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Speech processing, Transmission and Quality aspects (STQ); Harmonized Pan-European/North-American loss and level plan for voice gateways to IP based networks

Reference

DES/STQ-00021

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Foreword

This ETSI Standard (ES) has been produced by ETSI Technical Committee Speech processing, Transmission and Quality aspects (STQ) in collaboration with the Telecommunications Industry Association, TIA, and is now submitted for the ETSI standards Membership Approval Procedure.

The North American version of this joint work is the "Voice Gateway Transmission Requirements" standard. Within TIA it has the number TIA-912 [1].

1 Scope

The purpose of the present document is the harmonization of loss and level planning between the Pan-European and the North American regions, with a global view and the intent that the present document may be subject to adoption by other regions, in future. An equivalent standard has been published by the North American Telecommunications Industries Association (TIA 912 [1]).

The objective of the present document is to specify a loss plan for IP based networks particularly suitable for use by voice gateway manufacturers operating in several European countries. The recommended loss values are based on a Pan-European analogue telephone set as specified in TBR 38 [4] and typical European public network losses.

These loss values may not provide optimum performance in countries where the telephone set loudness ratings or network losses differ significantly from the values used in the present document. Manufacturers may need to modify their loss plans accordingly, but the principles described is the present document provide guidance on how to derive suitable loss values. A spreadsheet is also provided as an annex to assist manufacturers in determining the correct loss values.

The PBX ports referred to in the present document are as defined in ES 201 168 [2] or TIA 912 [1].

The present document introduces two general principles to loss and level planning to be applied to voice gateways to IP based networks, namely:

- The gateway handling the originating entity will set the ESLR of the sending end point to 8 dB at the ingress to the IP network thereby ensuring that there is a harmonized sending level for voice calls in IP based networks.

- The gateway handling the terminating entity will adjust the loss at the egress from the IP network to achieve the desired OLR at the receiving end point (this adjustment is necessary because of historical differences in loudness levels in analogue networks and terminals in different countries).

This will enable network planners to achieve the internationally recognized optimum overall loudness rating of 10 dB

The advantage of this approach is that:

- clear and separate responsibility is established for loss adjustments for the sending and receiving entities for each mouth to ear path,

- the party responsible for each voice gateway is able to make the adjustments necessary without requiring information or co-operation from any other party, and so loss planning and adjustment becomes a local/national issue.

NOTE: The approach taken in the present document is the same as is the half-channel approach that is now well established for PBXs in Europe.

The present document applies to all kind of 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 present document is initially limited to voice gateways to IP telephony networks. For the purposes of the present document, a voice gateway is considered to be a device that performs routing functions between:

- telephones (analogue, digital, IP);

- public and private network trunks;

- IP based networks.

The present document may apply also to loss planning in any purely digital transmission network which may include:

- circuit switched networks;
- frame relay or ATM networks;
- mobile networks.

While the present document does not apply to other services, which may be carried over the same infrastructure, e.g. Voiceband Data or Fax; nevertheless, complying with the present document will, in general, be advantageous for such other services.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

[1] TIA/EIA 912: "Voice Gateway Transmission Requirements".

[2] ETSI ES 201 168: "Speech processing, Transmission and Quality aspects (STQ); Transmission characteristics of digital Private Branch eXchanges (PBXs) for interconnection to private networks, to the public switched network or to IP gateways".

[3] ITU-T Recommendation Q.551: "Transmission characteristics of digital exchanges".

[4] ETSI TBR 38: "Public Switched Telephone Network (PSTN); Attachment requirements for a terminal equipment incorporating an analogue handset function capable of supporting the justified case service when connected to the analogue interface of the PSTN in Europe".

[5] ITU-T Recommendation G.107: "The E-Model, a computational model for use in transmission planning".

[6] ITU-T Recommendation G.101: "The transmission plan".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

send loudness rating: acoustic/electrical conversion characteristic of the originating equipment (typically a telephone), determined by measuring the conversion characteristics over the telephony frequency band and by applying a weighting factor for each third octave band

receive loudness ratings: electrical/acoustic conversion characteristic of the terminating equipment, determined by measuring the conversion characteristics over the telephony frequency band and by applying a weighting factor for each third octave band

loudness rating: Send Loudness Rating (SLR) and Receive Loudness Rating (RLR) are collectively referred to as the loudness rating (LR) of a telephone

NOTE: The loudness ratings are given in the order SLR and RLR, i.e. a digital telephone set with an SLR of 8 dB and RLR of 2 dB would be designated as having an LR of 8 dB and 2 dB.

equivalent loudness rating: Equivalent Loudness Rating (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

reference point: voice gateway-to-IP network connection point (in the send direction)

IP Send Loudness Rating (iSLR)**:** specific designation for the equivalent send loudness rating at the reference point

reference level: an IP Send Loudness Rating (iSLR) of 8 dB

encoding: analogue to digital conversion

decoding: digital to analogue conversion

transcoding: transcoding refers to the conversion from one voice coding algorithm to another

NOTE: E.g. ITU-T Recommendations G.711 to G.729.

telephony terminal: The terms telephone, terminal equipment, and end point are equivalent, and refer to the device that performs the acoustic/electrical conversion.

full-channel loss plan: loss plan where the losses are defined on a port-to-port basis

half-channel loss plan: loss plan where the losses are defined on a port-to-standard reference point basis

input connection: unidirectional path from an interface of a digital exchange to an exchange test point

output connection: unidirectional path from an exchange test point to an interface of a digital exchange

half-connection: bi-directional path comprised of an input connection and an output connection, both having the same exchange interface

full-connection: bi-directional path comprised of two half-connection, providing a complete connection though the digital exchange

Notes on connections:

NOTE 1: These terms may be qualified by the words analogue or digital, the qualification signifying the property of the exchange interface.

