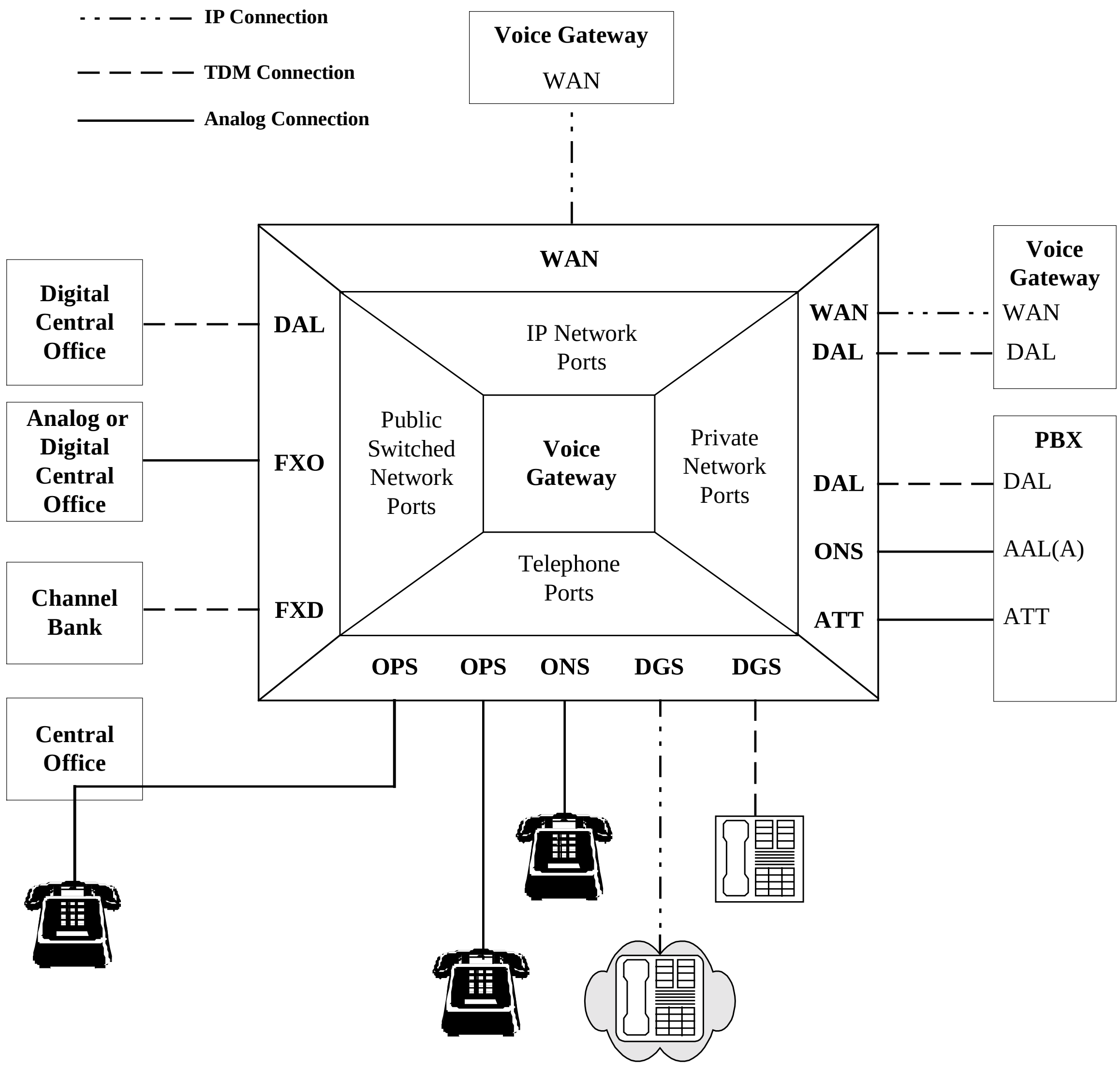
An FXO interface is used for analog connections to a central office. It is equivalent to the PBX term AAL(A), or analog access line (analog).

4.6.7 FXD – Foreign Exchange Digital

An FXD interface is used for digital connection, via a channel bank, to an analog central office. It is equivalent to the PBX term AAL(D), or analog access line (digital). A loss equivalent to the typical analog connection loss has to be inserted at the voice gateway, as the channel bank is located close to the central office.

4.6.8 ATT – Analog Tie Trunk

An ATT interface is used for four-wire analog private network connections, typically via the public network. This port also applies to two-wire voice gateway interfaces that use an external four-wire termination set (4WTS) to connect to the public network. PBX documents may either use the same term, ATT, or the older term, A/TT.



Voice Gateway Port Designation	Port Definition	PBX or CO Port Designation
ONS	**A**nalog **O**n Premise **S**tation	ONS
OPS	**A**nalog Off **P**remise **S**tation - can be via an analog connection to a CO	OPS
DGS	**D**i**g**ital **S**tation (telephone)	DGS
WAN	**W**ide **A**rea **N**etwork (IP connection)	-
DAL	**D**igital **A**ccess **L**ine - digital connection to a digital CO	DAL
FXO	**F**oreign E**x**change **O**ffice - analog connection to analog or digital CO	AAL(A)
FXD	**F**oreign E**x**change **D**igital - digital connection to an analog CO	AAL(D)
ATT	**A**nalog **T**ie **T**runk - private analog network connection to a gateway/PBX	ATT

Figure 2 – Voice Gateway Connection Definitions

4.7 Abbreviations and Acronyms

Abbreviations and acronyms, other than in common usage, are defined below.

AAL(A)	**A**nalog **A**ccess **L**ine (**A**nalog Interface at PBX)
AAL(D)	**A**nalog **A**ccess **L**ine (**D**igital Interface at PBX)
A/D	**A**nalog**-to-D**igital
ATT	**A**nalog **T**ie **T**runk (latest acronym format)
A/TT	**A**nalog **T**ie **T**runk (previous acronym format)
BRL	**B**alance **R**eturn **L**oss
CO	**C**entral **O**ffice
D/A	**D**igital**-to-A**nalog
DAL	**D**igital **A**ccess **L**ine
DID	**D**irect **I**nward **D**ialing
DGS	**Dig**ital **S**tation
DMW	**D**igital **M**illi**w**att
ELR	**E**quivalent **L**oudness **R**ating
ERLR	**E**quivalent **R**eceive **L**oudness **R**ating
ESLR	**E**quivalent Send **L**oudness **R**ating
FDM	**F**requency **D**ivision **M**ultiplexing
FXD	**F**oreign E**x**change **D**igital
FXO	**F**oreign E**x**change **O**ffice
FXS	**F**oreign E**x**change **S**tation (see ONS & OPS)
IP	**I**nternet **P**rotocol
iSLR	**I**P **S**end **L**oudness **R**ating
LAN	**L**ocal **A**rea **N**etwork
OLL	**O**pen **L**oop **L**oss
OLR	**O**verall **L**oudness **R**ating
ONS	**O**n **P**remise **S**tation
OPS	**O**ff **P**remise **S**tation
PBX	**P**rivate **B**ranch E**x**change
PCM	**P**ulse **C**ode **M**odulation
PSTN	**P**ublic **S**witched **T**elephone **N**etwork
RLR	**R**eceive **L**oudness **R**ating
SLR	**S**end **L**oudness **R**ating
SPL	**S**ound **P**ressure **L**evel
TDM	**T**ime **D**ivision **M**ultiplex
WAN	**W**ide **A**rea **N**etwork
ZLP	**Z**ero **L**evel **P**oint

5 Preamble

5.1 General

Digital interfaces in the United States and Canada use μ-law encoding/decoding as defined in ITU-T Recommendation G.711 (1988), 'Pulse Code Modulation of Voice Frequencies' [Ref. 8].

The transmission requirements contained in this standard are based on an industry-developed fixed loss and level plan. The requirements were developed with the objective of maintaining or improving the quality of service for connections within existing and evolving communication networks.

The requirements contained in this standard are based on current understanding of required performance and on the capabilities of present technology. As technology evolves, or as performance needs change, these requirements may become subject to change.

Note: For historical reasons the terms Stations, Sets, Telephones, and Terminals are used interchangeably in this standard.

5.2 Reference Impedance

Transmission requirements contained here apply with station and trunk interfaces terminated in a nominal impedance of 600 Ω, unless otherwise specified.

All measurements should be made at an equipment access point connected to the equipment by no more than 15 meters of cable, unless otherwise specified.

6 Loss and Level Plans

6.1 Introduction

A primary reason to establish a loss plan for voice communication systems is the desire to have the received speech loudness at a comfortable listening level. This received loudness will depend upon the speech level of the talker, the transmit and receive efficiencies of the voice terminals and the loss in the intervening network subsystems such as loops, trunks and switching systems. It is generally accepted among voice transmission experts that a connection with an overall loudness rating of 10 dB, which approximates a normal conversation between a talker and listener spaced 1 meter apart, will provide a high degree of satisfaction for the majority of users.

Another important reason to establish a loss plan is to economically minimize the effect of echo due to signal reflections that are caused by impedance mismatching at 2-to-4 wire conversions in the transmission path. In general, the insertion of loss in the transmission path reduces the impairment due to echo, but increases the impairment due to noise. Another consideration that must be taken into account is that insertion of too much loss will adversely affect customer satisfaction with the received listening level. Therefore, rather than increasing loss indefinitely on longer circuits, echo is controlled by the deployment of echo cancellers.

If digital telephones (i.e., conforming ANSI/TIA/EIA-810-A) are used at each end of the connection, and if the entire end-to-end connection is over a digital based network, then the loss plan is very simple. The digital sets provide the desired overall loudness rating, hence the voice gateway would not insert any loss or gain in the voice channel.

In practice, however, many voice gateway connections will involve the public network in one form or another, either by connections to the public switched network or by connections to private networks over public network facilities (e.g., tie trunks).

In general, connections through the public network can be analog either in whole or in part and involve
2-to-4 wire conversions. The allocation of loss among the public network subsystems affects the noise at the telephone receiver, the echo heard by the talker and the listener, the probability of hearing other conversations, and the probability of causing interference on connections being used by other customers. Also signals at too high a level can cause intermodulation distortion in some older carrier systems.

This standard recommends a loss and level plan for voice gateways that specifies the amount of loss or gain to be inserted by the gateway when interfacing with the various elements of public and private telecommunications networks. It is intended to be co-ordinated with the public network loss plan according to the principles of ANSI T1.508-1998 and it is intended to fully comply with TIA/EIA/IS-968 Standard.

The loss and level plan consists of two parts:

- A full-channel plan that defines the port-to-port losses for all connections types. This is similar to the PBX loss level plan as specified in PN-3-3673 (to be published as ANSI/TIA/EIA-464-C) [Ref. 9], and ensures satisfactory interworking with the existing TDM based public and private networks

- A half-channel plan that defines the port-to-IP network and IP network-to-port losses. This plan is specific to IP telephony and will facilitate the interworking of national and international IP telephony networks.

6.2 Port-to-Port Loss Allocation

The allocation of the port-to-port loss to the send and receive ports directly influences the available dynamic range of the PCM coding scheme. This may lead to substantial impacts on speech transmission quality as perceived by the user.

Care should be taken to ensure that excessive input gain or loss does not cause either PCM encoding overload, or a poor signal-to-noise ratio, at the zero-level point.

See Annex B.3 for further information on port-to-port loss allocation.

6.3 Digital Padding

Digital padding refers to the technique of implementing port-to-port losses by changing the level of the digitally encoded voice. Digital padding should be avoided wherever possible, as it has the potential to increase quantization distortion. Losses involving analog ports should be implemented before encoding or after decoding.

6.4 Voice Gateway Loss Plan Assumptions

The port-to-port losses for voice gateways were developed based on the loss plans for public and private networks. The voice gateway loss plan is intended to provide satisfactory grade-of-service (GoS) performance and compatibility with the public and private network loss plans. The following assumptions were the prime considerations:

(1) The transmission loss and level plan of the PSTN, which is primarily an all-digital network with some fixed loss as described in T1.508, forms a basis for the private network loss plan.

(2) Central offices insert the following variable losses on analog trunks connections to ensure network stability:

Connection Type	**CO Loss**	
Local Call	0 dB	
Regional Call	3 dB	(*Note: this is the assumed nominal loss for planning purposes)*
Long Distance Call	6 dB	

(3) Transmission facilities to be used have losses compatible with the voice gateway port-to-port losses.

(4) The ONS connection loss from the station to the voice gateway is typically low.

(5) The OPS connection loss from the station to the voice gateway is typically significant.

(6) FXD connections require a loss equivalent to a typical analog connection to be inserted at the voice gateway, as the channel bank is located close to the central office.

(7) Non-proprietary voice gateway stations have the following loudness ratings:

Station	SLR	RLR	Notes
OPS	11	-3	a, b)
ONS	8	-6	c)
DGS	8	2	d)

Notes:

a) These loudness ratings are defined as equivalent loudness ratings (ESLR & ERLR) values at the input to the voice gateway ports.

b) The OPS (off premise station) ratings are representative of 2500-type sets operating on 26 gauge/2.75 km loops with normal battery feed and impedance characteristics, as measured at a PSTN end office or voice gateway OPS port.

c) The ONS (on premise station) ratings are representative of 2500-type sets operating on short loops with the typical current-limited battery feed and 600 Ω impedance characteristics of voice gateway ONS ports. See PN-3-4350.110 (to be published as ANSI/TIA/EIA-470-110-C) for further details.

d) The DGS (digital station) ratings are chosen to improve interoperability with the ITU standard SLR/RLR levels for digital sets of 8 and 2. See ANSI/TIA/EIA-810-A for further details.

6.5 Full-Channel Loss and Level Plan

6.5.1 Insertion Loss Criteria for Voice Gateways

The nominal values for voice gateway port-to-port connections are given by the loss plan presented in Table 1. The voice gateway interfaces depicted in the loss plan are defined in Figure 2, and described in Section 4.6.

6.5.2 Port-to-Port Loss Table Interpretation

In Table 1, arrows at the row and column designators indicate the transmission direction in which the co-ordinate loss values are to be inserted.

For example, co-ordinate 1B indicates a nominal port-to-port loss of 3 dB from the ONS interface to the OPS interface, and co-ordinate 2A indicates a 3 dB loss in the other direction, from the OPS interface to the ONS interface.

Note: This is a loss plan, therefore negative values denote gain; e.g., -3 indicates 3 dB gain.

6.5.3 Port-to-Port Loudness Ratings Table Interpretation

Table 2 is provided to show the relationship between OLR and ESLR, Loss, and ERLR.

For example, for an ONS to OPS connection, the ONS ESLR is 8 dB, the voice gateway loss is 3 dB (from 1B in Table 1), and the OPS ERLR is –3 dB. The overall loudness ratings is therefore 8 + 3 – 3 = 8 dB.

See Annex B.2 for further information on loudness ratings.

Table 1 – Voice Gateway Loss Plan

			A	B	C	D	E	F	G	H
			ONS	**OPS**	**DGS**	**WAN**	**DAL**	**FXO**	**FXD**	**ATT**
		Loss	↑	↑	↑	↑	↑	↑	↑	↑
1	**ONS**	→	6	3	0	0	3	0	3	3
2	**OPS**	→	3	0	-3	-3	0	0	0	3
3	**DGS**	→	9	6	0	0	0	0	3	3
4	**WAN**	→	9	6	0	0	0	0	3	3
5	**DAL**	→	9	6	0	0	0	3	3	3
6	**FXO**	→	0	0	-9	-6	-3	0	0	0
7	**FXD**	→	3	0	-6	-3	-3	0	0	0
8	**ATT**	→	3	0	-3	-3	-3	0	0	0

Note 1: The units for all loss values are dB.

Note 2: Losses have been selected as multiples of 3 dB, in the assumption that this may make implementation easier.

Note 3: The is a potential risk of DTMF overload if an analog trunk connected voice gateway is located less than 2 km from the central office. In these cases it is recommended that the FXD/WAN settings be used for FXO/WAN connections instead of the FXO/WAN settings, as the FXD setting introduces an additional 3 dB loss in each direction. See Annex B.4 'DTMF Overload on Analog Trunks'

Table 2 – Voice Gateway Loudness Ratings

			ONS	OPS	DGS	WAN	DAL	FXO	FXD	ATT
		ERLR	-6	-3	2	2	2	3	0	-1
	ESLR	**OLR**	↑	↑	↑	↑	↑	↑	↑	↑
ONS	8	→	8	8	10	10	13	11	11	10
OPS	11	→	8	8	10	10	13	14	11	13
DGS	8	→	11	11	10	10	10	11	11	10
WAN	8	→	11	11	10	10	10	11	11	10
DAL	8	→	11	11	10	10	10	14	11	10
FXO	17	→	11	14	10	13	16	20	17	16
FXD	14	→	11	11	10	13	13	17	14	13
ATT	13	→	10	10	12	12	12	16	13	12

Note 1: The units for all loudness ratings (ESLR, ERLR, & OLR) are dB.

Note 2: The loudness ratings for the FXO and FXD ports include a nominal 3 dB CO loss.

Note 3: The loudness ratings for the ATT port include a nominal 2 dB trunk loss.

6.5.4 Voice Gateway Loss Ranges and TIA/EIA/IS-968 Requirements

6.5.4.1 Recommended Loss Ranges

The port-to-port losses in Table 1 are the recommended nominal values.

Although there are no mandatory loss ranges associated with these values, it is desirable that the average 1 kHz loss fall within ± 0.5 dB of the nominal loss values given in Table 1.

