ATIS-0100508.2003(R2013)

Loss Plan for Digital Networks

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ATIS-0100508.2003(R2013), *Loss Plan for Digital Networks*

Is an American National Standard developed by the **ATIS Network Performance, Reliability and Quality of Service Committee (PRQC)**.

Published by

Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005

(R2013)
ATIS-0100508.2003
(Formerly T1.508-2003, Revision of T1.508-1998)

American National Standard for Telecommunications

Loss Plan for Digital Networks

Secretariat

Alliance for Telecommunications Industry Solutions

Approved December 17, 2003

American National Standards Institute, Inc.

Abstract

This standard provides loss plan requirements for digital networks, including Digital End Offices, taking into account different network configurations and elements, and their associated transmission characteristics.

Foreword

The information contained in this Foreword is not part of this American National Standard (ANS) and has not been processed in accordance with ANSI's requirements for an ANS. As such, this Foreword may contain material that has not been subjected to public review or a consensus process. In addition, it does not contain requirements necessary for conformance to the standard.

This document is entitled *Loss Plan for Digital Networks*. This standard provides transmission recommendations for connections between network interfaces (NIs) that are all digital (NI-to-NI), or that are all digital from digital end office-to-digital end office) with digital or analog access lines from appropriate end-user terminals or interconnecting networks.

Footnotes are not officially part of this standard. This standard contains annexes that are for information only and are not considered part of the Standard.

Future control of this document will reside with Accredited Standards Committee — Telecommunications, T1. This control of additions to the specification -- such as protocol evolution, new applications, and operational requirements -- will permit compatibility among U.S. networks. Such additions will be incorporated in an orderly manner with due consideration to the ITU-T layered model principles, conventions, and functional boundaries.

Suggestions for improvement of this standard will be welcome. These should be sent to the Alliance for Telecommunications Industry Solutions, T1 Secretariat, 1200 G Street, NW, Suite 500, Washington DC 20005.

This standard was processed and approved for submittal to ANSI by the Accredited Standards Committee — Telecommunications, T1. Committee approval of this standard does not necessarily imply that all committee members voted for its approval. At the time it approved this standard, the T1 Committee had the following members:

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Loss Plan for Digital Networks

1 Scope, Purpose, and Application

This standard provides loss plan requirements for networks that are all digital from Digital End Office-to-Digital End Office (DEO-to-DEO). Such digital networks can be Time Division Multiplex (TDM) or packet-based. Access lines connected to the DEO may be digital or analog.

The term DEO includes the concept of a DEO Complex, for those instances when the DEO functional components are geographically distributed. The DEO Complex may include the DEO plus Remote Switching Unit (RSUs), Remote Digital Terminal (RDTs), and Optical Network Units (ONUs). The terms *DEO* and *DEO Complex* are used interchangeably throughout this document.

This standard is intended to support services carried on the Public Switched Telecommunications Network (PSTN). These requirements assume that all terminal equipment and networks connecting with the DEO are in compliance with standards pertinent to their transmission performance.

It is intended that these requirements be implemented in an evolutionary manner and not retroactively.

The purpose of this standard is to support the end-to-end transmission performance of voice-grade services by providing loss requirements for DEOs that connect to networks and terminals for which compatible transmission standards have been developed. An additional purpose is to ensure correct and easy interconnection of Incumbent and Competitive Exchange Carrier networks, Exchange Carrier (EC) and Interexchange Carrier (IXC) networks, as well as connections with terminal equipment and interconnecting networks for which compatible standards have been developed.

Figure 1 illustrates the how the DEO Complex connects voice terminals, voice gateways, Enterprise networks, and interconnecting networks. It also details the port/interface acronyms that are used for the loss plan requirements.