NOTE 2: An analogue input (output) (half) connection may be further qualified by the words 2-wire or 4-wire.

NOTE 3: Figure 1 shows the relationship between the terms full and half connection, as defined in ITU-T Recommendation Q.551 [3], and full and half channel as defined in the present document.

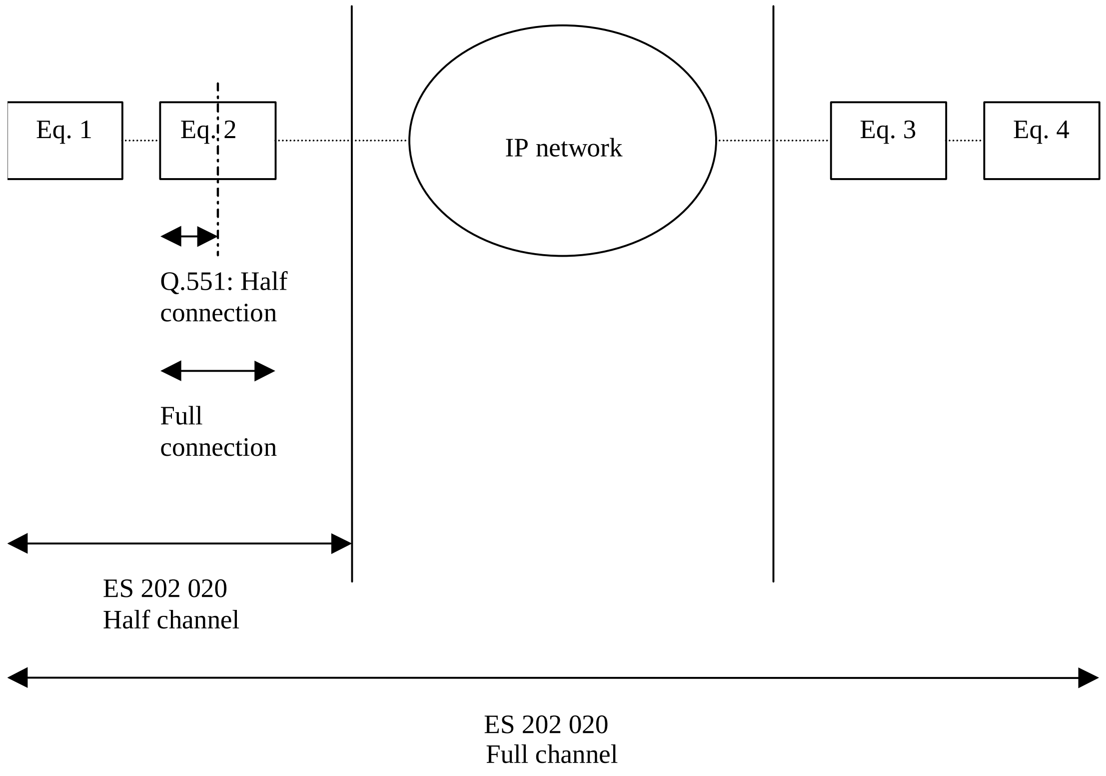

Figure 1: Connection versus channel

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ELR	Equivalent Loadness Rating
ESLR	Equivalent Send Loadness Rating
OLL	Open Loop Loss
OLR	Overall Loudness Rating
RLR	Receive Loudness Rating
SLR	Send Loudness Rating

4 Loss and level planning

4.1 Overview

Developing a loss plan can be 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, for each connection type must be known.

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

The complexity is introduced when the IP telephony network connects to analogue 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.

Voice gateways can also connect to existing analogue and TDM based digital networks. These connections require losses to be defined on a port-to-port basis for technical and sometimes regulatory reasons, and full-connection loss plans are required for voice gateways.

4.2 Half-channel loss and level plans

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 telephone sets 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 recent move to standardize the North American digital telephone set LRs to the ITU-T recommended levels now 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 regulatory requirements may make it difficult to implement.

4.3 Loudness rating definitions

4.3.1 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 the present document 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 dB and 2 dB.

4.3.2 Equivalent loudness rating

For the purpose of loss planning it is necessary to know the equivalent loudness rating (ELR) of an analogue voice gateway port (it can be assumed that a digital port would have a LR of 8 dB and 2 dB). 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.

EXAMPLE: An L2 analogue telephone with an LR of 3 dB and –8 dB, connected to a voice gateway via a 2-wire loop with 4 dB loss in each direction, would have an ELR of 7dB and –4 dB as shown in figure 2.