6.5.4.2 TIA/EIA/IS-968 Requirements and FCC Part 68 Rules

Tables 3 through 5 provide a cross reference from the Table 1 loss values to the equivalent limits for the IS-968 defined ports. It should be noted that a number of analog connections have a zero tolerance between the Table 1 values and the IS-968 limits.

Note: Zero tolerance means that the nominal values in Table 1 are identical to the absolute limits in IS-968.

Care should be taken to ensure that no analog connection violates the IS-968 standard for through-gain transmission (allowable net amplification between ports) [Ref. 8: IS-968, Section 4.5.2.5]. In general this is achieved by offsetting the nominal loss by an amount equal to the maximum expected component tolerance and measurement error.

The recommended losses are based on a desirable port-to-port OLR of 10 dB.

Unfortunately, some of the desirable ONS and OPS losses would violate the IS-968 requirements for through-gain transmission. In these cases the losses have been set to levels compliant with the IS-968 requirements, even though this results in a less than optimum OLR of 13 dB.

Two of the connections affected are ONS and OPS to DAL. Although the connection to the public switched network may be digital, there is no guarantee that the end-to-end PSTN connection will remain digital. Therefore the loss inserted by the gateway on connections between DAL and other gateway interfaces is subject to the limits specified in TIA/EIA/IS-968, Section 4.5.2.5.

The transmission path within the WAN is entirely digital, therefore the recommended losses for WAN connections can be optimized to provide an OLR of 10 dB.

Note: DAL must be used instead of WAN if the WAN connection provides tandem access to the PSTN via another voice gateway, to avoid violating the TIA/EIA/IS-968 requirements.

Table 3a – TIA/EIA/IS-968, Section 4.5.2.5 - Allowable Net Amplification Between Ports

To			Tie trunk type ports			Integrated services trunk	OPS ports (2-wire)	Public switched network ports (2-wire)		HCC digital PBX-CO (4-wire)
From (E)			2/4-wire	Subrate 1.544 Mbps satellite 4W	Subrate 1.544 Mbps tandem 4W					
			ATT	DAL	DAL	DAL	OPS	FXO	FXD	DAL
			1	2	3	4	7	8	9	10
1	ATT	2/4-Wire Tie	0 dB	3 dB	3 dB	3 dB	6 dB			
2	DAL	Subrate 1.544 Mbps Satellite 4W Tie	0 dB		3 dB	3 dB	6 dB			
3	DAL	Subrate 1.544 Mbps Tandem 4W Tie	-3 dB	0 dB	0 dB	0 dB	3 dB			
4	DAL	Integrated Services Trunk	-3 dB	0 dB	0 dB	0 dB	3 dB			
5	DGS	RTE Digital	0 dB	0 dB	0 dB	0 dB	3 dB	3 dB	3 dB	0 dB
6	ONS	RTE (B) PSTN/ONS	-3 dB	-3 dB	-3 dB	-3 dB	0 dB	0 dB	0 dB	-3 dB
7	OPS	OPS (2-Wire)	-2 dB	1 dB	1 dB	1 dB	4 dB	4 dB	4 dB	1 dB
8	FXO	Public Switched Network (2-Wire)					3 dB	3 dB	3 dB	
9	FXD	Public Switched Network (2-Wire)					3 dB	3 dB	3 dB	
10	DAL	HCC Digital PBX-CO (4-Wire)					3 dB			

Note: In this table, positive values denote gain and negative (-) denotes loss.

Table 3b – TIA/EIA/IS-968, Section 4.5.2.5 - Allowable Net Loss Between Ports

To			Tie trunk type ports			Integrated services trunk	OPS ports (2-wire)	Public switched network ports (2-wire)		HCC digital PBX-CO (4-wire)
From (E)			2/4-wire	Subrate 1.544 Mbps satellite 4W	Subrate 1.544 Mbps tandem 4W					
			ATT	DAL	DAL	DAL	OPS	FXO	FXD	DAL
			1	2	3	4	7	8	9	10
1	ATT	2/4-Wire Tie	0 dB	-3 dB	-3 dB	-3 dB	-6 dB			
2	DAL	Subrate 1.544 Mbps Satellite 4W Tie	0 dB		-3 dB	-3 dB	-6 dB			
3	DAL	Subrate 1.544 Mbps Tandem 4W Tie	3 dB	0 dB	0 dB	0 dB	-3 dB			
4	DAL	Integrated Services Trunk	3 dB	0 dB	0 dB	0 dB	-3 dB			
5	DGS	RTE Digital	0 dB	0 dB	0 dB	0 dB	-3 dB	-3 dB	-3 dB	0 dB
6	ONS	RTE (B) PSTN/ONS	3 dB	3 dB	3 dB	3 dB	0 dB	0 dB	0 dB	3 dB
7	OPS	OPS (2-Wire)	2 dB	-1 dB	-1 dB	-1 dB	-4 dB	-4 dB	-4 dB	-1 dB
8	FXO	Public Switched Network (2-Wire)					-3 dB	-3 dB	-3 dB	
9	FXD	Public Switched Network (2-Wire)					-3 dB	-3 dB	-3 dB	
10	DAL	HCC Digital PBX-CO (4-Wire)					-3 dB			

Note: In this table, positive values denote loss and negative (-) denotes gain.

Table 4 – Voice Gateway Recommended Net Loss Between Ports

To			Tie trunk type ports			Integrated services trunk	OPS ports (2-wire)	Public switched network ports (2-wire)		HCC digital PBX-CO (4-wire)
From (E)			2/4-wire	Subrate 1.544 Mbps satellite 4W	Subrate 1.544 Mbps tandem 4W					
			ATT	DAL	DAL	DAL	OPS	FXO	FXD	DAL
			1	2	3	4	7	8	9	10
1	ATT	2/4-Wire Tie	0 dB	-3 dB	-3 dB	-3 dB	0 dB	0 dB	0 dB	-3 dB
2	DAL	Subrate 1.544 Mbps Satellite 4W Tie	3 dB	0 dB	0 dB	0 dB	-6 dB	3 dB	3 dB	0 dB
3	DAL	Subrate 1.544 Mbps Tandem 4W Tie	3 dB	0 dB	0 dB	0 dB	6 dB	3 dB	3 dB	0 dB
4	DAL	Integrated Services Trunk	3 dB	0 dB	0 dB	0 dB	6 dB	3 dB	3 dB	0 dB
5	DGS	RTE Digital	3 dB	0 dB	0 dB	0 dB	6 dB	0 dB	3 dB	0 dB
6	ONS	RTE (B) PSTN/ONS	3 dB	3 dB	3 dB	3 dB	3 dB	0 dB	3 dB	3 dB
7	OPS	OPS (2-Wire)	3 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB	0 dB
8	FXO	Public Switched Network (2-Wire)	0 dB	-3 dB	-3 dB	-3 dB	0 dB	0 dB	0 dB	-3 dB
9	FXD	Public Switched Network (2-Wire)	0 dB	-3 dB	-3 dB	-3 dB	0 dB	0 dB	0 dB	-3 dB
10	DAL	HCC Digital PBX-CO (4-Wire)	3 dB	0 dB	0 dB	0 dB	6 dB	3 dB	3 dB	0 dB

Note: In this table, positive values denote loss and negative (-) denotes gain

Table 5 – Differences between TIA/EIA/IS-968, and Voice Gateway Port Losses

To			Tie trunk type ports			Integrated services trunk	OPS ports (2-wire)	Public switched network ports (2-wire)		HCC digital PBX-CO (4-wire)
From (E)			2/4-wire	Subrate 1.544 Mbps satellite 4W	Subrate 1.544 Mbps tandem 4W					
			ATT	DAL	DAL	DAL	OPS	FXO	FXD	DAL
			1	2	3	4	7	8	9	10
1	ATT	2/4-Wire Tie	**0 dB**	**0 dB**	**0 dB**	**0 dB**	6 dB			
2	DAL	Subrate 1.544 Mbps Satellite 4W Tie	3 dB		3 dB	3 dB	12 dB			
3	DAL	Subrate 1.544 Mbps Tandem 4W Tie	**0 dB**	0 dB	0 dB	0 dB	9 dB			
4	DAL	Integrated Services Trunk	**0 dB**	0 dB	0 dB	0 dB	9 dB			
5	DGS	RTE Digital	3 dB	0 dB	0 dB	0 dB	9 dB	3 dB	6 dB	0 dB
6	ONS	RTE (B) PSTN/ONS	**0 dB**	**0 dB**	**0 dB**	**0 dB**	3 dB	**0 dB**	3 dB	**0 dB**
7	OPS	OPS (2-Wire)	1 dB	1 dB	1 dB	1 dB	4 dB	4 dB	4 dB	1 dB
8	FXO	Public Switched Network (2-Wire)					3 dB	3 dB	3 dB	
9	FXD	Public Switched Network (2-Wire)					3 dB	3 dB	3 dB	
10	DAL	HCC Digital PBX-CO (4-Wire)					9 dB			

Note: In this table, positive values denote loss and negative (-) denotes gain

Notes:

1. *The IS-968 limits in Table 3a and Table 3b are only provided as a convenience to users of this standard. User should consult the latest IS-968 standard to ensure that they are in compliance.*
2. *The RTE(B) PSTN/ONS ports are for 2-wire on-premises station ports to separately registered terminal equipment.*
3. *HCC Digital PBX-CO (4-wire) ports are for 4-wire 1.544 Mbps High Capacity Circuit digital ports.*
4. *The numbers in the first column of each table are used in the first row of each table as references to the IS-968 port nomenclature*
5. *The shaded cells in Table 5 indicate analog connections with zero tolerance (see Section 6.5.4.2).*

6.6 Half-Channel Loss and Level Plan

6.6.1 Overview

Developing a loss plan is generally a complex process, as the objective is to ensure a satisfactory overall loudness rating (OLR) for all connections types. To do this the loudness ratings of the end points (telephones), and the transmission loss between the end points, must be known for each connection type.

This is a trivial exercise in a purely IP telephony environment, if one assumes that the end points are digital telephones with an LR of 8 and 2 (in line with ITU-T Recommendations), and that no gains or losses are introduced in the digital transmission path. In this case the OLR for any digital telephone-to-telephone connection world-wide is 10 dB, which is the ITU-T objective.

The complexity is introduced when the IP telephony network connects to analog telephones and trunks. In this case the LR of the telephones, and the ELR of the trunks vary, although a loss-less digital transmission path can be maintained.

A half-channel loss plan for national and international IP telephony networks can be implemented based on the premise that only the LRs and ELRs vary, and that the IP network does not introduce any additional gain or loss.

Full-channel loss plans are still required for voice gateways, as voice gateways can also connect to existing analog and TDM based digital networks. These connections require losses to be defined on a port-to-port basis for technical and regulatory reasons. A half-channel loss plan can therefore be considered a sub-set of a full-channel loss plan that is only applicable to IP network connections.

6.6.2 Concept

The basic concept in the half-channel loss plan is to normalize all transmit levels on an IP telephony network to the same equivalent SLR (ESLR) - digital telephones provide the reference SLR of 8 dB by definition. This is not an original concept, as it is the basis of the European dBr reference system. The move to standardize the North American digital telephone LRs to the ITU-T recommended levels makes this practical for IP telephony networks.

The same basic concept could be applied to the current non-IP PSTN and private networks, but existing industry standards and IS-968 requirements would make it difficult to implement.

6.6.3 Principle of Operation

- The originating entity will set the ESLR of the sending end point to 8 dB at the ingress to the IP network. The ESLR at this point is defined as the iSLR.
- The terminating entity will adjust the loss at the egress from the IP network to achieve the desired OLR at the receiving end point.

  Note: The internationally recognized optimum OLR is 10 dB.

The advantage of this approach is that neither entity requires knowledge of the other, and loss planning becomes a local issue.

6.6.4 Applicability

This loss plan applies to all kind of IP based voice services, irrespective whether they provide:

- real time conversational telecommunication between human subjects, or
- listening-only telecommunication from a machine interface (stored speech) to a human subject, or
- speaking-only telecommunication from a human subject to a machine interface;

The half-channel loss plan only applies to connections routed via IP networks; other voice gateway port-to-port connections are subject to the full-channel loss plan recommendations.

6.6.5 Reference Level Point

The reference or zero-level point for IP telephony is defined as the point where a connection is made to a packet based network. This is equivalent to the zero-level point in standard TDM circuit switched telephony.

Note: The reference level is defined as an iSLR of 8 dB, not a power level at 1004 Hz.

6.6.6 Recommended Loss Ranges

The transmit and receive losses in Table 6 are the recommended nominal values.

Although there are no mandatory loss ranges associated with these values, it is desirable that the average 1 kHz loss fall within ± 0.5 dB of the nominal loss values given in Table 6.

6.6.7 IP Network Losses

It is critical for the operation of a half-channel loss plan that no gain or loss is inserted during transmission through the IP network. Any level changes due to transcoding for example, should be less than 1 dB.

Note: Transcoding refers to the conversion from one voice coding algorithm to another, e.g. G.726 to G.729.

6.6.8 Half-Channel Loss Plan Table

Table 6 shows the North American voice gateway half-channel loss plan, given in loudness ratings and respective loss.

Table 6. Voice Gateway Half-Channel Loss Plan

WAN Zero-Level Point ↓

	a	b	c a + b	d	e	f c + d + e	g 8 + d + e
	ESLR	**Tx Loss**	**iSLR**	**ERLR**	**Rx Loss**	**OLR**	**Assumed OLR**
ONS	8	0	8	-6	9	11	11
OPS	11	-3	8	-3	6	11	11
DGS	8	0	8	2	0	10	10
DAL	8	0	8	2	0	10	10
WAN	8	0	8	2	0	10	10
FXO	17	-6	11	3	0	14	11
FXD	14	-3	11	0	3	14	11
ATT	13	-3	10	-1	3	12	10

Column **a** shows the ESLR of the telephones and trunks at the connection point to the voice gateway.

Column **b** shows the transmit loss required to achieve the required iSLR at the zero-level point.

Column **c** shows the resulting ESLR (iSLR) at the zero-level point (WAN).

Column **d** shows the ERLR of the receive side.

Column **e** shows the receive loss required to achieve the desirable OLR, based on the ERLR shown in column **d,** and an assumed iSLR of 8 dB**.**

Column **f** shows the resulting OLR

Column **g** shows the assumed OLR, based on an iSLR of 8

Notes:

1. *Losses have been selected as multiples of 3 dB, in the assumption that this may make implementation easier.*
2. *It is not possible to achieve the optimum iSLR of 8 dB for connections from analog networks due to the potential for DTMF signaling overload. See Annex B.4 'DTMF Overload on Analog Trunks'.*
3. *The OLR values shown in the table are as perceived by the listener, i.e. this is shown as a one-way connection.*

6.6.9 Network Stability

There is a potential for instability in connections involving 2 to 4 wire conversions if the open loop gain of the 4-wire loop approaches 0 dB.

Additional losses are inserted in public and private analog and digital TDM-based networks to ensure they are unconditionally stable, and the resultant high OLRs are accepted for mixed 2-wire and 4-wire networks.

The use of this approach for a half-channel plan would result in unacceptable OLRs for some connection types. Fortunately, the requirement for unconditional stability is not required in IP based networks as a digital 4-wire loop will only oscillate when the hybrids at both ends of the loop are unterminated, and under these conditions the 4-wire loop would not be connected to any analog loops that could be affected by digital loop oscillation.

Notes:

1. *Oscillation in a 4-wire loop has shown to cause harm to analog frequency division multiplexing (FDM) transmission systems by affecting the FDM equipment's signal level management. There is no potential for such harm in purely digital systems.*
2. *See Annex B.5 for further information on open loop loss and network stability*.

6.6.9.1 Stability Loss

ITU-T Recommendation G.122 [Ref. 11], specifies the minimum stability loss for national systems required to prevent instability on international calls. The stability loss of each national system represents one half of the open loop loss (OLL) of the 4-wire loops required to establish an international connection. This is equivalent to the voice gateway loss on either side of an IP network connection. A stability loss of 6 dB at all frequencies between 200 and 3600 Hz will ensure that the G.122 requirements are met. However, stability losses of between 6 dB and 0 dB will formally comply with the present requirements of G.122.

This need not be a requirement for purely IP based connections for the reasons discussed above.

7 Loss Parameters

7.1 Frequency Response

The frequency response recommendations for the A/D and D/A conversions are shown in numerical form in Table 7, and graphically in Figs 3a and 3b. All values are stated relative to the loss measured at 1004 Hz.

Note: Positive numbers indicate higher loss; i.e. negative numbers indicate lower loss than the loss at 1004 Hz.