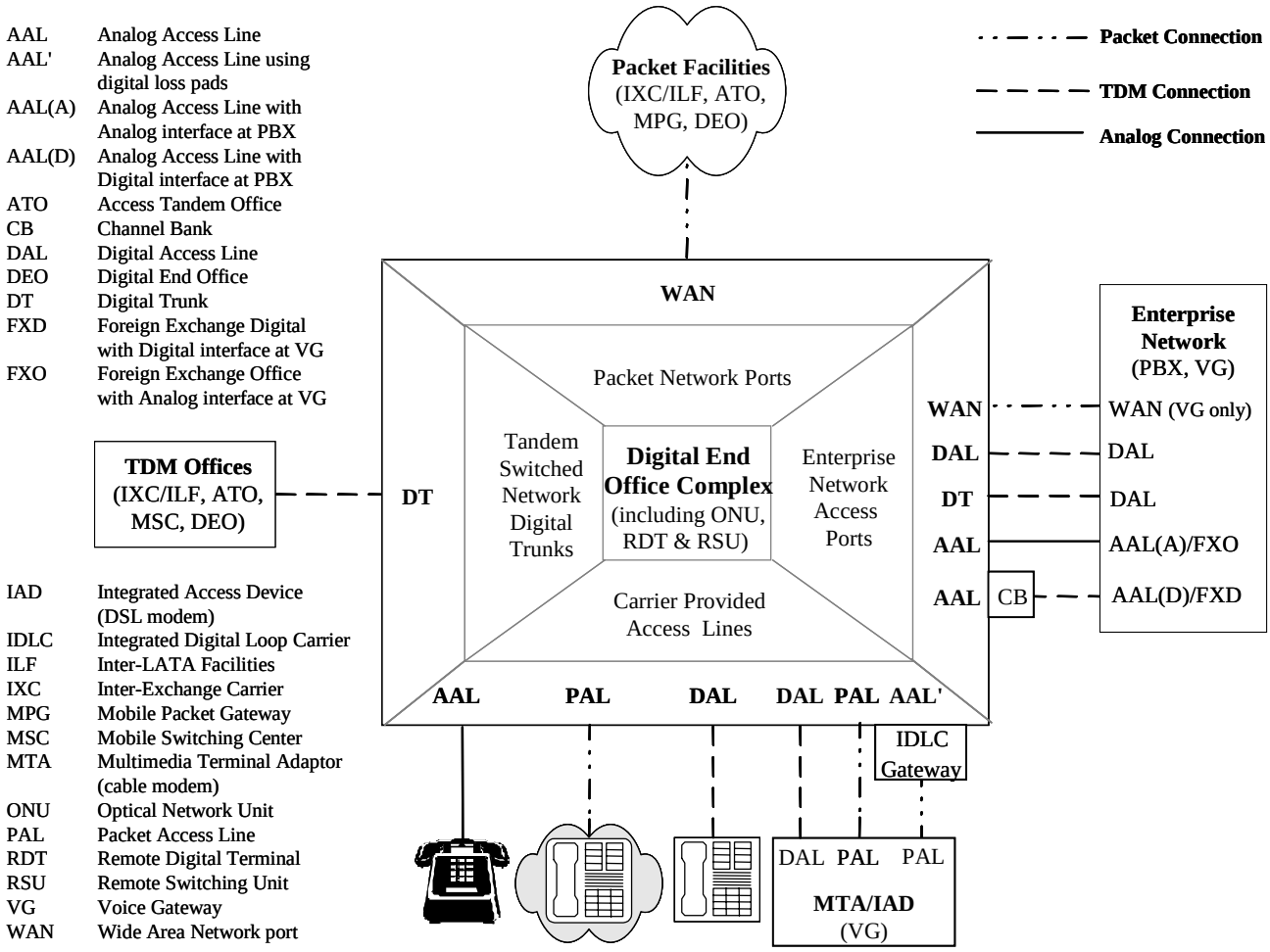

Figure 1 - DEO Loss Plan Ports and Interconnections

2 Normative References

The following standard contains provisions which, through reference in this text, constitute provisions of this American National Standard. At the time of publication, the edition indicated was valid. All standards are subject to revision, and the parties to agreements based on this American National Standard are encouraged to investigate the possibility of applying the most recent edition of the standard indicated below.

ITU-T Recommendation G.101 (11/03), *The transmission plan.*[1]

ITU-T Recommendation G.131 (11/03), *Control of talker echo.*[1]

ITU-T Recommendation G.711 (11/88), *Pulse code modulation (PCM) of voice frequencies.*[1]

[1] This document is available from the International Telecommunications Union. < http://www.itu.int/ITU-T/ >

3 Definitions

3.1 Access Line: A channel between an end-user's network interface and a local end office or remote switching unit.

3.2 Access Tandem Office (ATO): An exchange carrier switching system that provides a traffic concentration and distribution function for inter-LATA traffic originating/terminating within a LATA.

3.3 Analog Access Line (AAL and AAL'): An access line, with nominal bandwidth of 300 Hz to 3400 Hz, composed of analog facilities, or a combination of analog and digital facilities, having an analog interface at the local DEO complex and an analog or digital interface at the NI, end-user analog terminal, voice gateway, PBX, interconnecting network, etc. AAL ports have only analog loss pads and AAL' ports have only digital loss pads.

3.4 Analog Telephone: A wireline or cordless analog telephone that meets the requirements specified in ANSI/TIA/EIA-470.110-C.

3.5 Channel: A transmission path between two points (one or both points may be a POT or NI). The term *channel* may refer to a unidirectional path or a bidirectional path.

3.6 Connection: A temporary concatenation of transmission channels or telecommunication circuits, switching and other functional units set up to provide for a transfer of information between two or more points in a telecommunication network.

3.7 Digital Access Line (DAL): An access line using TDM technology, containing no analog sections, and having a digital interface at the local DEO and a digital interface at the end-user digital terminal (ISDN/TDM), voice gateway, PBX, or interconnecting network.

3.8 Digital Circuit Multiplication Equipment (DCME): A general class of equipment that permits concentration of a number of 64-kbps PCM speech circuits onto a reduced number of transmission channels.

3.9 Digital End Office (DEO): A complex using TDM or TDM and packet technology that includes the DEO and all connected Remote Switching Units (RSUs), Remote Digital Terminals (RDTs) and Optical Network Units (ONUs). A DEO is a switching system where access lines are terminated for the purposes of interconnection to each other and to trunks or packet network ports.

3.10 Digital Loop Carrier (DLC): A system that transmits information digitally in the local access line and multiplexes multiple channels into a composite signal.

3.11 Digital Telephone: An ISDN or IP digital telephone that meets the requirements specified in ANSI/TIA/EIA-810-A or TIA-920.

3.12 Digital Trunk (DT): A digital port using TDM technology, containing no analog sections, and having a digital interface at the local DEO and a digital interface at the adjacent DEO, ATO, or Inter-LATA facilities.

3.13 Echo Path Capacity: The maximum echo path delay for which an echo canceler is designed to operate.

3.14 Echo Path Delay: The delay from the R_{out} port to the S_{in} port of an echo canceler due to the delays inherent in the echo path transmission facilities, *including* dispersion time due to the network elements. In case of multiple echo paths, all delays and dispersions of any individual echo path are included. The dispersion time, which varies with different networks, is required to accommodate the band-limiting and hybrid transit effects.

3.15 Echo Return Loss (ERL): A frequency-weighted average, over the middle of the voiceband, of the return losses, RL (f), at any interface in a transmission path, with the transmission path terminated in a standard impedance at a specified point. The weighting is given in ANSI/IEEE 743. The 3 dB bandwidth of the weighting is 560 Hz to 1965 Hz.

3.16 Equivalent Send or Receive Loudness Rating (ESLR or ERLR): The equivalent loudness rating of a port is the SLR or RLR of the terminal plus any loss or gain in the circuit between the terminal and the port. For instance, an analog telephone with loop compensation and SLR = 6 dB and RLR = -5 dB on a 0 km loop would have an ESLR = 8 dB and ERLR = -3 dB at the DEO 0 dBr point, if the AAL port was connected via a two-wire loop of 2.7 km.

3.17 Exchange Access (EA): See *switched exchange access network*.

3.18 Exchange Carrier (EC): The telecommunications common carrier franchised to provide service in specific geographic areas.

3.19 Integrated Digital Loop Carrier (IDLC): A digital loop carrier system in which the digital switch and the carrier system interface using TDM technology.

3.20 Interexchange Carrier (IC): A carrier that provides telecommunication services between LATAs.

3.21 Local Access and Transport Area (LATA): A geographic area within each EC's franchised area where a LEC may offer local access and transport telecommunications services as well as local telecommunications services.