Frequency	Analog to Digital Loss (dB)		Digital to Analog Loss (dB)	
	Min	Max	Min	Max
60 Hz	20	-	N/A	-
200 Hz	0	3	0	2
300 Hz	-0.25	0.5	-0.25	0.5
3 kHz	-0.25	0.5	-0.25	0.5
3.2 kHz	-0.25	0.75	-0.25	0.75
3.4 kHz	0	1.5	0	1.5
3.4 to 4 kHz	-14x+14	-	-14x+14	-
4 to 4.6 kHz	-18x+14	-	-14x+14	-
4.6 to 12 kHz	32	-	28	-

$x = \sin\{[\pi * (4000-F)]/1200\}$

Table 7 - Frequency Response Recommendations

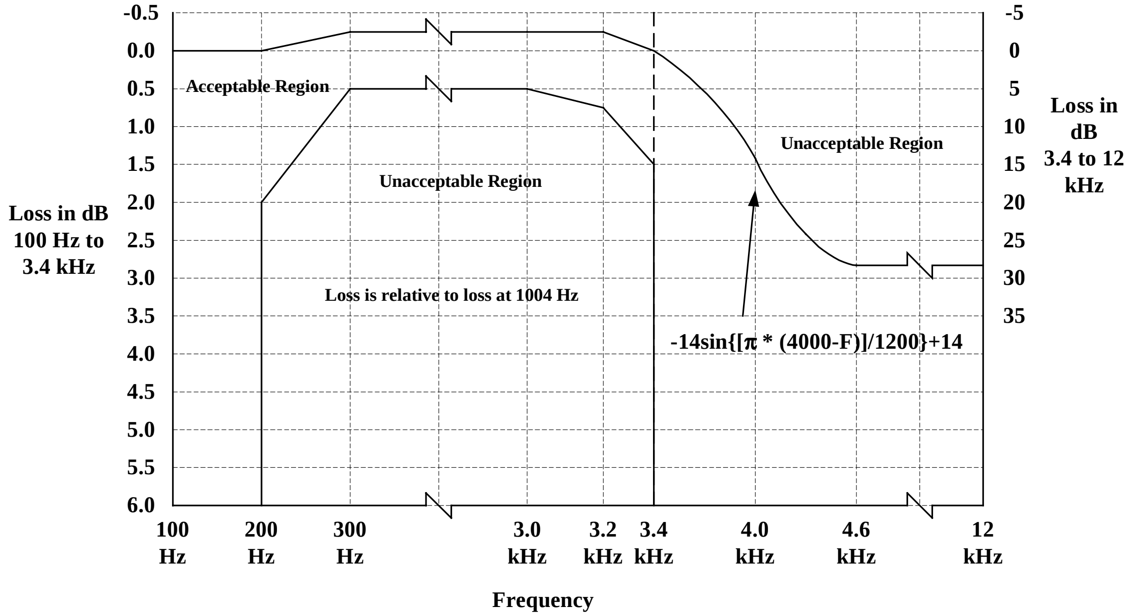


Figure 3a - Voice Gateway Frequency Response - Analog to Digital

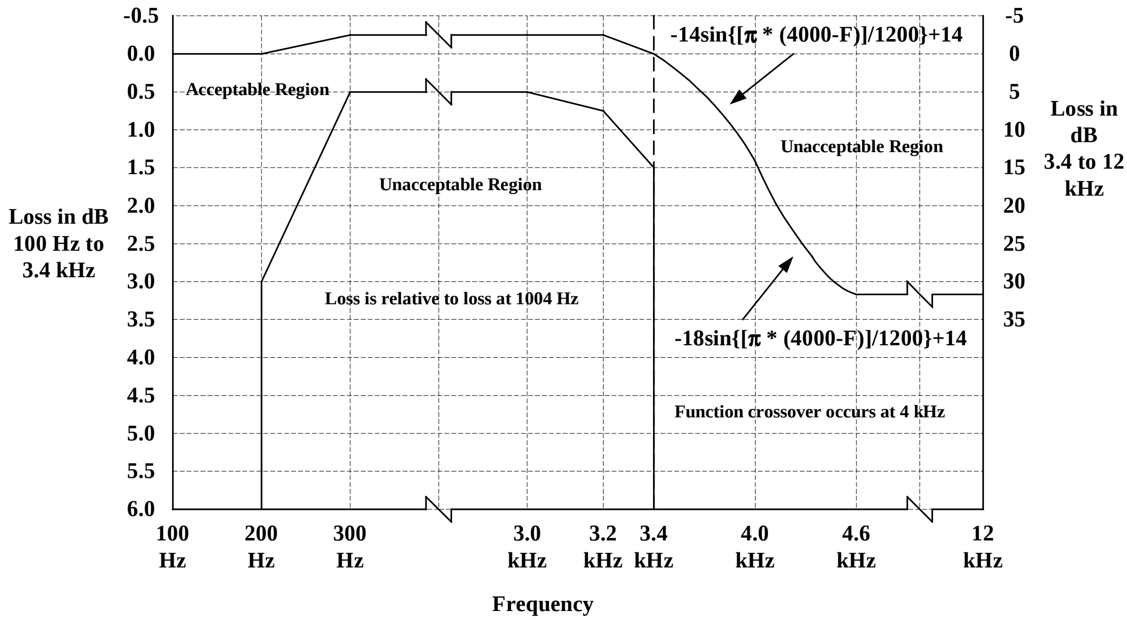


Figure 3b - Voice Gateway Frequency Response - Digital to Analog

7.2 Tracking Error and Overload Compression

7.2.1 Tracking Error

The tracking error for A/D and D/A conversions should not exceed the limits shown in Table 8.

The tracking error for all port-to-port connections should not exceed the limits shown in Figure 4.

Table 8 - Voice Gateway Tracking Error Limits

Input Signal	Tracking Error (dB)	
Level Range (dBm)	Maximum	Average
0 to -37	±0.25	±0.125
-37 to -50	±0.5	±0.25

7.2.2 Overload Compression

For all port-to-port connections, the compression of a 1004 Hz input signal relative to a 1004 Hz, 0 dBm input signal should not exceed the values shown in Figure 4.

Note: Care should be taken to ensure that port-to-port losses are not implemented in a manner that causes PCM encoding overload. See Annex B.3 for further information on port-to-port loss allocation.

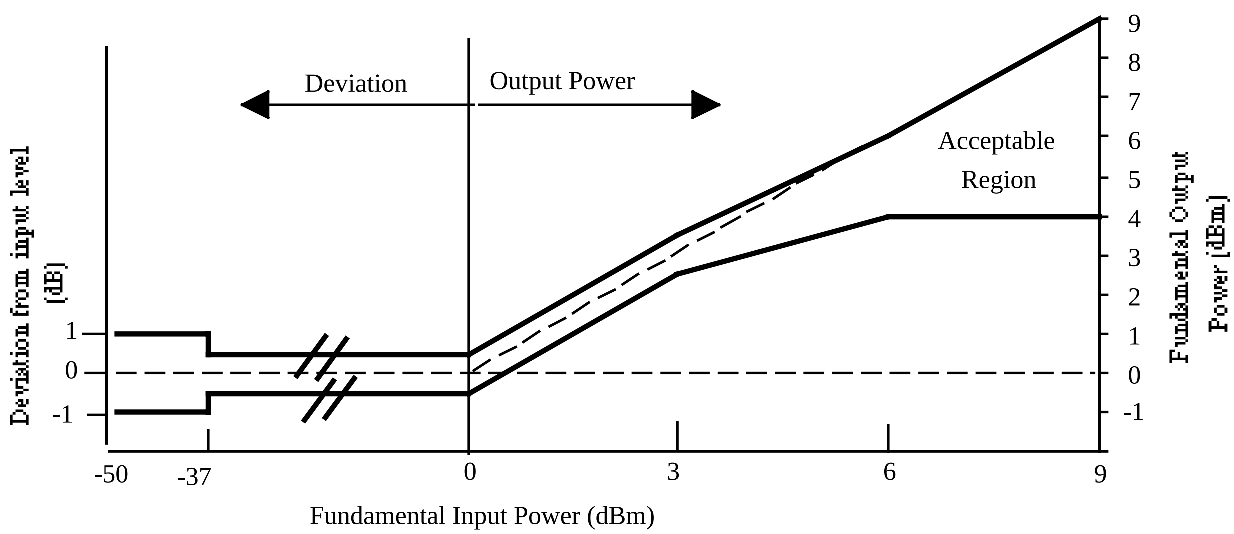


Figure 4 – Tracking Error and Overload Compression

8 Echo Control and Return Loss

Echo in transmission systems is due to reflections from impedance mismatches, and the most common source of impedance mismatches in telephone networks is in the 2-to-4 wire converter or hybrid. The echo return loss is defined for two directions:

- 4-wire to 4-wire, which is a function of the hybrid balance
- 2-wire, which is a function of the 2-wire input impedance

Active echo control using echo cancellers or suppressers is beyond the scope of this section.

8.1 Hybrid Balance Requirements

For each interface, the line or trunk side 2-wire analog port should be terminated with the appropriate reference impedance (see 8.1.1.1). The reference impedances should consist of passive elements. The hybrid balance, when measured as described in Annex A.5, should exceed the values in Table 9 on 95 percent of the interfaces.

Note: Two-wire analog trunks include FXO, DID, and ATT ports.

Table 9 - Voice Gateway Minimum Hybrid Balance Recommendations

Frequency Band	Hybrid Balance
200 to 500 Hz	Equal to or greater than the values located on a straight line intersecting 17 dB at 200 Hz and 22 dB at 500 Hz.
500 Hz to 2.5 kHz	22 dB
2.4 kHz to 3.4 kHz	Equal to or greater than the values located on a straight line intersecting 22 dB at 2.5 kHz and 17 dB at 3.4 kHz.

Note: All points are plotted on a log/linear scale with the impedance values in dB on the linear axis and the frequency in Hz on the logarithmic axis.

8.1.1.1 Hybrid Balance Reference Impedances

8.1.1.1.1 ONS Ports

A reference impedance of 600 Ω is recommended for ONS ports. This network has been selected to match the distribution of telephone set impedances expected in the voice gateway on-premises environment.

8.1.1.1.2 OPS and Two-Wire Analog Trunk Ports With Line Treatment

A reference impedance of 600 Ω is recommended for OPS and two-wire trunk ports that connect to facilities with line treatment[1].

[1] As used in this context, the term "line treatment" means any equipment (e.g., an impedance compensator, a repeater, or a range extender) that presents a nominal impedance of 600 Ω at the interface connecting to the port.

8.1.1.1.3 OPS and Two-Wire Analog Trunk Ports Without Line Treatment

The reference impedance shown in Figure 5 is recommended for OPS and two-wire trunk ports that connect to facilities without line treatment. This impedance has been found to provide the best single compromise to the distribution of OPS line and 2-wire trunk impedances expected in the North American telephone network.

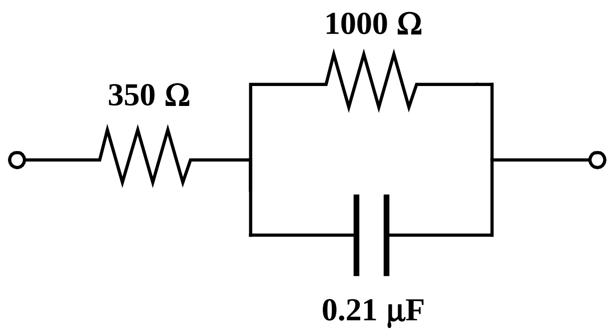


Figure 5 - OPS/2-Wire Trunk Reference Impedance

8.2 Input Impedance Requirements

Requirements are given only for paths through the switch for which the connecting port interface (the interface on the other side of the switch) is 4-wire. In this way, the measured results are independent of any feedback resulting from imperfect balance of the far-end hybrids on 2-wire interfaces. While such imperfect balance does influence the 2-wire interface input impedance and the loss of the switch when it connects two 2-wire lines, the effects on in-service performance are controlled by having separate requirements for hybrid balance (Section 8.1) and insertion loss (Section 6).

The requirements for the input impedance are given in terms of a reference impedance (Z_R) and minimum return loss. The return loss is defined in Annex C where the input impedance is denoted by Z_I and reference impedance by Z_R. The return loss is a function of frequency, and increases without limit as the input impedance approaches the reference impedance.

For each 2-wire and 4-wire analog port, the input impedance (Input Z) in terms of return loss (single frequency return loss and echo return loss) should exceed the values in Tables 10 or 11 on 95 percent of the interfaces, when measured as described in Annex A.6.

8.2.1.1 Input Impedance Reference Impedances

8.2.1.1.1 ONS Ports

ONS ports should meet the minimum input impedance requirement when using a reference impedance consisting of either 600 Ω or a complex reference impedance as described in 8.2.1.1.4.

8.2.1.1.2 OPS Ports

OPS ports should meet the minimum input impedance requirement with a reference impedance of 600 Ω and may, optionally, also meet the input impedance requirement with a complex reference impedance as described in 8.2.1.1.4.

8.2.1.1.3 Two-Wire Analog Trunk Ports

Two-wire analog trunks ports should meet the minimum input impedance requirement with a reference impedance of 600 Ω.

Note: Two-wire analog trunks include FXO, DID, and ATT ports.

8.2.1.1.4 Complex Reference Impedances

This option allows flexibility in the design of line input impedance for specific applications or terminations. The recommended reference impedance for measuring the return loss of lines designed with a complex input impedance network is either:

- The network shown in Figure 5
- A network consisting of a 275 Ω resistor in series with a parallel circuit of a 780 Ω resistor and 0.15 µF capacitor
- A network consisting of a 600 Ω resistance in series with a 2.16 µF capacitor

Table 10 - Voice Gateway Return Loss Recommendations for 600 Ω Z_R

	Return Loss	
Frequency Band	**Minimum**	**Desirable**
200 to 500 Hz	Equal to or greater than the values located on a straight line intersecting 14 dB at 200 Hz and 22 dB at 500 Hz.	Equal to or greater than the values located on a straight line intersecting 14 dB at 200 Hz and 26 dB at 500 Hz.
500 Hz to 2.5 kHz	22 dB	26 dB
2.4 kHz to 3.4 kHz	Equal to or greater than the values located on a straight line intersecting 22 dB at 2.5 kHz and 14 dB at 3.4 kHz.	Equal to or greater than the values located on a straight line intersecting 26 dB at 2.5 kHz and 14 dB at 3.4 kHz.

Table 11 - Voice Gateway Return Loss Recommendations for Complex Z_R

	Return Loss	
Frequency Band	**Minimum**	**Desirable**
200 to 500 Hz	Equal to or greater than the values located on a straight line intersecting 14 dB at 200 Hz and 22 dB at 500 Hz.	-
500 Hz to 2.5 kHz	22 dB	-
2.4 kHz to 3.4 kHz	Equal to or greater than the values located on a straight line intersecting 22 dB at 2.5 kHz and 14 dB at 3.4 kHz.	-

Note: All points are plotted on a log/linear scale with the impedance values in dB on the linear axis and the frequency in Hz on the logarithmic axis.

9 Noise and Distortion Impairments

9.1 Idle-Channel Noise

Idle-channel noise is the short-term, average, absolute noise power measured with a flat or C-message weighting in the absence of signal.

Compliance with the idle-channel noise requirements should be determined as described in Annex A.7.

9.1.1 3 kHz Flat Noise

(1) The 3 kHz flat weighted noise should not exceed 35 dBrn on 50 percent of the connections.

(2) The 3 kHz flat weighted noise should not exceed 39 dBrn on 95 percent of the connections.

For interface transmission levels other than 0 dB, the 3 kHz flat weighted noise requirement should be shifted by a value that corresponds to the difference between the transmission level at that interface and 0 dB.

9.1.2 C- Message Weighted Noise

The maximum (95 percent) C-message weighted absolute noise power at an interface should not exceed the values given in Table 12. It is desirable that the mean C-message weighted absolute noise power also comply with the values shown in the table.

For interface transmission levels other than 0 dB, the C-message weighted noise requirement should be shifted by a value that corresponds to the difference between the transmission level at that interface and 0 dB.

Table 12 - Voice Gateway C-Message Weighted Noise Requirements

C-Message Weighted Noise (dBrnC) in the Absence of Signal		
Connection Type	Mean (Desirable)	95% (Maximum)
Analog-to-analog	16	20
Analog-to-digital	15	19
Digital-to-analog	9	13

9.2 Longitudinal Balance

The voice gateway interfaces that are subject to longitudinal balance requirements include loop/ground start CO/FXO trunks, reverse battery (DID) trunks, OPS lines, and digital service trunks.

9.2.1 Longitudinal-to-Metallic Balance

(1) Definition.

The longitudinal-to-metallic balance is defined as:

$$\text{Longitudinal Balance (dB)} = 20 \log|Vs / Vm|.$$

where Vs is the disturbing longitudinal rms voltage, and Vm the resulting metallic rms voltage of the same frequency.

The longitudinal-to-metallic balance recommendations only apply to analog ports. A low conversion of longitudinal into metallic noise is required to limit noise on the talking circuit.

(2) Recommendations.

The longitudinal-to-metallic balance of analog ports, when measured as described in Annex A.8.1, should meet or exceed the criteria of Table 13. It is desirable that the average balance be within the region labeled "desirable" in Figure 6.

Table 13 - Voice Gateway Longitudinal-to-Metallic Balance Requirements

Frequency (Hz)	Minimum Balance (dB)	Average Balance (dB)
200	58	63
500	58	63
1 kHz	58	63
3 kHz	53	58

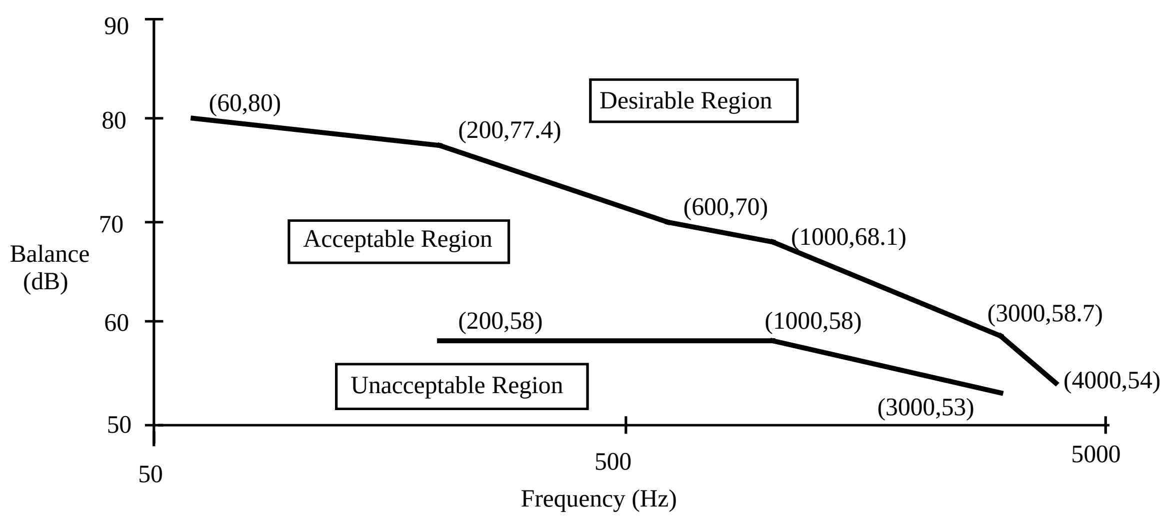


Figure 6 - Longitudinal Balance Limits

9.2.2 Metallic-to-Longitudinal (Transverse Balance) Balance

9.2.2.1 Definition

The transverse balance is defined as:

$$\text{Transverse Balance (dB)} = 20 \log |Vm / Vs|$$

where Vs is the longitudinal rms voltage produced across a longitudinal termination Z_1, and Vm is the metallic rms voltage across the tip-and-ring interface terminals of the voice gateway.

Transverse balance (metallic-to-longitudinal balance) is specified in IS-968 to ensure that a metallic signal is not converted into a longitudinal signal that could cause excessive noise in other pairs of a multi-pair cable.

9.2.2.2 Requirements.

The Voice Gateway should meet the minimum transverse balance requirements as given in Table 14a for analog interfaces and Table 14b for digital service interfaces, when measured as described in Annex A.8.2.

(1) Analog Interfaces

The criteria given in Table 14a should be met for all possible combinations of through transmission paths between analog CO trunk and station line interfaces of the voice gateway.

Table 14a - Transverse Balance Requirements for Analog Interfaces

Interface	State	Frequency Range	Minimum Balance (dB)
CO Trunk-Loop Start	On-hook	200 Hz to 1 kHz	60
		1 kHz to 4 kHz	40
	Off-hook	200 Hz to 4 kHz	40
CO Trunk-Ground Start	Off-hook	200 Hz to 4 kHz	40
Reverse Battery (DID)	Off-hook	200 Hz to 4 kHz	40
OPS Line	Off-hook	200 Hz to 4 kHz	40

(2) Digital Interfaces

Table 14b - Transverse Balance Requirements for Digital Interfaces

Interface	Frequency Range	Minimum Balance (dB)
1.544 Mbps Digital Trunk	15 kHz to 1.544 MHz	36

9.3 Crosstalk

9.3.1 Requirement

The crosstalk coupling loss for every port-to-port connection, over the 200 to 3400 Hz frequency band, should comply with the following criteria:

(1) The crosstalk coupling loss between any established connection through the voice gateway and at least 95 percent of all other through connections should be at least 75 dB, and it is desirable that this loss be at least 80 dB.

(2) The crosstalk coupling loss between any established connection through the voice gateway and any other through connection should be at least 70 dB.

Compliance with the crosstalk coupling loss recommendations should be determined as described in Annex A.9.

9.4 Quantization Distortion

Ninety-five percent or more of all A/D and D/A connections should comply with the recommendations in Table 15, where the input signal is a 1004 Hz sinewave and the output distortion is measured using C-message weighting.

Table 15 - Voice Gateway Quantization Distortion Requirements

Input Signal Level (dBm)	Input/Output Level Ratio (dB)
0 to -30	35
-40	29
-45	25

9.5 Single-Frequency Distortion

Ninety-five percent or more of all A/D and D/A connections in each connection category should comply with the following distortion limit:

- For input signals at a constant 0 dBm level and any single frequency in the range of 0 to 12 kHz, the corresponding output signal power level at any other single frequency should not exceed -28 dBm.[2]

[2] The -28 dBm limit is dependent upon the characteristics of the transmit and receive filters of the Voice Gateway. In the 0 to 3400 Hz frequency range, the limit value is influenced by the characteristics of the receive filter; in the 3.4 to 4.6 kHz range the limit value is dependent upon both transmit and receive filters; and in the 4.6 to 12 kHz range, the limit value is dependent upon the characteristics of the transmit filter and should be -32 dBm.

10 Other Impairments

The following requirements apply to voice gateways intended to pass voiceband data. The following requirements are given, in addition to the voice requirements, to verify that the voice gateway will function in a manner that will not be seen as an impairment to the performance of voiceband data modems.

10.1 Intermodulation Distortion

Intermodulation or harmonic distortion is caused by nonlinearities present in the electric-to-electric transfer function of the voice gateway. This form of distortion is of primary concern to the transmission of data.

Intermodulation distortion is measured using the four-tone method that employs two pairs of equal-level tones transmitted at a total, composite power level of -13 dBm. One pair consists of the frequencies 857 and 863 Hz; the second pair uses the frequencies 1372 and 1388 Hz.

Intermodulation distortion is measured as the second- and third-order products resulting from the application of the four tones. The second- and third-order products are denoted as R2 and R3, respectively. R2 is the average power level in the 503-to-537 Hz and 2223-to-2257 Hz frequency bands, expressed in dB below the received power level. R3 is the total power level in the 1877-to-1923 Hz frequency band, expressed in dB below the received power level.

At input port signal power levels other than 0 dBm, the four-tone signal power levels should be shifted by a value that corresponds to the difference between the signal level at the interface and 0 dBm. Since the R2 and R3 products are expressed in dB below the received signal level, their values are not affected by non-zero reference signal levels.

The intermodulation distortion limits on 95 percent or more of all connections should meet or exceed the values given in Table 16.

Table 16 - Intermodulation Distortion Limits for Voice Gateways

Distortion Limits (dB below received level)	
R2	R3
46 dB	56 dB

10.2 Envelope Delay

10.2.1 Definitions

Envelope Delay (ED) of a system is the propagation time through the system of a low-frequency sinusoidal envelope of an amplitude modulated sinusoidal carrier. If the frequency range of interest is denoted by R[3], the carrier frequency is varied throughout R to obtain the ED as a function of frequency. The carrier is 50 percent amplitude-modulated with a sinusoidal signal of frequency 83.3 Hz.

Relative Envelope Delay (RED) is the difference between the ED at a given frequency f and the global minimum ED within the range R.

10.2.2 Relative Envelope Delay (RED) Requirements

(1) Station Interface-to-Trunk Interface and Trunk Interface-to-Trunk Interface

On 95 percent of the connections of this type

- in the frequency range from 800 through 2700 Hz, the RED curve should lie below curve A in Figure 7;
- it is desirable that in the frequency range from 500 through 3000 Hz, the RED curve lie below curve B in Figure 7.

(2) Station Interface-to-Station Interface

The requirements for this connection allow twice the RED of those in 10.2.2(1). More precisely, let A' be a curve obtained by multiplying by 2 each ordinate of the curve A in Figure 7 and let B' be a curve obtained by multiplying by 2 each ordinate of the curve B in Figure 7. Then the RED requirements for connections of this type are obtained from those in 10.2.2(1) by substituting A' for A and B' for B.

[3] A general symbol R is used here because the frequency range of interest may change with application. For example, for mandatory requirements of this section R = [800 Hz, 2700 Hz], while for objective requirements R = [500 Hz, 3000 Hz].

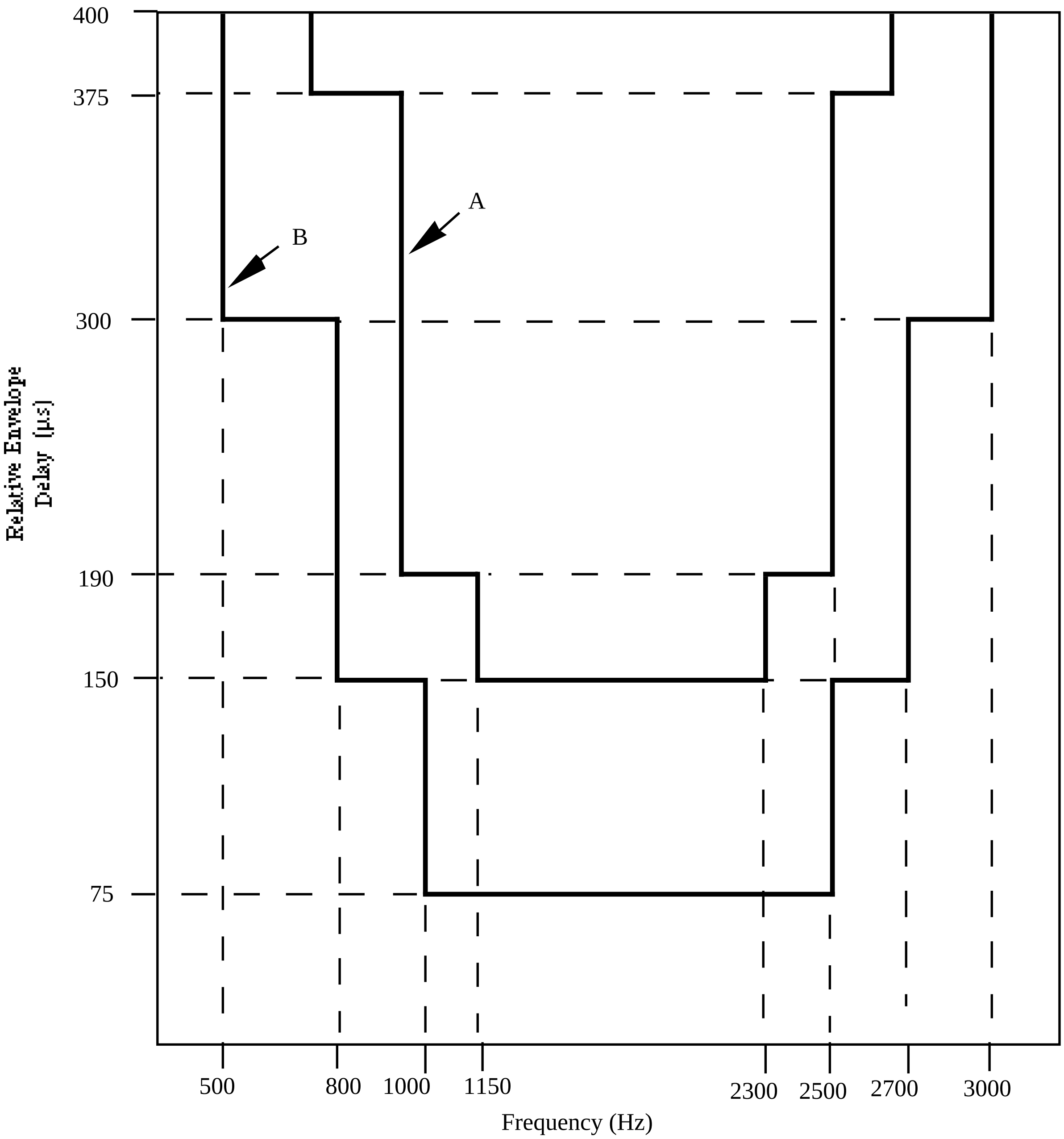


Figure 7 - Relative Envelope Delay vs. Frequency

10.3 Impulse Noise

The following impulse noise limits should be met under fully loaded busy-hour voice gateway traffic conditions.

(1) On 95 percent or more of all connections through each connection category, the impulse noise level should not exceed zero counts above 55 dBrnC over a measurement interval of five minutes.

(2) It is desirable that the impulse noise level not exceed zero counts above 47 dBrnC over a measurement interval of five minutes.

10.4 Jitter

Jitter on a port-to-port connection should not exceed 2 degrees within the 4-to-300 Hz frequency band.

10.5 Gain Hit

There should be no more than one gain hit per hour at a threshold level of 3 dB. If there is more than one hit in a period of 1 hour, the subsequent hour should have zero hits. A gain hit is an incidental modulation resulting in a rapid positive or negative shift of signal gain lasting for a period of at least 4 ms.

10.6 Phase Hit

There should be no more than one phase hit per hour exceeding a threshold of 20 degrees. If there is more than one phase hit in a period of 1 hour, the subsequent hour should have zero hits. A phase hit is an incidental modulation resulting in a rapid positive or negative shift of signal phase lasting for a period of at least 4 ms.

10.7 Dropout

There should be no more than one dropout per hour exceeding a threshold of 6 dB. If there is more than one dropout in a period of 1 hour, the subsequent hour should have zero dropouts. A dropout is a negative gain hit lasting a period of at least 10 ms.

10.8 Peak-to-Average Power Ratio

The voice gateway should pass a signal, analog port-to-analog port, with a peak-to-average power ratio (P/AR) of 95.

P/AR = 100 x ((2Ep/EFWA)-1), where Ep is the normalized peak and EFWA is the normalized full rectified average of the envelope.

11 Signal Levels

The voice gateway should comply with TIA/EIA/IS-968, Section 4.5, for the following signal level limitations:

(1) In-band Signal Power Limits
  (a) Internal Signal Sources Not Intended for Network Control Signaling
  (b) Internal Signal Sources Intended Primarily for Network Control Signaling
  (c) Through Transmission
  (d) Idle State Circuit Stability for Tie Trunks
  (e) Metallic Signal Power at Frequencies in the range 3995 to 4005 Hz
  (f) Longitudinal Voltage in the 100- to 4000 Hz Frequency Range

(2) Out-of-Band Signal Voltage Limits
  (a) Metallic Voltage
  (b) Longitudinal Voltage

The above listed signal limitations should apply to:

(1) Analog Trunk interfaces (ground start, loop start, DID)

(2) OPS Lines

(3) Analog Tie Trunks

(4) Digital Trunk interface (ground start, loop start, DID, ISDN Basic Rate and Primary Rate) with encoded analog contents

(5) Digital OPS Lines with encoded analog contents

(6) Digital Tie Trunk interface with encoded analog contents

Annex A (informative) - Measurement Guidelines

This annex is informative only and is not part of this standard

A.1. Gain Ripples in the Measurement Path

In any digital voice gateway, connections from a 2-wire analog port to a 2-wire analog port will constitute a closed-loop feedback system. The feedback signal will cause ripples in the net through-gain response of a 2-wire to 2-wire connection. If care is not exercised, the ripple effect will influence measurement accuracy. The following two techniques for avoiding ripple influence on measurements are suggested:

(1) Perform test measurements on a 2-wire to 4-wire basis. This approach eliminates the feedback signal. Where appropriate, the requirements contained in these sections have been divided into transmit and receive portions to facilitate this approach. It will be necessary to employ a digital test meter and designated digital test sequences for these types of measurements.

Alternatively, a half-channel test, in which the two directions of transmission are terminated within the switching fabric, may be used (see Section 8).

(2) Maintain a high-quality impedance match at each 2-wire to 4-wire interface to minimize the feedback signal. This approach requires use of a test impedance that closely matches the hybrid balance impedance. To satisfactorily reduce the ripple caused by the feedback signal, a hybrid balance of 25 dB (or greater) should be maintained at each 2-wire to 4-wire interface in the test connection.

A.2. Transmission Level Translation

Each of the requirements in this section has been written with respect to the zero-level point in the switch. A 0 dBm0 signal at this point will decode to 0 dBm, or 1 mW in 600 Ω.

In many cases, the interface level will be different from the zero-level point due to the losses introduced to meet the loss plan. In these cases (unless otherwise stated) the appropriate interface loss should be included when determining compliance with the requirements.

See Annex B.6 for further information on zero-level points, the definition of 0 dBm0, and transmission level translation.

A.3. Dial-up Port for Trunk Testing

To avoid the gain variations that can occur among different terminations, it is recommended that a special dial-up port be designated for trunk testing. However, it should be recognized that even when using a single termination to test all trunks, consideration for gain tracking variation in the dial-up port must be included. The gain tracking variation will occur because different trunk losses will cause different signal levels at the dial-up port. The tracking variation will be limited by the tracking error requirements at the various interfaces (see Section 7.4)

A.4. Digital Test Port Availability

For circuit and line-up purposes, the voice gateway should have provisions for a test port, or equivalent, that enables zero-level point testing at a digital interface.

A.5. Hybrid Balance

The measurement technique described in this section is applicable to line or trunk units that connect to 2-wire analog interfaces.

Examples of test arrangements for testing for compliance with hybrid balance requirements are shown in Figure A1 (full-channel method) and Figure A2 (half-channel method).

When using the full-channel method (Figure A1), the impedance of the signal generator, Z_G, and the signal detector, Z_O, should match the impedance of the 4-wire port.

When using the digital half-channel method (Figure A2), the detector and generator should be equivalent digital instruments.

An analog half-channel method may also be used (similar to Figure A2). The tests can be made via a line or trunk unit access point, or a path can be set up between the line or trunk unit and a 4-wire interface. If test access is from an analog interface, the generator impedance (Z_G) and the detector impedance (Z_O) should match the interface impedance.

Note: A Return Loss Measuring Set (RLMS), conforming to ANSI/IEEE Standard 743-1995 [Ref. 11] is recommended as the signal generator and detector shown in Figure A1 (or an equivalent digital instrument for Figure A2).

The steps for measuring the hybrid balance are :

1. Measure the 2-to-4 wire and 4-to-2 wire loss through the hybrid at 1004 Hz, with the 2-wire port terminated in the appropriate reference impedance.
2. Measure the 4-wire to 4-wire loss (echo return loss) through the hybrid at frequencies over the range of 200 to 3400 Hz, with the 2-wire port terminated in the appropriate reference impedance.
3. Subtract the sum of the losses determined in Step 1 from the values measured in Step 2 to determine the hybrid balance over the specified frequency range.

A.6. Input Impedance

Examples of test arrangements for testing compliance with the input impedance requirements are shown in Figures A3 through A6.