3.22 Local Number Portability: The term used to identify that the number portability capability is limited to the case where the directory number may only be ported within the same rate center – a geographic area used to distinguish rate boundaries.

3.23 Loop Compensation Loss: The loss that could be provided by a digital loop carrier system or a fiber-in-the-loop system when a short customer analog access cable results in below-average attenuation.

3.23a Loudness Rating (LR): A measure, expressed in decibels, for characterizing the *loudness* performance of *complete telephone connections* or of parts thereof such as *sending system*, *line*, *receiving system*.

3.24 Mobile Switching Center (MSC): A complex that provides switching between mobile telephone users and between mobile telephone users and the PSTN via an interface between the mobile network and the EC's network.

3.25 Network Interface (NI): The point of demarcation between the carrier's facilities and the end-user installation that establishes the technical interface and division of operational responsibility.

3.26 Network Loss: Loss added to the network to control echo.

3.27 Number Portability: A circuit switched network capability that allows a North American Numbering Plan number associated with an end-user to be moved from one serving switch in a network to another serving switch in the same or different network without changing the association between the end-user and the number.

3.27a Overall Loudness Loss: Loudness rating associated with the end-to-end loudness performance of a connection.

3.28 Optical Network Unit (ONU): A network element that is part of a fiber-in-the-loop system interfacing the customer analog access cables and the fiber facilities.

3.29 Packet Access Line (PAL): An access line using packet-based technology, containing no analog sections, and having a packet interface at the local DEO and a packet interface at the end-user digital telephone (Centrex IP) or voice gateway. Other end-user devices are supported by the WAN port.

3.30 Packet Network Port (PNP): A digital port using packet technology, connected to a voice gateway, an adjacent DEO, an ATO, or Inter-LATA facilities.

3.31 Point of Termination (POT): The point of demarcation between the facilities of two carriers that establishes the technical interface and division of operational responsibility.

3.32 Ported Call: A call that terminates on a ported number.

3.33 Ported Number: A directory number in a Portable NPA-NXX that resides on a switch other than the switch to which it is assigned in the Local Exchange Routing Guide.

3.34 Private Branch Exchange (PBX): End-user TDM switching equipment that meets the requirements specified in ANSI/TIA/EIA-464-C.

3.35 Public Switched Telecommunications Network (PSTN): A switched network accessible by the public for the purpose of originating and terminating telecommunications messages.

3.36 Receive Loudness Rating (RLR): Loudness Rating associated with the receiving system.

3.37 Remote Digital Terminal (RDT): An intelligent network element that interfaces between customer access lines and digital facilities.

3.38 Remote Switching Unit (RSU): The subtending remote switching device that depends in part on its host switch for call control but is capable of providing intra-unit switching.

3.39 Return Loss (RL): The ratio, expressed in dB, of the transmitted power to the reflected or returned power at a single frequency, at any interface in a transmission path with the transmission path terminated in a standard impedance at a specified point.

3.40 Send Loudness Rating (SLR): Loudness Rating associated with the sending system.

3.41 Switched Exchange Access Network: The network of switching systems, interconnecting facilities, and equipment provided by an EC to provide telecommunications services between the EO and IC-POT.

3.42 Terminal Equipment: Equipment at the end of a communication circuit, such as telephone sets, PBXs, voiceband data modems, and teletypewriters.

3.43 Transmission Reference Point: A hypothetical point at or near the sending end of each channel (preceding the virtual switching point), used as the "zero relative level point", in the computation of nominal relative levels. In the case of a digital end office, the main distribution frame is considered the 0 dBr point. (Source: ITU-T Recommendation G.100 - *1988 Blue Book*)

3.44 Voice Gateway (VG): End-user device that routes packetized voice from one end-point to another and meets the requirements specified in ANSI/TIA-912-A. The function of a voice gateway is analogous to a PBX and the loss plans are the same, but voice gateways route packetized voice, while PBXs switch TDM channels.

3.45 Weighted terminal coupling loss (TCL_W): A measure of echo performance, over the voice band under normal conversation, of a digital telephone set. Details of the measurement and calculation of TCL_W can be found in ANSI/EIA/TIA-810-A.

3.46 Wide Area Network (WAN): An access line or trunk using packet-based technology, containing no analog sections, and having a packet interface at the local DEO and a packet interface at the end-user voice gateway, or interconnecting network.

4 Abbreviations, Acronyms, and Symbols

0 dBr	Zero relative level
AAL *or* AAL'	Analog Access Line
AAL(A)	Analog Access Line with Analog interface at PBX
AAL(D)	Analog Access Line with Digital interface at PBX
ANSI	American National Standards Institute
ATIS	Alliance for Telecommunications Industry Solutions
ATO	Access Tandem Office
CB	Channel Bank
DAL	Digital Access Line
DAT	Digital Access Tandem
DCME	Digital Circuit Multiplication Equipment
DCS	Digital Crossconnect System
DEO	Digital End Office
DI	Digital Interface
DLC	Digital Loop Carrier
DSN	Digital Service Level N
DT	Digital Trunk
EA	Exchange Access
EC	Exchange Carrier
ECAN	Echo Canceler
EDSX (1/1)	A type of digital crossconnect system
EO	End Office

EPL	Echo Path Loss
ERL	Echo Return Loss
ERLR	Equivalent Receive Loudness Rating
ESLR	Equivalent Send Loudness Rating
FXD	Foreign Exchange Digital with digital interface at VG
FXO	Foreign Exchange Office with analog interface at VG
IAD	Integrated Access Device
IDLC	Integrated Digital Loop Carrier
ILEC	Incumbent Local Exchange Carrier
ILF	Inter-LATA Facilities
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunications Union – Telecommunications Standardization Sector
IXC	Inter-Exchange Carrier
LATA	Local Access and Transport Area
LI	Line Interface
LNP	Local Number Portability
M 13 MUX	DS1 to DS3 multiplexer
MPG	Mobile Packet Gateway
MSC	Mobile Switching Center
NI	Network Interface
OLR	Overall Loudness Rating
ONU	Optical Network Units
PAL	Packet Access Line
PBX	Private Branch Exchange
PCM	Pulse Code Modulation
PNP	Packet Network Port
POT	Point of Termination
PSTN	Public Switched Telephone Network
RDT	Remote Digital Terminal
RL	Return Loss
RLR	Receive Loudness Rating
RSU	Remote Switching Unit
SLR	Send Loudness Rating
TCL_W	Weighted Terminal Coupling Loss
TDM	Time Division Multiplex
TELR	Talker Echo Loudness Rating
VG	Voice Gateway
WAN	Wide Area Network port