The steps for measuring the input impedance are :

1. Terminate the 4-wire analog ports not under test in impedances (Z_T) that match the interface impedances.
2. Terminate the return loss measuring set with the reference impedance (Z_R) for the port under test, as defined in 8.2.1.
3. Measure the single frequency return loss (SFRL) and echo return loss (ERL) over the range 200 to 3400 Hz to determine compliance with the return loss requirements in Table 10 (600 Ω Z_R) or Table 11 (Complex Z_R). See Annex C for the SFRL and ERL definitions.

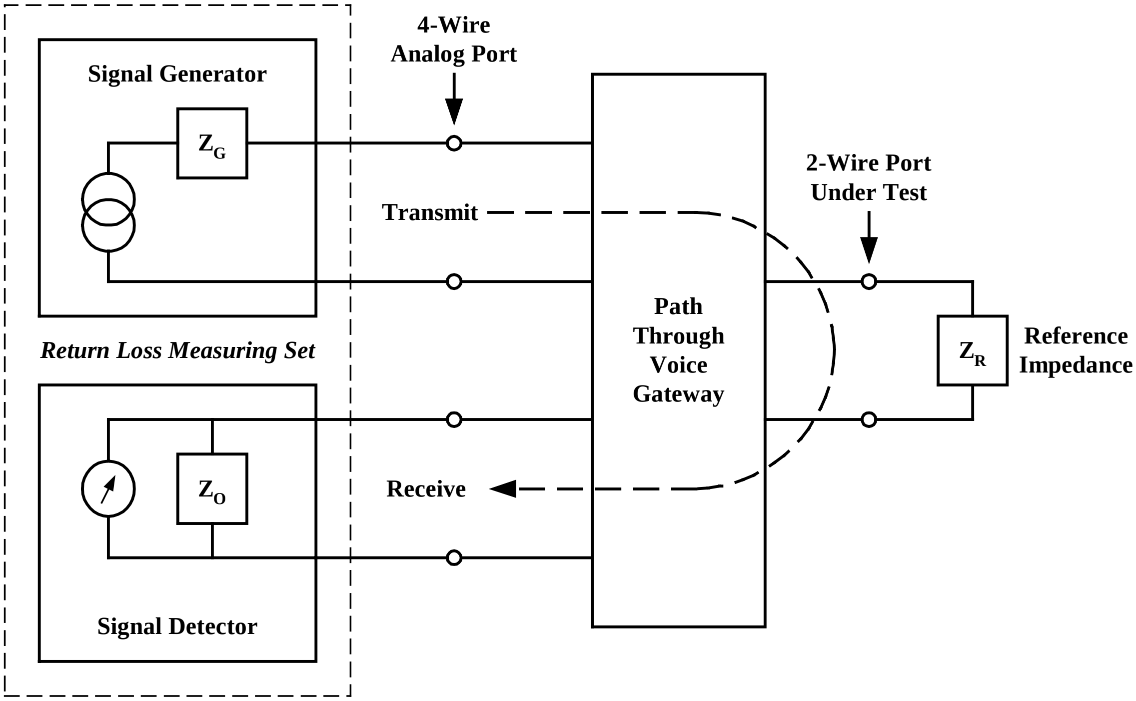


Figure A1 - Equipment Connections for Testing the Hybrid Balance of a 2-Wire Analog Port Using the Full-Channel Method

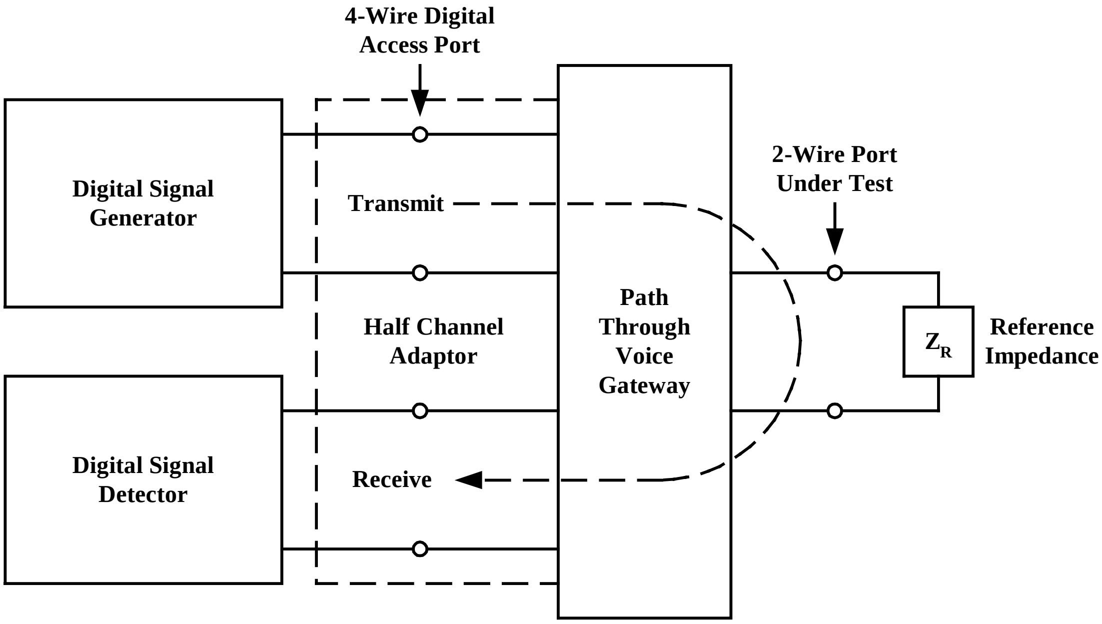


Figure A2 - Equipment Connections for Testing the Hybrid Balance of a 2-Wire Analog Port Using the Digital Half-Channel Method

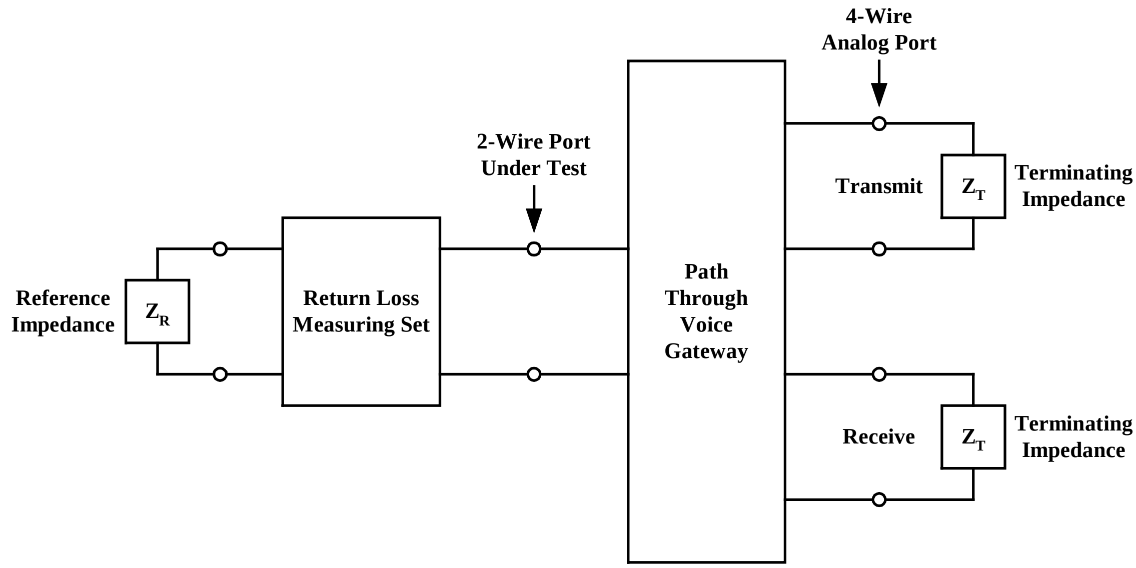


Figure A3 - Equipment Connections for Testing a 2-Wire Analog Port Input Impedance Using the Full-Channel Method

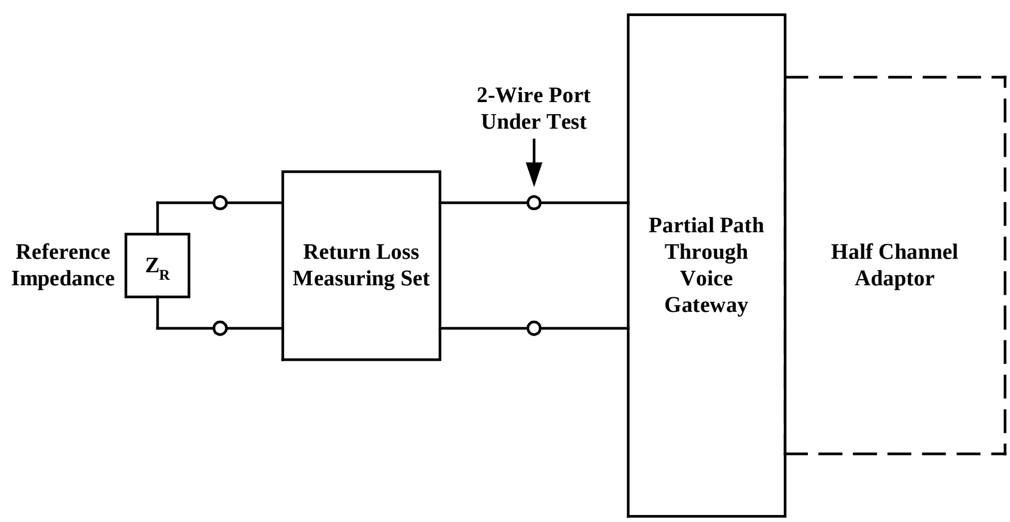


Figure A4 - Equipment Connections for Testing a 2-Wire Analog Port Input Impedance Using the Half-Channel Method

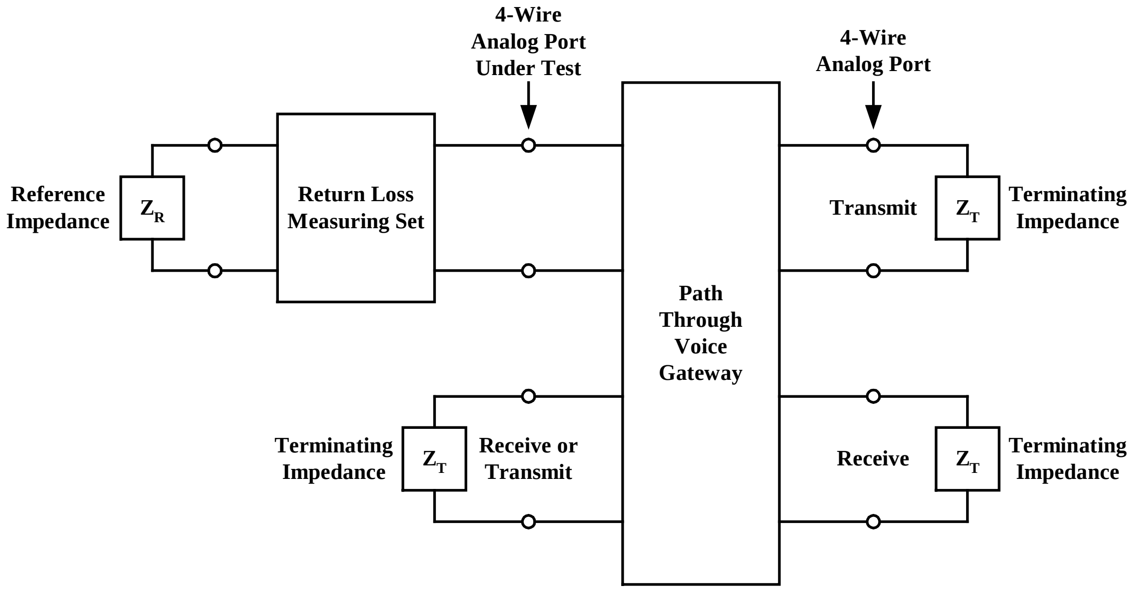


Figure A5 - Equipment Connections for Testing a 4-Wire Analog Port Input Impedance Using the Full-Channel Method

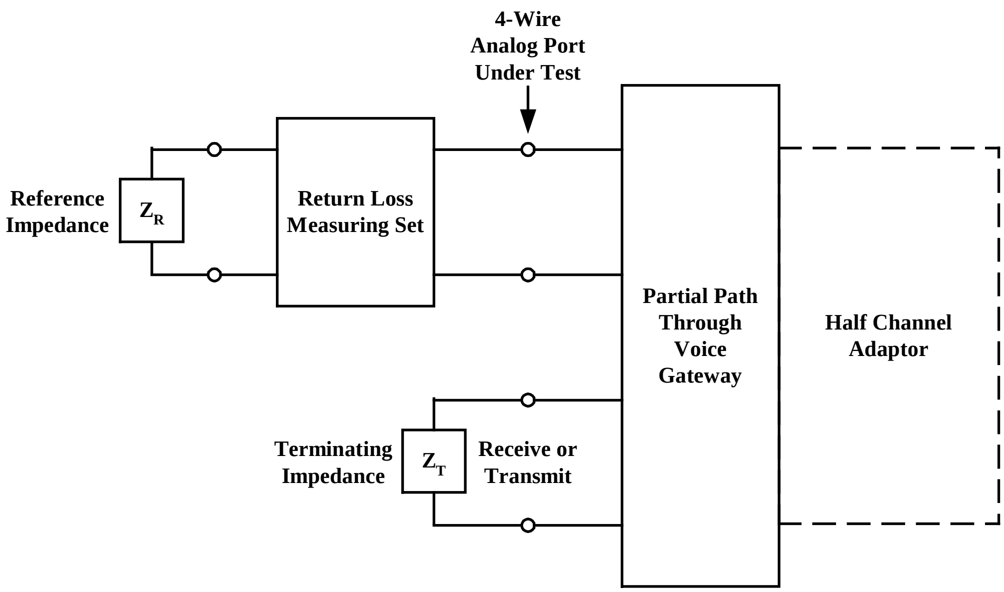


Figure A6 - Equipment Connections for Testing a 4-Wire Analog Port Input Impedance Using the Digital Half-Channel Method

A.7. Idle Channel Noise

Idle-channel noise should be measured with a 3A type noise measuring set or equivalent, with flat or C-message weighting.

When testing a connection, all analog interfaces, except the one(s) being tested, should be terminated with appropriate impedances, and all digital input ports, other than the one(s) being tested, should be supplied with a digital equivalent of zero V.

Compliance with the 13 dBrnC noise requirement for D/A units should be tested by feeding 19 dBrnC of noise in digital form to the D/A unit input. This can be accomplished as shown in Figure A7. With the switch in position 1, the noise generator output should be varied until the D/A test set shows 19 dBrnC. With the switch in position 2, the noise measuring set indicator should not exceed 20 dBrnC.

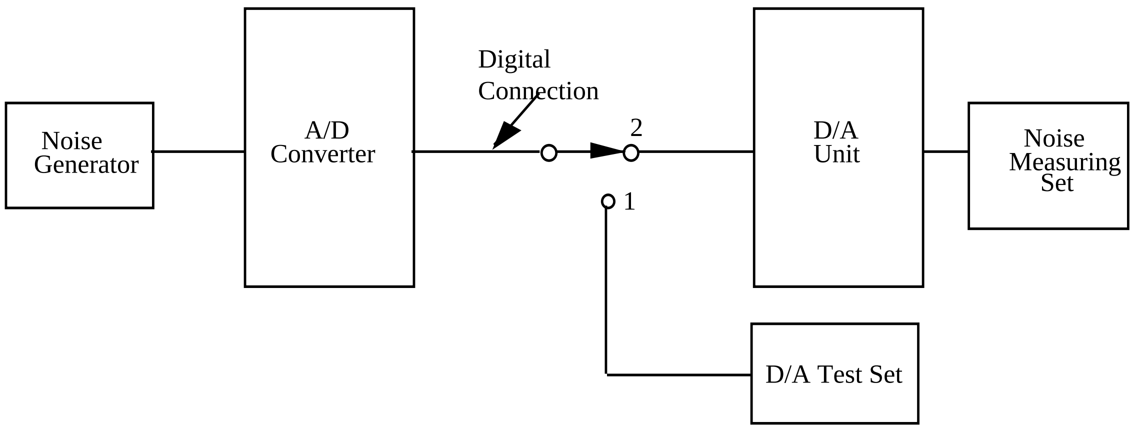


Figure A7 – Idle-Channel Noise Test Arrangement

A.8. Longitudinal Balance

A.8.1 Longitudinal to Metallic Balance

The test procedure is detailed in ANSI/IEEE Standard 455-1992 [Ref. 12]. It is recommended that a frequency selective voltmeter is used. These tests should be conducted for only the off-hook state of the voice gateway.

A.8.2. Metallic to Longitudinal (Transverse) Balance

The test procedure is outlined in the IS-968 standard and TIA/EIA TSB31-B the Part 68 Rationale and Measurement Guidelines [Ref. 13]. Test circuits that satisfies the stated conditions are shown in Figure A8 (Analog), and Figure A9 (Digital).

A metallic voltage should be applied from a balanced source with a metallic impedance Z_0 at suitable points over the frequency range f_1 to $f_{2,}$ and set so that Vm equals E volts when a termination of Z_0 is substituted for the voice gateway. The Z_0 termination should then be replaced by the voice gateway and the longitudinal voltage Vs measured.

The frequency ranges and terminations for each service are defined in Table A1.

The conditions for performing these measurements on the voice gateway are:

(a) All values of dc loop current that the interface under test is capable of drawing when connected to the IS-968 loop simulator circuit for CO trunk interfaces, or the IS-968 line simulator circuit for OPS and DID interfaces.

(b) All reasonable conditions of application of earth ground to the voice gateway under test.

(c) All CO trunk or OPS interfaces not under test, terminated in their appropriate networks or in some cases grounded (see (h), below).

(d) All other than CO and OPS interfaces terminated in circuits appropriate to those interfaces.

(e) Both on-hook and off-hook states applicable to the interface under test.

(f) Impedances of the balance test circuit should be (Z_0) metallic and (Z_1) longitudinal, as defined in Table A1.

(g) Termination of all interfaces not being measured should be follows:

CO trunk	Figure A10
OPS, off-hook	Figure A11
OPS, on-hook	(unterminated).

(h) For station line interfaces designed to isolate longitudinal currents introduced through fully-protected premises wiring or through non-registered equipment, or both, either of the T or R conductors of all ONS station interfaces should be grounded, and the T or R conductors should be both:

  (i) Terminated in an impedance that will reflect correct impedance to the network port to which it is connected for through transmission (see Figure A12).

  (ii) Unterminated.

(i) For station line interfaces not designed to isolate longitudinal currents introduced through unprotected premises wiring, the T&R conductors of all ONS station interfaces should be both:

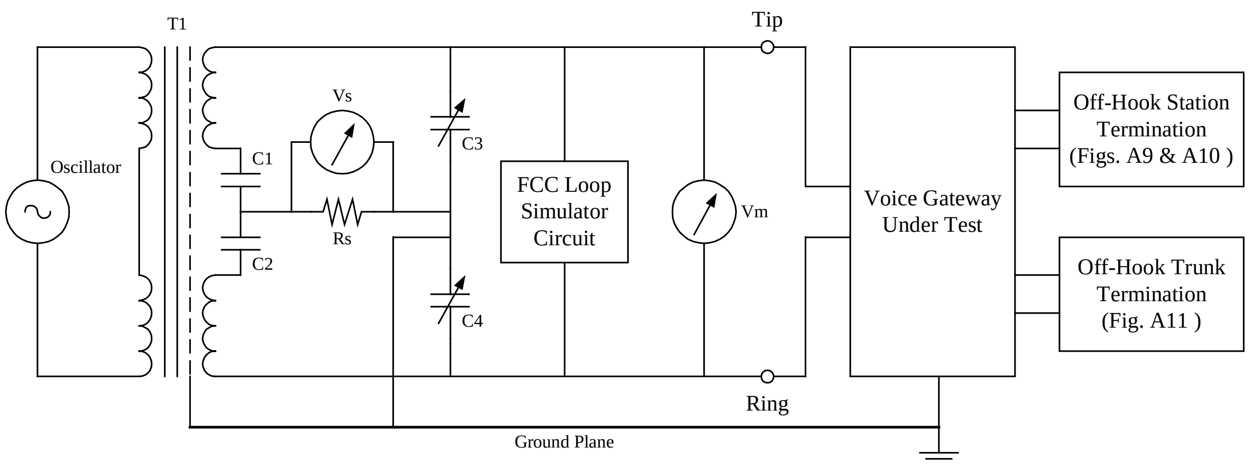
  (i) Terminated in a metallic resistance of 600 Ω and a longitudinal resistance of 150 Ω (see Figure A11).

  (ii) Unterminated.

Table A1 - Termination and Frequency Ranges

	Analog voiceband	1.544 Mbps digital
Longitudinal Termination (Z_1)	500 Ω	90 Ω
Metallic Impedance (Z_0)	600 Ω	100 Ω
Lower Frequency (f_1)	200 Hz	10 kHz
Upper Frequency (f_2)	4 kHz	1.544 MHz
Metallic Voltage for Test (E)	0.775 V	0.316 V

A.8.2.1. Transverse Balance Test (Analog)



Oscillator	Audio oscillator with a source impedance of 600 Ω
T1	WECo 111C or 119E, or ADC 118F, or equivalent
C1, C2	8 µF, 400V dc, matched to within 0.1%
C3, C4	100 to 500 pF trimmer capacitor
Rs	500 Ω
⏚	Voice gateway ground

Figure A8 - Transverse Balance Test Circuit (Analog)

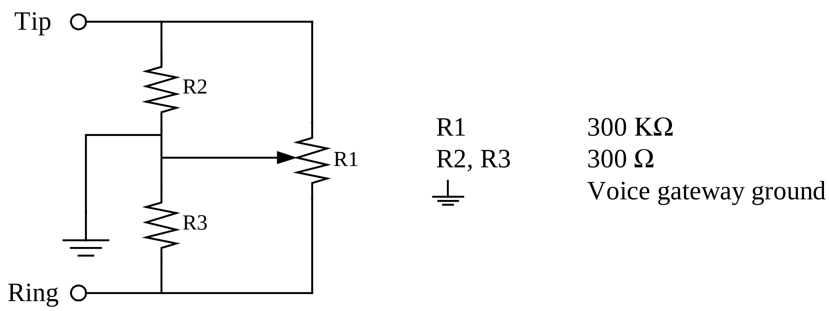


Figure A9 - Off-Hook Termination for OPS or Unprotected ONS Interfaces

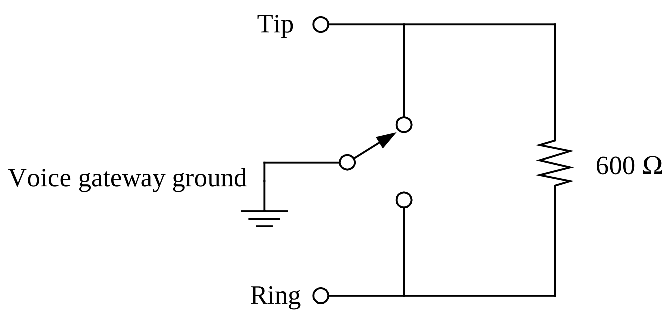


Figure A10 - Off-Hook Termination for Station Interfaces with Longitudinal Current Isolation

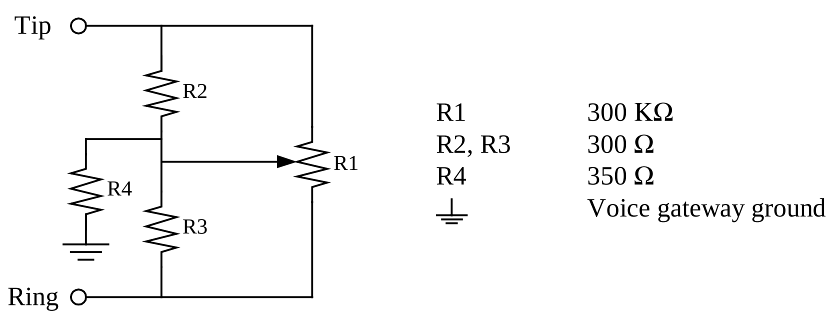
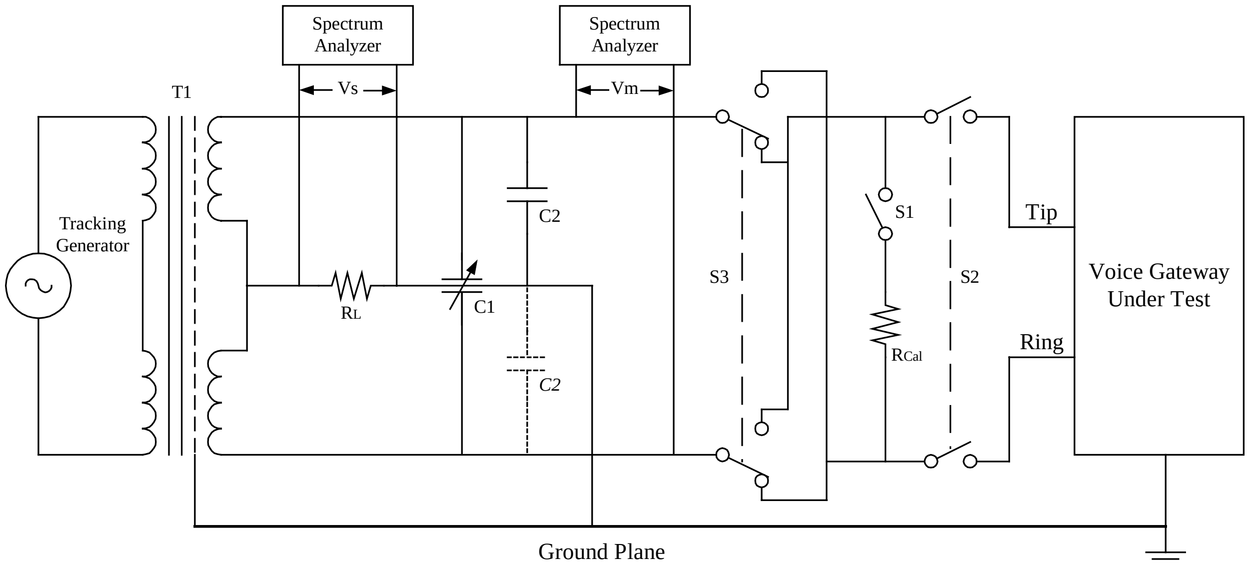


Figure A11 - Off-Hook Termination for CO Interfaces

Notes:

1. *Vm should not be measured at the same time that Vs is measured.*
2. *Use trimmer capacitors C3 and C4 to balance the test circuit to 20 dB greater balance than the equipment standard for all frequencies specified, with a 600 Ω resistor substituted for the voice gateway.*
3. *R1 of the off-hook terminations shall be adjusted to obtain a balance of ≥ 60 dB from 200 Hz to 1 kHz and ≥ 40 dB from 1 kHz to 4 kHz.*
4. *Exposed conductive surfaces on the exterior of the voice gateway should be connected to the ground plane.*
5. *Use the TIA/EIA/IS-968 loop simulator for CO trunks with resistor R1 of the simulator disconnected.*
6. *Use the TIA/EIA/IS-968 line simulator for OPS lines and DID trunks with resistor R1 of the simulator disconnected.*
7. *Use the off-hook termination shown in Figure A10 for station interfaces with longitudinal current isolation as described in Section A.8.2 (h).*

A.8.2.2. Transverse Balance Test (Digital)



Tracking Generator	Tracking generator with a source impedance of 100 Ω
T1	1:1 100 Ω wide-band transformer
C1	20 pF differential capacitor
C2	3 pF composition RF capacitor
R_L	90 Ω non-inductive precision resistor
R_{Cal}	100 Ω non-inductive precision resistor
⏚	Voice gateway ground

Figure A12 - Transverse Balance Test Circuit (Digital)

Notes:

1. *Capacitor C1 should be a dual-stator, air-variable RF capacitor, that maintains a constant capacitance between stators, while providing a variable capacitance from either stator to ground.*
2. *Capacitor C2 may be placed on either line of the test set (as shown), to obtain proper balancing of the bridge.*

A.9. Crosstalk Coupling Loss

A test arrangement for measuring crosstalk coupling loss is shown in Figure A13. It may be desirable when making crosstalk measurements, that a 10 dBrnC level of noise exist at the input interface of the disturbed connection.

Test Procedure

(1) Vary resistors R1 through R4 (referring to Figure A13) to obtain the full loop current ranges specified for the voice gateway interfaces under test, as measured by the 0-200 mA dc meters.

(2) Calibrate the 600 Ω oscillator at 1004 Hz to a level of 0 dBm into a separate 600 Ω resistor. Then, reconnect the oscillator, without changing its level, into the test circuit as shown in Figure A13.

(3) Take readings on both wave analyzers in the disturbed connections:

(a) Select the higher power reading and subtract it from the calibrated level of the oscillator. The result is value A.

(b) Interchange the oscillator and load resistor R5 and repeat (a). The result is value B.

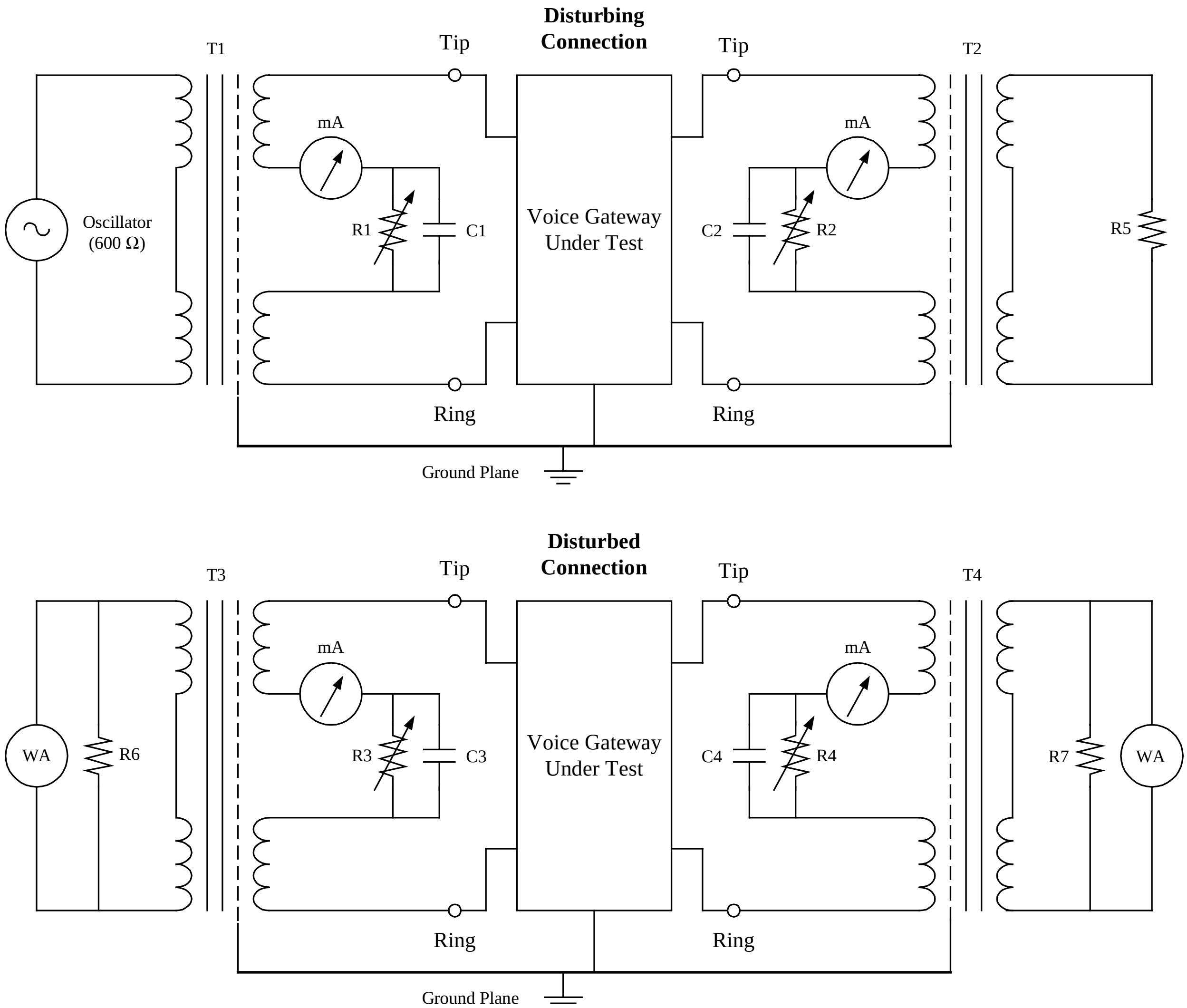
(c) Interchange the disturbing and disturbed connections and repeat (a) and (b). The results are values C and D.

(d) Select the lowest power value of A, B, C, and D. This is the value of crosstalk coupling loss for that pair of connections.

(4) Repeat (3) for all frequencies over the range 200 to 3400 Hz.

(5) Repeat (4) for all loop current values according to (1).

(6) Repeat (5) for all pairs of connections of the voice gateway.



Oscillator	Audio oscillator with a source impedance of 600 Ω
WA	Audio wave analyzer
mA	0 to 200 mA ammeter
T1 to T4	WECo 111C or 119E, or ADC 118F, or equivalent (Zin = Zout)
R1 to R4	2 kΩ adjustable
R6 to R7	600 Ω
C1 to C4	10 µF, 400V dc

Figure A13 - Crosstalk Coupling Loss Test Circuit

Annex B (informative) - Telephony Loss Level Planning Overview

This annex is informative only and is not part of this standard

B.1. Introduction

Telephony loss planning is concerned with the end-to-end loss between the sender and receiver over a telephony network.

It is called a loss plan, as the primary purpose is to approximate the free air loss between a talker and listener in a normal conversation. A secondary purpose is to control echo due to impedance mismatches in connections with long delays.

The loss plan is also related to the optimization of signal levels in equipment involved in the end-to-end connection, and to the provisions of IS-968 regarding the prevention of network harm.

B.2. Send and Receive Levels

The objective of a telephone connection is to simulate a 1 meter free air path between two talkers. This simulation involves several objective and subjective factors that are not present in the 1 meter air path. These include monaural listening, narrowband frequency response, the preferred listening level and others. For any telephone connection, the optimum OLR to achieve the preferred listening level is 10 dB. In a digital connection, the network loss is zero; therefore, the required loudness ratings are adjusted in the send and receive sections of the digital telephone set.

The send and receive levels of a telephone relate the conversion of acoustic pressure to electrical power and vice versa. The acoustic pressure units are in dBPa (Pascals), and the electrical power units are in dB mW.