5 General Criteria and Assumptions

5.1 General Criteria

Voice transmission quality depends primarily on received level, delay, echo, and distortion from digital signal processing. *Talker echo* is a particular source of voice transmission degradation as a result of increased delays in

digital networks. Talker echo is a delayed reflection of the talker's voice back to the talker's own ear. The perception of talker echo depends on the amplitude of the reflected signal and its delay with respect to the talker's voice. Control of talker echo is achieved for short delays by the insertion of loss to attenuate the talker's reflected signal. For longer delays, control of talker echo is achieved by the addition of echo cancelers.

ITU-T Recommendation G.107, *The E-model*, is a computational model that is useful for quantifying end-to-end transmission performance. Figure A.1 of Annex A, developed using the E-Model, shows the effect on voice quality of talker echo as a function of talker echo path loss for short delay situations.

In the case of a connection with four-wire digital access lines to both four-wire end-user terminals and a connecting digital network, echo can only occur as a result of acoustic coupling or crosstalk within the analog circuitry of the terminal. The digital terminal design requirements defined in ANSI/TIA/EIA-810-A were selected to provide sufficient talker echo control to preclude the need for network-based echo control devices, regardless of the delay across the connection. In the case of a connection with a digital access line and an analog access line, or two analog access lines, signal reflections can occur at four-wire to two-wire conversion equipment. In these cases, insertion of loss and/or the use of echo cancelers may be needed.

5.2 Fundamental Loss Planning Principles

The following fundamental principles were used in developing this standard:

1. The all-digital network, including both TDM and packet networks, is in principle bi-directional and lossless.
2. The ingress interface to the all-digital network is defined as a 0 dBr point. The digital egress interface is also a 0 dBr point, since the all-digital network is defined as lossless. In the PSTN, DEOs (i.e., Class 5 offices) use ITU-T Recommendation G.711 μ-law encoding.
3. Any loss inserted in the DEO Complex is inserted at the receive end of the connection, and is preferably inserted in the analog domain.
4. The terminal/gateway devices (which may be analog or digital), in conjunction with the DEO Complex loss, determine end-to-end acoustic levels. Distributed loss plans can potentially cause loudness, quantization distortion, reduced dynamic range, and echo issues.
    - ♦ The optimum loudness is OLR = 10 dB (ITU-T Recommendation P.79).
    - ♦ Digitally encoded speech levels on the all-digital network should be set to provide optimum dynamic range. This will be achieved if the Send and Receive Levels are partitioned as: SLR = 8 dB and RLR = 2 dB (ITU-T Recommendation P.310).
5. Echo sources should not exist within the all-digital network.
6. Echo paths, between corresponding egress and ingress interfaces to the all-digital network, must be controlled to eliminate degradation due to echo (ITU-T Recommendation G.131) and ensure stability (ITU-T Recommendation G.121).
7. The number of transcodings should be minimized.
8. Echo control devices should be located as close as possible to the source of the echo, to minimize echo path delay.
9. Transcoding devices should not be located in the echo path, due to a potential decrease in echo canceling capability.
10. Local Number Portability (LNP) may have been implemented as a network operating feature in the Incumbent Local Exchange Carrier (ILEC) network.

It is recognized that specific exceptions to these principles can occur when the benefits of such exceptions well outweigh the disadvantages. Care should be taken that any exceptions do not cause inter-operability problems with complying terminals or networks.

5.3 Assumptions

The loss requirements in this standard assume that the attached customer premises equipment complies with the TIA standards listed in Table 1, or later revisions.

Table 1 - Loss Plan Related Customer Premises Equipment Standards

Equipment Type	TIA Standard	Relevant Parameters
Analog Telephones	ANSI/TIA-470.110-C-2004	Nominal LRs: SLR = 8 dB, RLR = -3 dB
Narrowband Digital Telephones	ANSI/TIA/EIA-810-A-2000	Nominal LRs: SLR = 8 dB, RLR = 2 dB
Wideband Digital Telephones	TIA-920-2002	Nominal LRs: SLR = 8 dB, RLR = 2 dB
Wireless Digital Telephones	No Relevant Standard	Assumed LRs: SLR = 8 dB, RLR = 2 dB
PBXs	ANSI/TIA-464-C-2002 + ANSI/TIA-464-C-1-2004	Loss plan harmonized with T1.508
Voice Gateways	ANSI/TIA-912-A-2004	Loss plan harmonized with T1.508
V.90 and V.92 Modems	TIA-968-A-2002 TIA-968-A-1-2003	Encoded Analog Content < -6 dBm
Other Modems	TIA-968-A-2002 TIA-968-A-1-2003	Encoded Analog Content < -9 dBm

To be complete, a loss planning document must consider end-to-end loss and echo return loss. As echo return loss (ERL) is controlled by both LEC and customer actions, it is inappropriate to specify an ERL requirement in this standard. However, traditionally this standard has been used to document the industry agreed ERL assumption. Specifically, the assumed value for ERL during the development of this standard was: ERL = 14 dB with a standard deviation of 3 dB.

NOTE - ERL refers to the loss across the hybrid with the access line terminated at the NI with a standard impedance (typically 600 ohms in series with 2.16 μf), and it does not include any network losses.

For echo cancelers to operate satisfactorily, the combination of implemented DEO Complex loss and ERL actually encountered must provide sufficient level separation between the main talker and talker echo signals.

Automatic level control devices, if used, are ITU-T Recommendation G.136 compliant.

Loudness ratings are defined according to ITU-T Recommendation P.79.

6 DEO Complex Loss Plan Requirements

6.1 General

The purpose of a loss plan is to set the correct loudness level and to control echo. The digital end office complex has always played a major role in loudness level and echo control because the analog to digital and digital to analog conversion points for most voice traffic reside in the DEO Complex. This clause specifies the DEO Complex loss plan requirements.