Pressure is measured in Newtons per square meter (Pascals), and the relationship between dBSPL and dBPa is shown below.

dBSPL	dBPa	
94	0	One Pascal
89.3	-4.7	Average speech level
0	-94	Lower limit of human hearing

B.2.1. Telephone Equipment Loudness Ratings

The loudness ratings of a telephone are the unit-less acoustic-to-electrical-to-acoustic conversion factors as shown in Figure B.1. As the acoustic and electrical units are both relative levels in dBs, the conversion factors are also in dBs

.

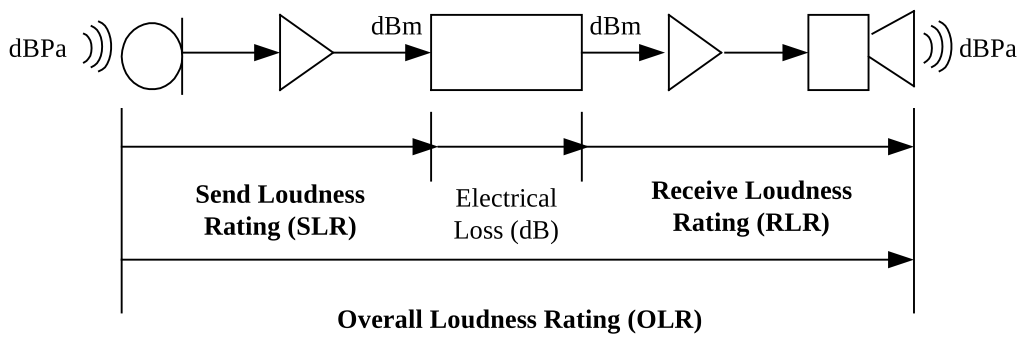


Figure B1 – Terminal Loudness Ratings

The loudness ratings of the three telephone ports defined in this standard are shown in Table B1.

Telephone Port	SLR (dB)	RLR (dB)	Notes
OPS	11	-3	1
ONS	8	-6	2
DGS	8	2	3

Table B1 – Telephone Loudness Ratings

Notes:

1. *The OPS loudness ratings are representative of 2500-type analog telephones operating on 26 gauge/2.75 km loops with normal battery feed and impedance characteristics, as measured at a PSTN end office or voice gateway OPS port. See PN-3-4350.110 (to be published as ANSI/TIA/EIA-470-110-C) for further details.*
2. *The ONS loudness ratings are representative of 2500-type analog telephones operating on short loops with the typical current-limited battery feed and 600 Ω impedance characteristics of voice gateway ONS ports. See PN-3-4350.110 (to be published as ANSI/TIA/EIA-470-110-C) for further details.*
3. *The DGS loudness ratings of SLR = 8 dB and RLR = 2 dB conform to the requirements specified in ANSI/TIA/EIA-810-A.*

B.2.2. Overall Loudness Ratings

The Overall Loudness Rating (OLR) of a connection is the sum of the sending terminal SLR, any system or network loss, and the receiving terminal RLR. This is illustrated in Figure B2 for set-to-set calls within a voice gateway.

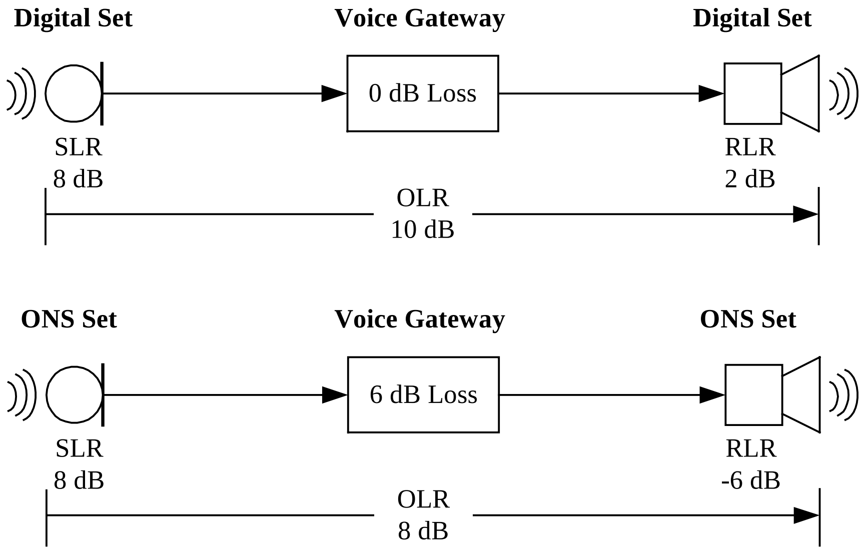


Figure B2 – Telephone-to-Telephone Overall Loudness Ratings

B.2.3. Optimum Overall Loudness Ratings

Figure B3 shows a plot of OLR versus R-Value using the E-Model. The majority of the OLR values in Table 2 are at or above an R-Value of 90, which puts them in the 'very satisfied' category. See TIA/EIA TSB32-A for information on use of the E-Model.

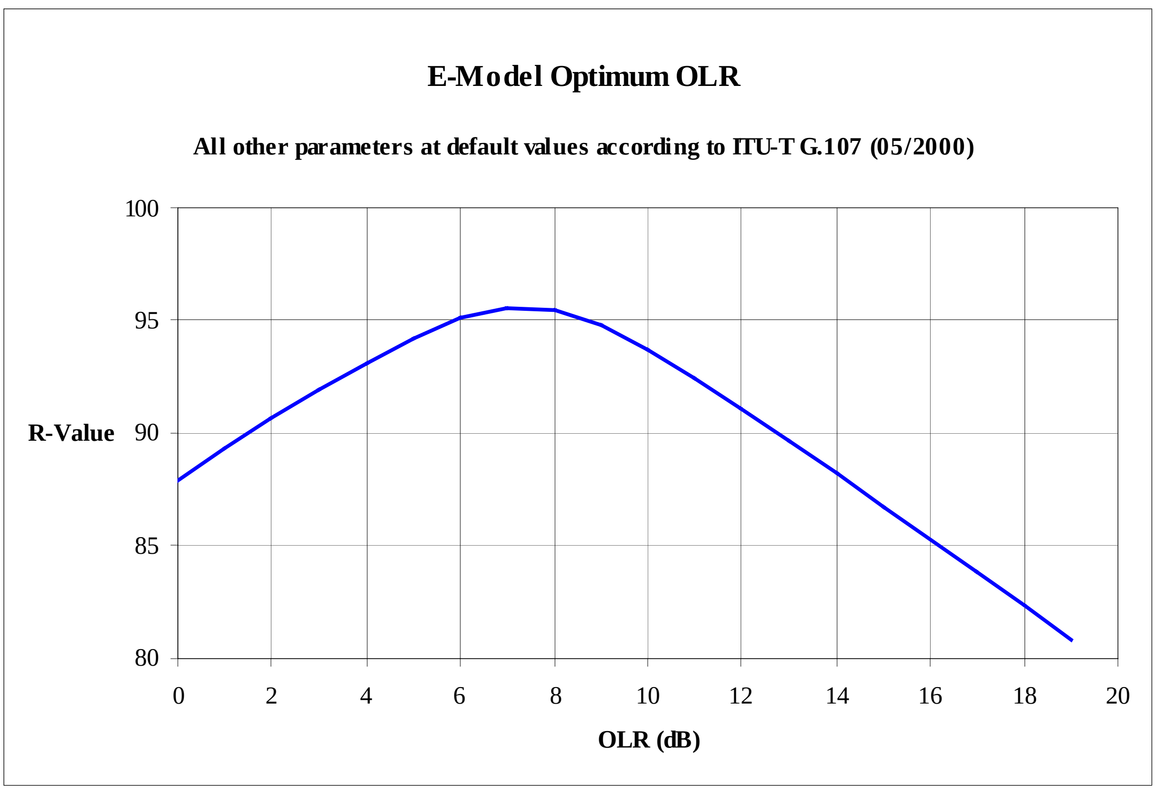


Figure B3 – E-Model Optimum Overall Loudness Rating

B.2.4. Network Interface Equivalent Loudness Ratings

The network interface equivalent loudness ratings (ELRs) are derived from the combination of terminal loudness ratings and nominal network losses. The example in Figure B4 shows the derivation of the network interface ESLR & ERLR for the ATT analog network interface:

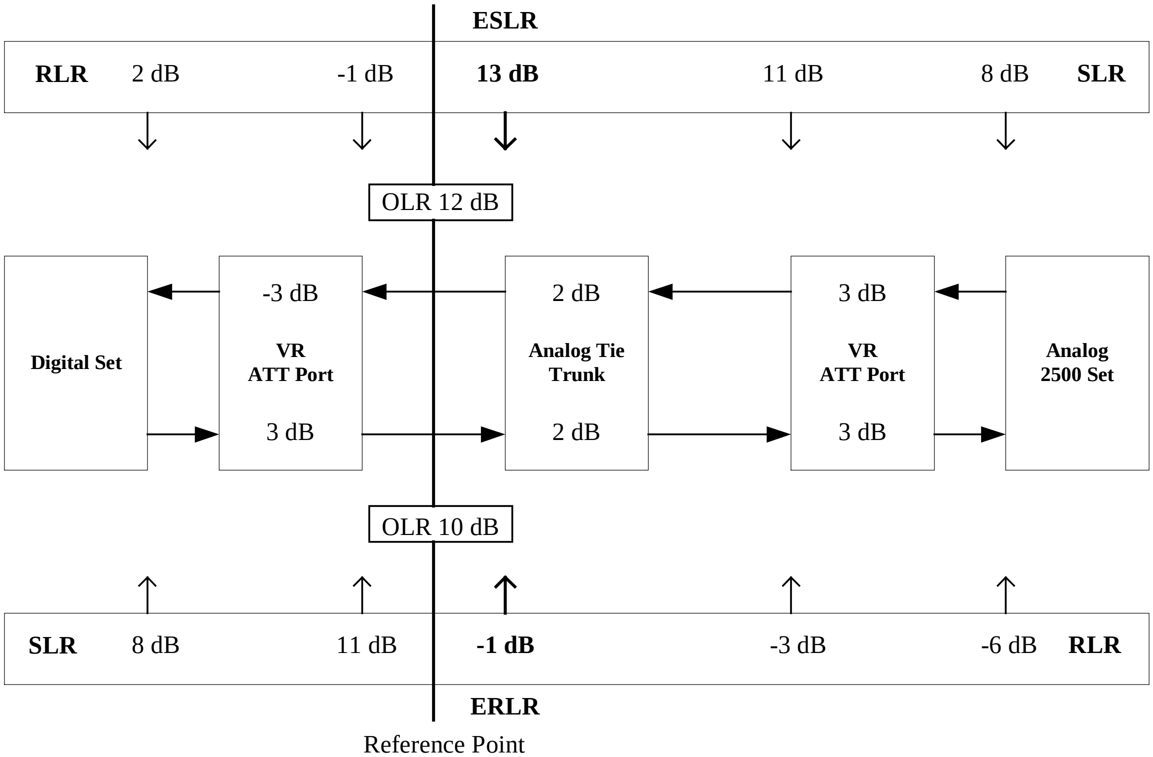


Figure B4 – ATT Network Interface Equivalent Loudness Ratings

B.3. Port-to-Port Loss Allocation

It should be noted, that the actual allocation of the port-to-port loss to send and receive direction directly influences the available dynamic range of the PCM coding scheme. This may lead to substantial impacts on speech transmission quality as perceived by the user.

Care should be taken to ensure that excessive input gain or loss does not cause either overload, or a poor signal-to-noise ratio, at the zero-level point.

Example:

The ONS to OPS loss is specified as 3 dB.

(Note: This is a loss plan, therefore gains are negative.)

This could be implemented (in an extreme case) as an ONS input loss of -9 dB (9 dB gain), and an OPS output loss of 12 dB. The overall loss would be 3 dB, but the effective SLR at the zero-level point (ZLP) would be -1 dB (ONS SLR = 8, loss = -9).

At an average talker level of 88 dB SPL, the average power level at the ZLP would be approximately -3 dBm. The codec overload level is +3 dBm, and as voice peaks are typically 10 dB higher than the average, the peaks would be at +7dBm, resulting in clipping.

Conversely, an ONS input loss of 9 dB, and an OPS output loss of –12 dB, would result in lower power at the ZLP, and a reduction in the signal-to-noise ratio.

B.4. DTMF Overload on Analog Trunks

There is a potential for codec overload on the analog trunk interface when using the FXO to WAN setting, and in-band DTMF signaling is used for voice mail access, credit card verification, etc. This is only likely to happen in the situation where both the subscriber doing the signaling and the voice gateway are close to the central office.

Given that some subscribers are going to be located close to the central office, the decision on when to introduce additional loss in the voice gateway analog trunk interface has to be based on the distance of the voice gateway from the central office. The generally excepted definition of a short loop is 2 km or less (less than approximately 3 dB loss), and therefore the recommendation is that an additional 3 dB of loss be inserted in the analog trunk interface for short loops when making FXO to WAN connections. The simplest way to do this is to use the FXD/WAN setting. This will also have the added advantage of reducing the OLR of the connection to a more comfortable level.

Figure B5 illustrates the subscriber to voice gateway losses, and the resultant DTMF level at the analog trunk interface, for both long and short loops. It is assumed that the analog set is transmitting DTMF signals at the maximum level of 0 dBm, rather than the nominal level of -2 dBm.

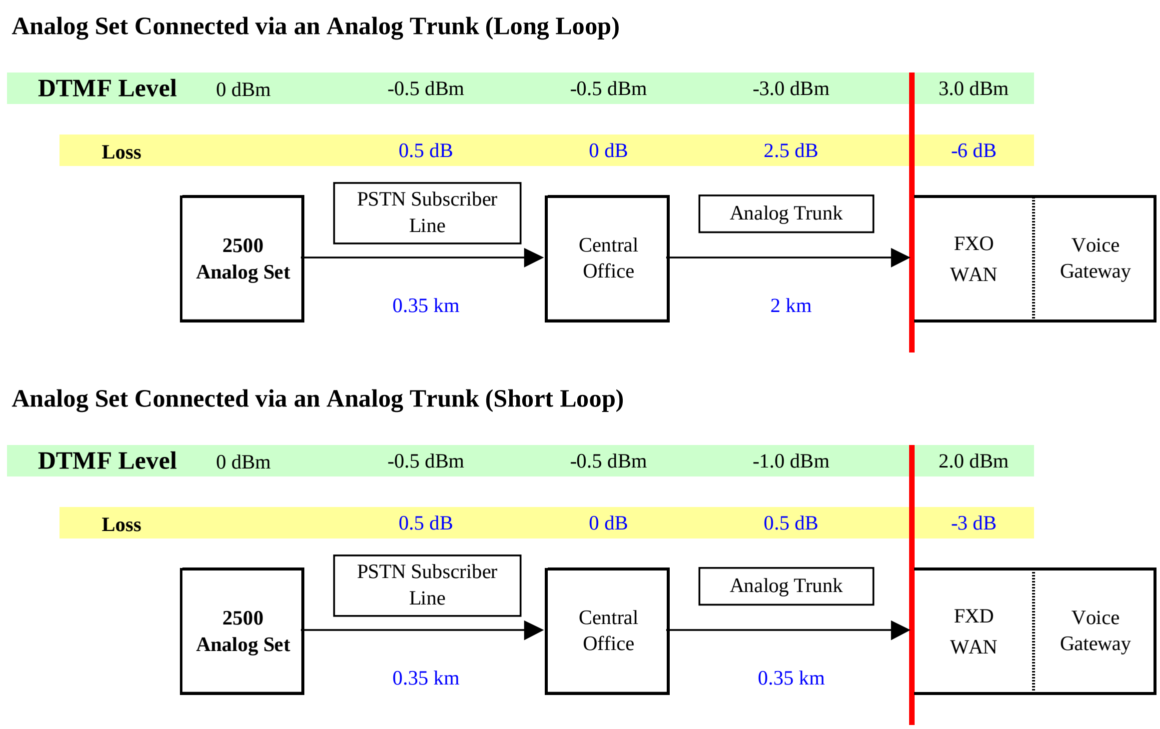


Figure B5. DTMF Levels for Long and Short Loop Analog Trunks

B.5. *Open Loop Loss and Network Stability*

Annex A6 of TIA/EIA/TSB32-A provides an overview of 4-wire loop stability or 'Singing'. The fundamental principle is that a 4-wire loop will oscillate if the sum of all losses and gains around the loop at one single frequency is equal to or less than 0 dB.

The most critical point for stability is during call set-up and release of a connection, when the 2-wire side of the hybrid can be close to open or short circuit, and the Balance Return Loss (BRL) will decreases to values close to 0 dB. Under these conditions the Open Loop Loss (OLL) can reach 0 dB, if there is additional gain in the 4-wire loop.

The basic loop stability model is illustrated in Figure B6 for a simple subscriber-to-subscriber call via a central office.