6.2 Loss Plan Requirements

The DEO Complex loss plan requirements are specified in Table 2. The DEO Complex shall insert loss only in the receive path. The DAL/DT and PAL/WAN requirements are identical, but are presented separately to reflect the different transport technologies and the different termination technologies.

The objective of these requirements is to obtain an OLR of 10 dB for all digital connections and OLR between 10 and 14 dB for typical all-analog or mixed analog/digital connections, assuming terminals with nominal loudness ratings.

Table 2 - DEO Complex Loss Plan Requirements

			A	B	C
			AAL/AAL'	DAL/DT	PAL/WAN
	Loss (dB)		↑	↑	↑
1	AAL/AAL'	→	6[1]	0	0
2	DAL/DT	→	6	0	0
3	PAL/WAN	→	6	0	0

NOTE 1 - The TDM Intra-Switch value may be between 0 and 6 dB, depending on the delay and distance of the elements in the DEO (see Annex A).

The Packet Intra-Switch value may be between 0 and 6 dB if the DEO can instruct the AAL gateway to apply loss on a per-call basis to optimize OLR. Otherwise, the Packet Intra-Switch AAL-to-WAN and WAN-to-AAL values apply, rather than the AAL-to-AAL value.

When LNP has been implemented within the LATA and when ported calls may tandem at the donor switch, the loss must be set at 6 dB for all ported calls.

6 dB is required if either DEO is packet based and is preferred in all other cases. The 6 dB option will provide improved performance when IP-PBXs are terminated on either DEO and when interconnection is made to a DEO that uses packet technology.

For TDM, Intra-LATA connections, between different ECs where the connection between DEOs does not involve a tandem office; a 3 dB option may be applied. However, 6 dB loss is preferred.

See Clause 8 for private network and cellular network connections.

6.3 Recommended Loss Ranges

The losses specified in Table 2 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 2.

6.4 Implications of implementing the Loss Plan

Table 3 shows the expected OLR values that would result from implementing the loss plan in a DEO or DEO complex. The AAL ESLR (Equivalent Send Loudness Rating) and ERLR (Equivalent Receive Loudness Rating) values assume nominal loudness ratings for analog terminals plus the loop loss for a typical 2.7 km connection. The DAL/DT and PAL/WAN ESLR and ERLR values assume nominal loudness ratings for digital terminals.

The values in Table 3 are consistent with ITU-T recommended values (G.121).

Table 3 - DEO Complex expected OLRs

			AAL/AAL'	DAL/DT	PAL/WAN
	(dB)	ERLR	-3[1]	2	2
	ESLR	OLR	↑	↑	↑
AAL/AAL'	8	→	11[2]	10	10
DAL/DT	8	→	11	10	10
PAL/WAN	8	→	11	10	10

NOTE 1 - Based on 2 dB loop loss at 2.7 km (A nominal combination of 2 dB loop compensation in the telephone at 0 km and 4 dB loop loss at 2.7 km.)

NOTE 2 - OLR = ESLR +ERLR + DEO Loss from Table 2

7 Delay and echo considerations

7.1 General

Delay can have two effects on connection performance. In the absence of noticeable echo, delay can interfere with dynamics of voice communication and also affect voiceband data throughput and response time. In the presence of noticeable echo, increasing connection delay makes echo effects worse.

End-user perception of talker echo depends on the amplitude and round-trip delay of the returned echo. Increase in round-trip delay necessitates more attenuation of the returned echo. However, insertion of too much network loss causes the primary voice signal to be excessively attenuated. At a certain amount of delay, echo cancelers are required as a means to control the echo without inserting excessive loss.

Connections requiring echo cancelers should use devices that at least meet the requirements of either ITU-T Recommendation G.165 or ITU-T Recommendation G.168. Most modern canceler designs satisfy or exceed the performance specified in these recommendations.

7.2 Delay limits

For voice applications on connections using all-terrestrial facilities, round-trip delays between two end-users should be kept below 300 ms. This limit is believed to be stringent enough to preclude essentially all difficulties potentially encountered in conversational interactions, and applies to international connections.

The effect of delay on the voiceband data user is highly application dependent. For example, round-trip delays on the order of 100 ms can degrade the throughput of certain half-duplex applications, and noticeably increase the response time of many multipoint applications. Since such absolute delay is well below the point at which voice applications are affected, the sensitivity to delay of many voiceband data applications should be reflected in digital network planning and design. It is recommended that round-trip delays for voiceband data applications be kept below 100 ms whenever possible.

For connections using a satellite link (full-hop or half-hop), general guidance is available in ITU-T Recommendation G.114.

7.3 Echo control guidelines

While detailed echo control guidelines for specific services cannot be meaningfully presented here, there are excellent general guidelines for dealing with echo on voice connections. For example, ITU-T Recommendation G.131 provides a thorough set of practical rules for connections with and without echo control devices. The information provided in ITU-T Recommendation G.131 is generally consistent with traditional planning rules used in North America. Such rules, however, were based on specific assumptions about routing schemes used and echo path losses encountered, and these same assumptions may not apply to connections realized by the wide variety of current and future network configurations possible in a multivendor environment.

It is not required that echo cancelers be deployed in fully digital connections that terminate in digital voice terminals. Echo control in such connections will be achieved by digital terminals meeting the requirements of ANSI/TIA/EIA-810-A. It is assumed, however, that echo cancelers will remain in the network as necessary to terminate connections in an analog terminal.

8 Interconnection guidelines

This clause contains guidelines for the interconnection of the PSTN and any other networks or services. There may be many different types of networks and services that can be interconnected with the PSTN. Subclause 8.1 provides general interconnection guidelines, applicable for any interconnection type, and 8.3 provides additional specific guidelines for mobile services. The guidelines cover speech signal levels, echo control, and delay requirements that are necessary to maintain end-user satisfaction over connections traversing the PSTN and interconnected networks.

The following guidelines are established with the assumption that the interconnected networks or services are all-digital with digital or analog access lines from appropriate end-user terminals.

8.1 General guidelines

Networks and services interconnected to the PSTN should provide and expect voice signal levels at the NI or POT similar to those at an equivalent point in the normal PSTN connection. The send level is the level in the direction toward the PSTN and the receive level is the level in the direction toward the attached network.