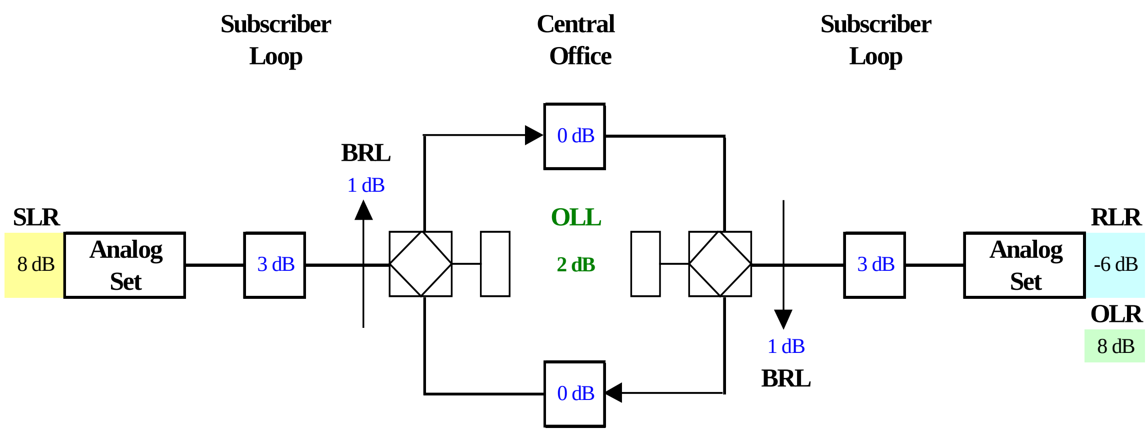


Fig. B6 - Subscriber-to-Subscriber Connection via a Local Central Office

The loop is unconditionally stable in the case above, as the OLL is 2 dB. The central office loss is assumed to be 0 dB (no additional CO loss).

A more complex situation arises when there are multiple 4-wire loops in tandem. This is illustrated in Figure B7 for two central offices connected via two voice gateways over a digital trunk. This is a worse case condition, and should be unconditionally stable.

It is assumed in this case that the COs have inserted an additional 3 dB loss as described in Section 6.4 (2). Figure B7 shows the FXO-to-DAL and DAL-to-FXO losses required to achieve unconditional stability. The 20 dB OLR for the connection would generally be unacceptable, but is a necessary consequence of the need to maintain network stability.

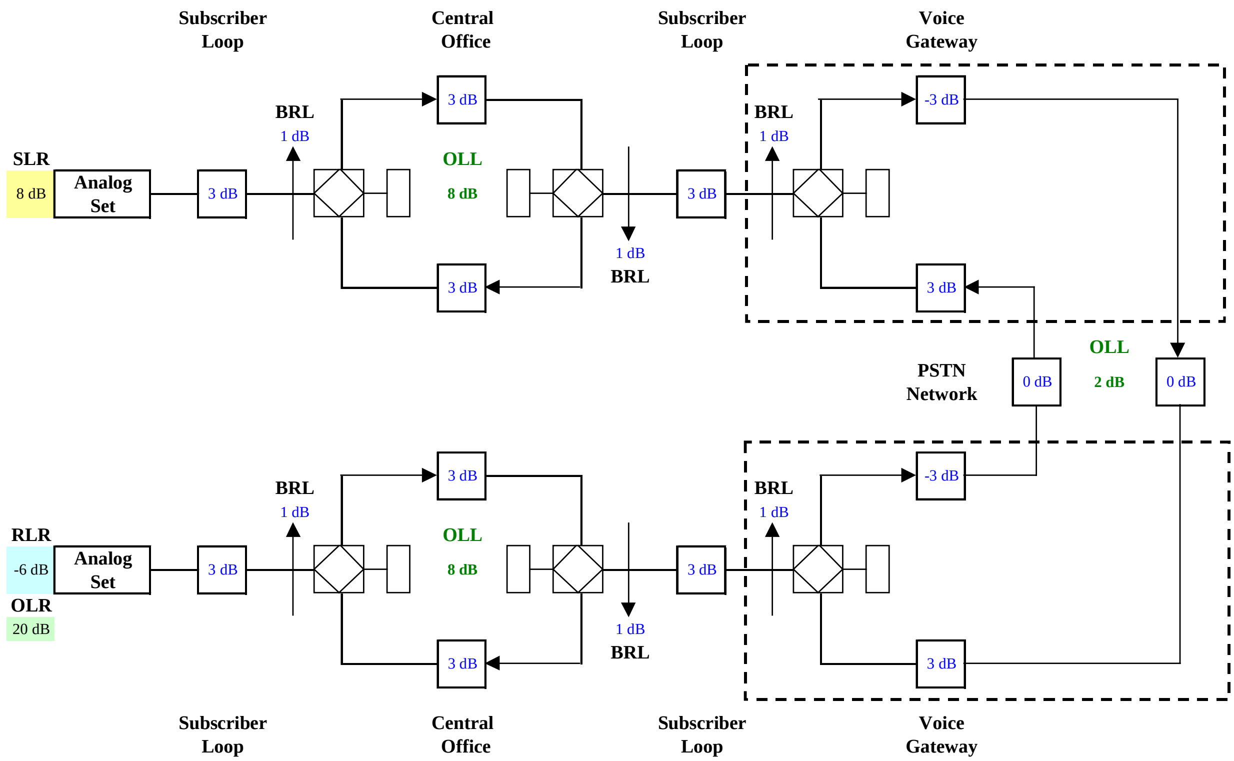


Figure B7 - Subscriber-to-Subscriber Connection via Central Offices and Voice Gateways

B.6. Reference Levels

B.6.1 Zero-Level Point

The zero-level point (ZLP) generally represents the digital (PCM) switching point in a voice gateway. A 0 dBm0 signal at this point will decode to 0 dBm, or 1 mW in 600 Ω.

B.6.2 0 dBm0 Definition

The 0 dBm0 level corresponds to the digital milliwatt (DMW) and is defined as the absolute power level at a digital reference point of the same signal that would be measured as the absolute power level, in dBm, if the reference point was analog.

The absolute power in dBm is defined as 10 log (power in mW/1 mW). When the test impedance is 600 Ω resistive, dBm can be referred to a voltage of 775mV, which results in a reference active power of 1 mW.

0 dBm0 corresponds to an overload level of approximately 3 dBm in the A/D conversion.

B.6.3 Digital Milliwatt

The digital reference level is the Digital Milliwatt (DMW) as defined in ITU-T Recommendation G.711, Tables 5 (A-law) and 6 (μ-law).

A 1 kHz signal at a nominal value of 0 dBm0 will be present at the output of a perfect codec if the periodic PCM code sequence specified in Table 5 or 6 is present at the input of the decoder.

The use of an exact 1 kHz signal can cause problems with some transmission and measuring equipment, so digital periodic sequences representing reference frequencies of 1004 Hz (IEEE) or 1020 Hz (ITU) are generally used instead.

These reference signals can be at either -10 dBm0 or 0dBm0 +/- 0.03 dB.

B.6.4 Transmission Level Translation

The section requirements that are affected by signal level are specified with respect to the zero-level point. The gain or loss from the port interface to the zero-level point has to be taken into account when making measurements.

In the case of input ports, the input level should be increased or decreased by the amount equivalent to the loss or gain from the interface to the zero-level point.

In the case of output ports, the output measurement should have an amount equivalent to the loss or gain from the zero-level point to the Interface, added or deleted.

Note: This amount is not the same as specified in Table 1, as Table 1 is for port-to-port connections, and the port-to-ZLP-to-port losses and gains are defined by the voice gateway manufacturer.

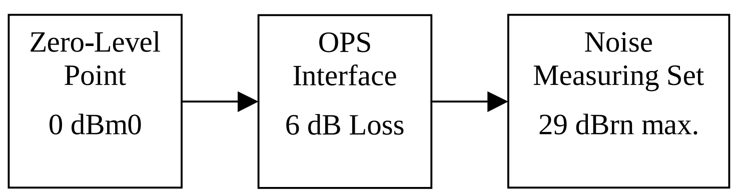
It is generally easier to set up the test connection as an analog-port to digital-port connection, although this is not a requirement.

Example:

Output Interface – Idle-Channel Noise

Requirement: The 3-kHz flat weighted noise should not exceed 35 dBrn on 50 percent of the connections.

Given the case of a DAL to OPS port, and assuming the manufacturer has implement the 6 dB loss in the OPS interface, then the maximum allowable noise level would be adjusted by 6 dB, as shown below:



Annex C (informative) - Loss Definitions

This annex is informative only and is not part of this standard

C.1. Echo Return Loss (ERL)

Echo return loss (ERL) is a weighted average of the return loss values over the frequency range 400 to 3400 Hz. Frequency multiples of 8 kHz must be avoided; the table below shows one convention for avoiding multiples. ERL is calculated as follows:

$$\mathrm{ERL} = -10\log_{10}\left\{\frac{\sum_{i=1}^{N} W(f_i)10^{\frac{-RL(f_i)}{10}}}{\sum_{i=1}^{N} W(f_i)}\right\}\ \mathrm{dB}$$

where

$RL(f_i) =$ Return loss or transhybrid loss, in dB, at frequency f_i,

$W(f_i) =$ Weighting factor at frequency f_i (see the following table).

Frequency fi (Hz)	ERL Weights W(fi)
402	0.0631
602	0.6761
803	1.0000
1004	1.0000
1205	1.0000
1405	1.0000
1606	0.9550
1807	0.7586
2008	0.4467
2208	0.1995
2409	0.0813
2610	0.0324
2811	0.0129
3011	0.0051
3212	0.0021
3413	0.0009

C.2. Return Loss (RL)

Return loss (RL) at an impedance discontinuity in a transmission path is the ratio (in dB) of the power level of an incident signal to the power level of the resulting reflected signal. The general expression for return loss is:

$$\mathbf{RL = 20\ Log\left|\frac{Z_I + Z_R}{Z_I - Z_R}\right| dB}$$

where Z_I and Z_R are the input and reference impedances respectively.

Single-frequency return loss (SFRL) is the lowest value of non-weighted return loss occurring in the frequency range 200 to 3200 Hz.

Single-frequency transhybrid loss (SFTHL) is the lowest value of loss, from the input pair to the output pair of the same 4-wire interface, occurring in the frequency range 200 to 3200 Hz.

C.3. Transhybrid Loss (THL)

Transhybrid loss (THL) is the loss from the input pair to the output pair of the same 4-wire interface.

Annex D (informative) - Pan-European Loss and Level Plans

D.1. Half-Channel Loss and Level Plan

One of the objectives of the voice gateway transmission standard is the harmonization of loss and level planning between the Pan-European and the North American regions. An ETSI/STQ Pan-European standard [Ref. 15] was developed in parallel with this standard, with close liaison between the two standards committees.

The Pan-European standard is limited in scope to a half-channel loss plan, with a global view that this form of standard may be subject to adoption in the future by other regions.

Table D1 shows the voice gateway half-channel loss plan for the Pan-European region, given in equivalent loudness ratings and respective loss.

Table D1. Pan-European Voice Gateway Half-Channel Loss Plan

				WAN Zero-Level Point ↓				
		a	**b**	**c**	**d**	**e**	**f**	**g**
				a + b			**c + d + e**	**8 + d + e**
NA Designation	**European Designation**	**ESLR**	**Tx Loss**	**iSLR**	**ERLR**	**Rx Loss**	**OLR**	**Assumed OLR**
ONS	**L2**	**3**	**5**	**8**	**-8**	**10**	**10**	**10**
DIG	**LD**	**8**	**0**	**8**	**2**	**0**	**10**	**10**
DAL	**KD**	**8**	**0**	**8**	**2**	**0**	**10**	**10**
WAN	**WAN**	**8**	**0**	**8**	**2**	**0**	**10**	**10**
FXO	**K2**	**16**	**-6**	**10**	**5**	**-3**	**12**	**10**
ATT	**M4**	**10**	**-2**	**8**	**4**	**-2**	**10**	**10**

Column **a** shows the ESLR of the telephones and trunks at the connection point to the voice gateway.

Column **b** shows the transmit loss required to achieve the required iSLR at the zero-level point.

Column **c** shows the resulting ESLR (iSLR) at the zero-level point (WAN).

Column **d** shows the ERLR of the receive side.

Column **e** shows the receive loss required to achieve the desirable OLR, based on the ERLR shown in column **d,** and an assumed iSLR of 8 dB**.**

Column **f** shows the resulting OLR

Column **g** shows the assumed OLR, based on an iSLR of 8 dB

Notes:

1. *The OLR values shown in the table are as perceived by the listener, i.e. this is shown as a one-way connection.*

D.2. Full-Channel Loss and Level Plan

The following is extracted from Appendix II of the Pan-European standard:

Historically, separate national transmission plans have been enforced and utilized in European countries. Such national transmission plans were, in general, based on the appropriate ITU-T Recommendations. Therefore, the inter-country, intra-European telephony connections were ruled by the International transmission plan as per ITU-T Recommendations G.101, G.111 and G.121. Hence, there was no reason to issue a Pan-European Loss and Level Plan.

Regulatory treatment of a telephony connection in Europe consist of two parts: regulation of the public network (through the directives on an Open Network Provision) and regulation of the terminal market (through a ”terminal directive”). Both of these regulations are undergoing changes with the effect that national regulatory authorities do not intervene where quality is ensured through effective competition.

The new directive for Radio equipment and Telecommunications terminal equipment (the ”R&TTE” directive) includes a possibility for the Commission to issue regulation regarding voice performance. However as long as the market actors behave in a responsible manner, there will be no EU regulation of voice performance of customer premises equipment connected to a public network.

Regarding regulation of public networks, major changes will take place. Telecommunications services delivered over all types of ”communications infrastructures” will be covered, including CATV and IP networks. Obligations to provide services with adequate quality will remain, however with increased choices allowed regarding quality levels. It is not foreseen that any pan-European level plan will emerge due to regulation of ”communications infrastructure”.

For the telecommunications industry it is however of value to arrive at a common transmission plan for future networks, to ensure successful global communications.

Annex B of the Pan-European standard contains an Excel workbook deriving the Pan-European equivalent loudness ratings, the resultant recommended half-channel loss plan, and also a full-channel loss plan.

The full-channel loss plan is show in Table D2.

Table D2. Pan-European Voice Gateway Full-Channel Loss Plan

Voice Gateway Loss Plan			**L2**	**LD**	**WAN**	**KD**	**K2**	**M4**
			↑	**↑**	**↑**	**↑**	**↑**	**↑**
ONS	**L2**	**→**	**15**	**5**	**5**	**5**	**2**	**3**
DGS	**LD**	**→**	**10**	**0**	**0**	**0**	**-3**	**-2**
WAN	**WAN**	**→**	**10**	**0**	**0**	**0**	**-1**	**-2**
DAL	**KD**	**→**	**10**	**0**	**0**	**0**	**-1**	**-2**
FXO	**K2**	**→**	**2**	**-6**	**-6**	**-6**	**-6**	**-6**
ATT	**M4**	**→**	**8**	**-2**	**-2**	**-2**	**-2**	**-4**

Voice Gateway Loudness Ratings			**L2**	**LD**	**WAN**	**KD**	**K2**	**M4**
		ERLR	**-8**	**2**	**2**	**2**	**5**	**4**
	ESLR	**OLR**	**↑**	**↑**	**↑**	**↑**	**↑**	**↑**
L2	**3**	**→**	**10**	**10**	**10**	**10**	**10**	**10**
LD	**8**	**→**	**10**	**10**	**10**	**10**	**10**	**10**
WAN	**8**	**→**	**10**	**10**	**10**	**10**	**12**	**10**
KD	**8**	**→**	**10**	**10**	**10**	**10**	**12**	**10**
K2	**16**	**→**	**10**	**12**	**12**	**12**	**15**	**14**
M4	**10**	**→**	**10**	**10**	**10**	**10**	**13**	**10**

Annex E (informative) - IP Transmission Impairments

The effects of impairments introduced by IP networks are described in TIA/EIA/TSB116, *Voice Quality Recommendations for IP Telephony*. The effects of impairments are evaluated using the E-model [Ref. 14], which includes the effects of both the terminal equipment (end points) and the transmission network.

IP networks impact two of the impairment parameters; end-to-end delay and packet loss. Unfortunately both these have significant effects on speech quality.

A voice gateway is also likely to perform TDM/IP conversions, and will need to employ echo cancellers if there are any 2-to-4 wire converters on the TDM side of the conversion.

E.1 Delay

End-to-end delay is a combination of the:

- The sending end point encoding and packetizing delay
- The network processing and propagation delay
- The receiving end point jitter buffer, depacketizing, and decoding delay

There is an interaction between the network delay and the receiving end point delay, in that the end point has to compensate for network delay variances via a jitter buffer, and increased variances lead to longer average delay in the end point.

Note: Other processes such as transcoding or encryption will also add delay.

E.2 Packet Loss

Packets can be lost either in the network, or can be discarded by the end point because they are too late (delayed beyond the range of the jitter buffer). In this case there is an interaction between network delay variances and packet loss in the end point.

E.3 Voice Gateways and Network Performance

The purpose of standards for voice transmission is to ensure that voice quality is achieved by adherence to the standards.

In a PBX standard the TDM transmission and switching is a dedicated resource within the PBX framework, and under normal conditions has little or no impact on voice quality.

In a Voice Gateway standard transmission and switching is via IP networks, which can introduce impairments as noted above, and may also be shared by other entities beyond the control of the voice gateway. Ensuring good voice quality will therefore require the use of managed networks and other techniques, which is beyond the scope of this standard.

E.4 Voice Quality of Service

There are a number of standards groups addressing the issue of voice quality of service (QoS). Users of this standard should ensure they are familiar with the work of these groups, as the methods and techniques for managing voice QoS are constantly evolving.

Some of the major groups involved in IP telephony voice QoS are listed below:

TIA Telecommunications Industry Association

www.tiaonline.org

TR-41 *User Premises Telecommunications Requirements*

IETF Internet Engineering Task Force

www.ietf.org

MPLS *Multiprotocol Label Switching*

DIFFSERV *Differentiated Services*

ITU-T International Telecommunication Union - Telecommunication Standardization Sector

www.itu.int/ITU-T

ITU-T SG 12 *End-to-end transmission performance of networks and terminals*

ETSI European Telecommunications Standards Institute

www.etsi.org

TIPHON *Telecommunications and Internet Protocol Harmonization over Networks*

STQ *Speech Processing, Transmission and Quality Aspects*

TIA

TIA