8.1.1 Interconnection from a digital access line

If the portion of a network interconnected to the PSTN is fully digital, including the terminal, the equivalent send and receive levels should be the same as when the call is terminated at the NI by a digital terminal. In addition, the loudness ratings, the transmit and receive loudness rating characteristics of the interconnected network or service at the NI (when using a handset terminal in normal talking conditions) should be the same as that specified for an ANSI/TIA/EIA-810-A terminal. (R2013)

The loss between a digital access line in the interconnected network and the PSTN connecting point (NI or POT) should be 0 dB in each direction.

The weighted terminal coupling loss (TCL_W) in the digital access portion of the connection should be equal to or higher than the TCL_W specified in ANSI/TIA/EIA-810-A.

8.1.2 Interconnection from an analog access line

For those situations in which the network interconnected to the PSTN is terminated by an analog terminal, the equivalent send and receive levels, and loudness rating characteristics at the network connecting point (NI or POT), should be within the ranges of expected equivalent levels and loudness ratings on the trunk side of the DEO in the normal PSTN connection. This is because the PSTN will not provide the required loss that is usually introduced in a PSTN-only connection -- e.g., the receive loss of a DEO/RSU/RDT.

The echo path loss (EPL) of the portion attached to the PSTN connection should be within, or higher than, the EPL range normally expected for the analog access lines in the PSTN (see 5.3 for ERL assumption), including normally applied PSTN network losses. The return loss value at the network interface looking toward the interconnecting network should be higher, so that the above value is achieved at the DEO when combined with the return loss effect of the interconnecting trunk. A value of 18 to 20 dB is currently suggested.

8.1.3 Echo control guidelines

Delay introduced by the interconnecting network can cause two types of echo problems. When no echo cancelers are present in the PSTN part of the connection, echo may become objectionable due to the added delay of the interconnecting network. The other problem occurs when cancelers are present in the PSTN part of the connection, and the added echo path delay may preclude the PSTN cancelers from operating properly. The echo control guidelines and delay considerations provided in clause 7.2 of ITU-T Recommendation G.131 may also be useful for interconnecting networks. However, it is recognized that the stringent criteria used for PSTN applications may be too restrictive for private networks. If such PSTN guidelines are relaxed, however, caution must be exercised to ensure that interconnecting network delay does not increase to a value that precludes proper operation of cancelers in the PSTN.

If the whole connection is digital, and echo cancelers are not provided, the terminal in the interconnected network should comply with ANSI/TIA/EIA 810-A (see Table 1).

8.1.4 Delay limits

Networks and systems interconnected to the PSTN should limit their added delay as much as possible in order to minimize performance degradation caused by effects of pure delay (i.e., in the absence of echo). In general, any such end-to-end interconnections should not exceed limits for delay specified for the PSTN voice and voiceband data services in 7.2.

8.1.5 Distortion due to digital signal processing

Digital signal processing devices cause distortion. Examples are quantizing distortion (typical of waveform codecs), and the non-linearities due to non-waveform codecs. ITU-T Recommendation G.113 provides guidance on the effects of both types of distortion.

The performance impact of quantizing distortion depends upon the transmitted signal level and spectrum, the specific digital processes encountered, and network losses. An all-digital DEO-to-DEO connection over the PSTN should, in general, have few sources of significant quantizing distortion. Inter-connections to the PSTN, however, may introduce significant quantizing distortion due to, for example, the many different types of low bit-rate voice codecs used in voice messaging systems, mobile services, and digital circuit multiplication equipment. The amount of quantizing distortion caused by such interconnections must be limited to ensure acceptable speech quality. Although recommended limits cannot be provided at this time, it is desirable that interconnections to the PSTN should limit quantizing distortion so that, as a worst case, the limit specified by ITU-T Recommendation G.113 for international connections is satisfied -- see clause 5.6 of ITU-T Recommendation G.113 (02/96).

For non-waveform codecs, ITU-T Recommendation G.113 (02/01) provides E-model (see ITU-T Recommendation G.107) impairment factors that characterize coder performance.

8.2 PBX/Private network interconnection

This subclause addresses the interconnection with the PSTN of PBX/private networks that are all digital and have a digital network interface with either digital or analog access lines from end-user terminals. Interconnection of such PBX/private networks should follow the guidelines of clause 7 depending upon the classification of the access lines (i.e., analog or digital) from the end-user terminals. The end-to-end performance of PBX-PSTN connections will depend heavily on appropriate loss treatment applied by the PBX. Guidelines regarding the loss treatment options for the digital PBX can be found in ANSI/TIA-464-C + ANSI/TIA-464-C-1.

8.3 Mobile network interconnection

This subclause provides transmission loss, levels, and echo control planning rules to ensure optimum transmission performance for digital (dual mode) and analog mobile units when interconnecting digitally with the PSTN.

8.3.1 Level considerations

In order to ensure optimum performance when interconnecting with the PSTN, the nominal transmitted speech level at the point of termination (POT) should be equal to that which would be generated in the public network by a digital set[2] that conforms to the transmission characteristics specified in ANSI/TIA/EIA-810-A. The received level at the POT is dictated by the transmission characteristics of the public networks and in general will be the same as that of a PSTN end office.

8.3.2 Digital mobile terminal characteristics[3]

In order to be compatible with the rest of the network when digitally interconnected to the PSTN, the SLR and RLR of the mobile unit and the base station measured at the POT should be derived from the specifications for digital sets (ANSI/TIA/EIA-810-A). That is, the loudness rating values when operating in a conventional non-mobile noise environment should be as follows:

- SLR = 8 dB
- RLR = 2 dB

Given that the mobile subscriber will often be operating in inherently noisy conditions, one can assume that the average talking and the preferred listening levels in the mobile setting are likely to be higher than those in the home or office environments. Therefore, in order to generate the appropriate signal levels at the POT and at the mobile set receiver, the SLR and RLR of the digital mobile set may be adjusted.

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[2] Equivalently, the transmit level at the POT should be equal to that which would be generated in the PSTN by an average analog set on an average loop.

[3] Specifications for hand-held terminals are covered in this clause. Hands-free terminal specifications are not included in this standard.

8.3.3 Guidelines for echo control

The mobile set is assumed to have a TCL_W of 45 dB, which is intended to safeguard the public network subscriber from the effects of talker echo. Consequently, no additional echo control device is required.

However, echo cancelers *are* required in order to eliminate talker echo effects at the mobile end of the connection (i.e., that experienced by the mobile user) because of added delay in the mobile radio system. Given the evolving public network architecture (e.g., the deployment of SONET self-healing rings) and echo control rules, echo cancelers with an echo path capacity of up to 48 ms of round-trip delay should be used. However, there may be rare cases in which 48 ms echo cancelers are not sufficient to cancel echoes generated in the PSTN. Hence, mobile service providers must evaluate these particular situations and use echo cancelers of greater capacity if warranted to minimize potential impairments for their customers. Figure 2 depicts the architecture used to arrive at this conclusion.

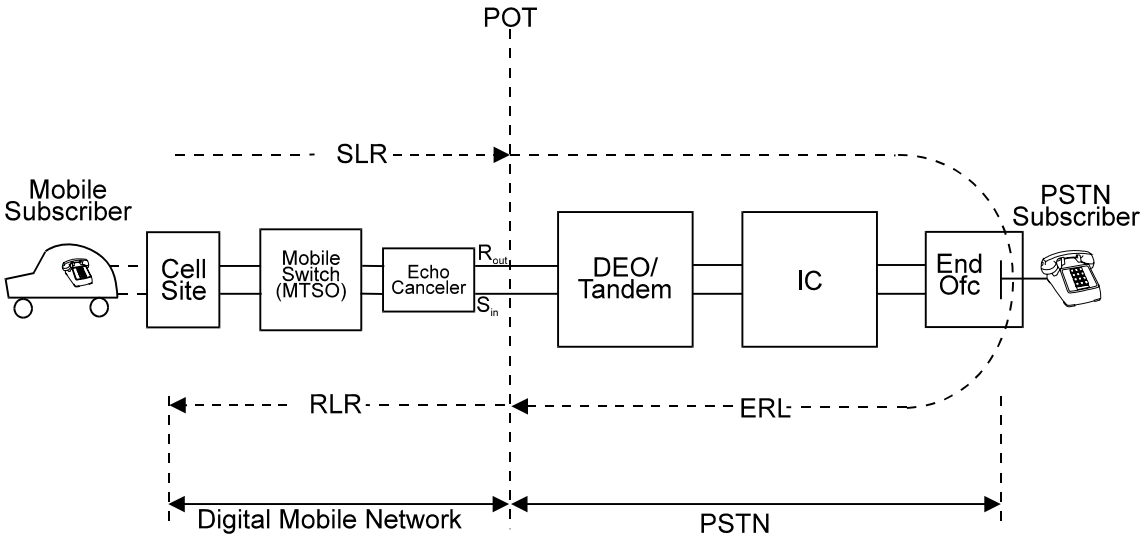

Figure 2 - Echo control in digital mobile – PSTN connections

Annex A

(informative)

A Connection Performance as a Function of Talker Echo Path Loss

A.1 The Relationship between Talker Echo and Loss Pads in Short Delay Situations

Figure A.1 is an E-model plot showing how Transmission Rating (R) degrades as one-way delay increases without active echo control. Notice that the Rating is high near 0 ms for the 0, 3 and 6 dB pads, the TELR = 19, TELR = 25 and TELR = 31 curves respectively, but as one-way delay increases beyond about 3 ms, the performance of the 0 and 3 dB pads degrades dramatically favoring the 6 dB pads. By 25 ms, the performance curve relating to the 6 dB pads degrades to about R = 70; at this point, active echo control must be enabled per ITU-T Recommendation G.131. The TELR = 66 curve illustrates the nominal performance that can be expected with an echo canceler. The ability of an echo canceler to achieve this performance level may depend on the insertion of a 6 dB loss pad at the DEO Complex.

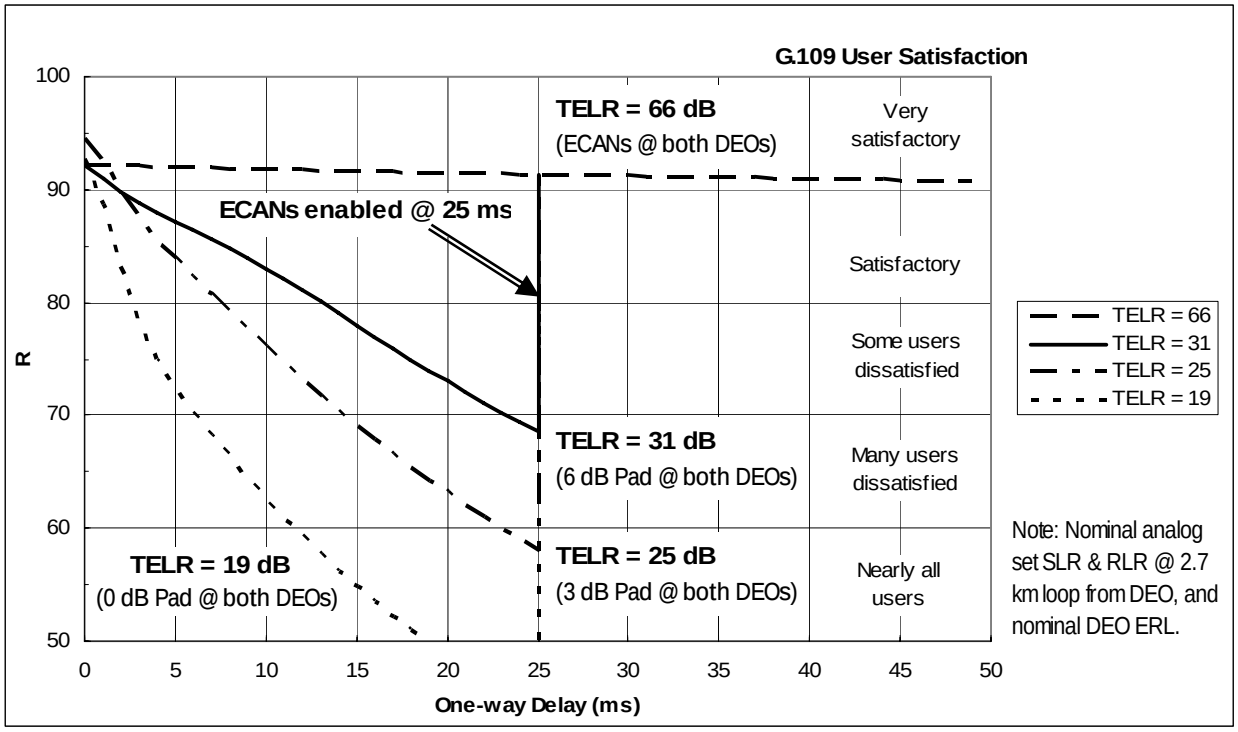

Figure A.1 - Connection Quality as a function of Talker Echo Path Loss

NOTES:

1. This diagram assumes than an echo canceler is provisioned at 25 ms of one way delay, per ITU-T Recommendation G.131.
2. For the proper functioning of echo cancelers, the echo path provided by the network and the access must be linear and non-variant.

A.2 Intra-LATA (same EC) – TDM Guidelines

Flexibility is needed in the choice of the exact value of network loss to be used for intra-LATA connections involving a single exchange carrier to avoid the use of an echo canceler. The network loss should be between 0 and 6 dB, typically 0 dB, 3 dB, or 6 dB.

The following general guidelines should be applied when choosing the network loss for intra-LATA connections:

1. For intra-DEO/RSU connections between metallic access lines, 0 to 6 dB is recommended for DEOs or RSUs with line lengths up to 12 kft, and 0 dB for DEOs or RSUs with line lengths above 12 kft. The selection of appropriate loss values will be administered by the EC and is not required to be an automatic function of the switch.

2. For connections involving an RSU/RDT or multiple RSUs/RDTs, two considerations apply: (1) the length of the connected metallic loops; and (2) the value of round trip delay between the host and the remote. When considering the length of the connected metallic loops, the desired value of network loss is the same as that for intra-DEO/RSU connections between metallic access lines. The following guidelines can be applied for the selection of the network loss based on the round-trip delay between the host and the remote:

    a. 0 dB can be used up to 3.2 ms round-trip delay (approximately 80 mi, or 130 km).

    b. 3 dB can be used up to 8 ms round-trip delay (approximately 360 mi, or 580 km).

    c. 6 dB can be used up to 12 ms round-trip delay (approximately 600 mi, or 945 km).

The network loss value is administered at the DEO/RSU by the EC and is applied to a portion of, or the total, DEO/RSU.

Annex B

(informative)

B Delay Guideline Methodologies

B.1 Methodology used in developing delay guidelines

This annex describes the approach used in developing round-trip delay calculations. Round trip delay consists of two components: (1) the *propagation delay*; and (2) the *processing delay*. There are many possible models that could illustrate the need to account for every instance of propagation and processing delay. The example below intends to be neither overly simplistic nor overly complex.

The *propagation delay* is a function of the route distance. It is assumed that all of the facilities in Figure B.1 have the delay of fiber, such that the round-trip propagation delay (in ms) is equal to 0.0168 X one-way distance (in miles) between the NI and POT.

The *processing delay* depends on the number and types of network elements included in the NI-to-POT path. The characteristics of processing delay of select network elements are summarized by a typical value and a range as displayed in Table B.1.

There are four representative NI-to-POT paths depicted in the reference architecture (see Figure B.1). Three processing delay values can be computed for each path using the low, the typical and the high network element values of Table B.1, for a total of 12 values. The value used as the typical processing delay is taken to be the value of the path using a direct line to an RSU evaluated with typical network element values.

Table B.1 - Assumed range and typical round-trip delay values of network elements (ms)

Network element	Typical value	Low value	High value
DEO/HOST DI to LI	1.3	0.75	1.9
DEO/HOST DI to DI	1.0	0.9	1.4
RSU DI to LI	1.2	1.0	1.7
RSU DI to DI	0.8	0.7	1.0
RDT	0.6	0.5	1.3
DCS	0.8	0.6	1.5
DAT	1.0	0.9	1.3
M13 MUX	0.04	0.04	0.04
EDSX (1/1)	0.2	0.2	0.2
NOTE – The choice of particular access arrangement is not meant to constrain delay values to those generated as estimates from the reference model. Deployment of systems with new capabilities and features in the access network is bound to increase delay in that network, since enhanced capabilities and expanded feature sets are associated, by necessity, with additional signal processing. For example, the typical delay given for the RDT applies only for basic service. Additional features such as Time Slot Interchange, Concentration of IDLC Channels, etc., would increase the typical round-trip delay.			

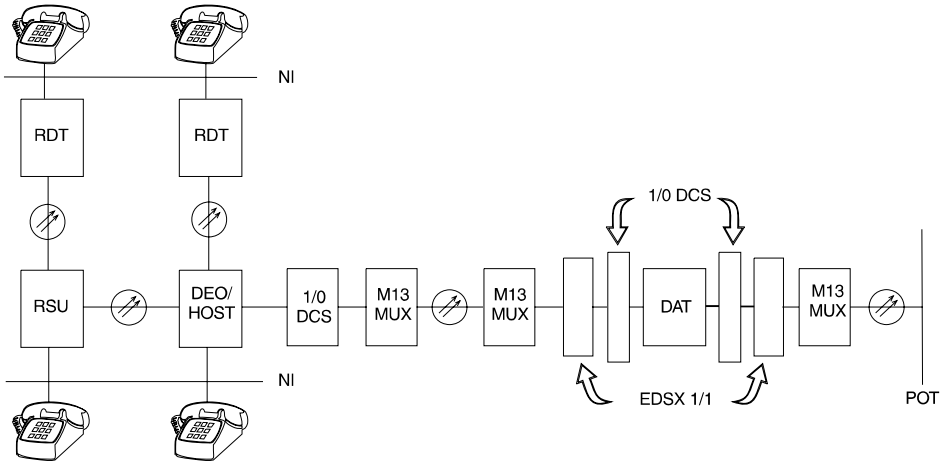

Figure B.1 - NI-to-POT digital reference model architecture

Annex C

(informative)

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