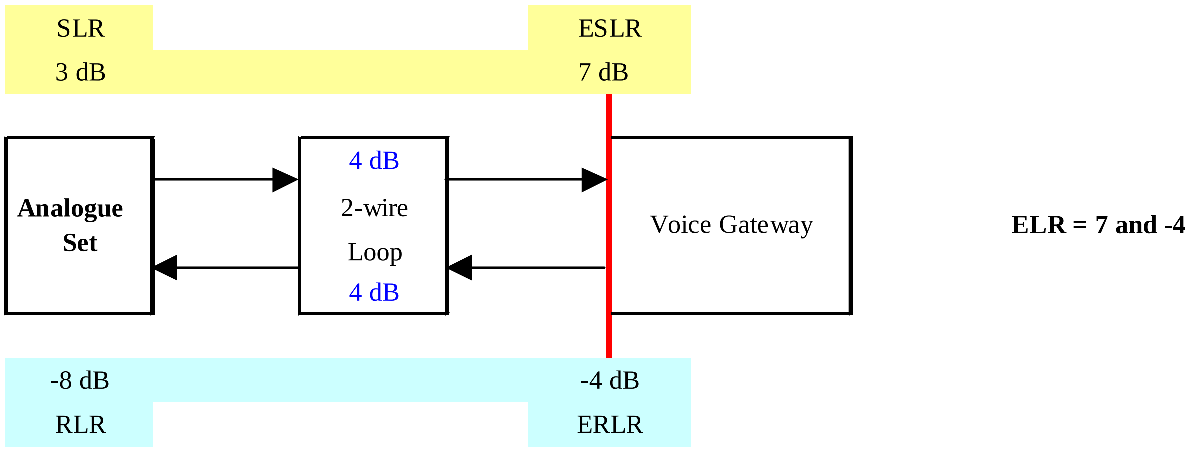

Figure 2: Equivalent loudness rating example

4.3.3 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 IP SLR (iSLR).

4.4 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 a 0 dBr point in standard TDM circuit switched telephony. This is illustrated in the "ELR Derivations" spreadsheet in annex B.

The reference level is defined as a equivalent send loudness rating of 8 dB, not a power level at 1 020 Hz.

4.4.1 Network level changes

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.

4.5 Harmonized half-channel loss and level plan

The present document gives the requirements for the harmonized half-channel loss and level plan for two regions, the Pan-European region and the North American region.

These harmonized half-channel plans define the following (for IP network connections):

- the LR or ELR of each port;
- the transmit losses required to achieve an ESLR of 8 dB for each port at the IP zero-level point;
- the receive losses required to achieve a satisfactory OLR for each receiving end point;
- the stability loss for each port.

4.6 Regional full-connection loss and level plans

Annexes C and D of the present document provide the full-connection loss and level plans for two regions, the Pan-European region and the North American region. Each loss plan consists of two sections:

1) An overview defining the loss and level plan ports

2) A full-connection plan defining:

   - the LR or ELR of each port;

   - the port-to-port loss for each port to every other port;

   - The OLR from each port to every other port based on the LRs or ELRs, and the corresponding port-to-port loss.

NOTE: For the reasons stated in annex D, there is no Pan-European full-connection loss plan available at this time.

5 Requirements for the harmonized half-channel loss and level plan

- The originating entity will set the ESLR of the sending end point to 8 dB at the ingress to the IP network.

- The terminating entity will adjust the loss at the egress from the IP network to achieve the desired OLR at the receiving end point.

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

The tables provided in the following clauses for either region are structured according to the same principles. A reference configuration is illustrated in the "ELR Derivations" spreadsheet in annex B.

5.1 The Pan-European half-channel loss plan

Table 1 shows the voice gateway half-channel loss plan for the Pan-European region, given in equivalent loudness ratings and respective loss.

Table 1: 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
	ESLR	Tx Loss	iSLR	ERLR	Rx Loss	OLR	Asmd. OLR
L2	3	5	8	-8	10	10	10
LD	8	0	8	2	0	10	10
KD	8	0	8	2	0	10	10
WAN	8	0	8	2	0	10	10
K2	16	-6	10	5	-3	12	10
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.

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

5.2 The North American half-channel loss plan

Table 2 shows the voice gateway half-channel loss plan for North America, given in loudness ratings and respective loss.

Table 2: Voice gateway half-channel loss plan

	a	b	c (WAN Zero-Level Point ↓)	d	e	f	g	h
			a + b			c + d + e	8 + d + e	b + e
	ESLR	Tx Loss	iSLR	ERLR	Rx Loss	OLR	Asmd. OLR	Stablty. Loss
ONS	8	0	8	-6	9	11	11	9
OPS	11	-3	8	-3	6	11	11	3
DGS	8	0	8	2	0	10	10	0
DAL	8	0	8	2	0	10	10	0
WAN	8	0	8	2	0	10	10	0
FXO	17	-6	11	3	0	14	11	-6
FXD	14	-3	11	0	3	14	11	0
ATT	13	-3	10	-1	3	12	10	0

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 RLR of the receive side.

Column **e** shows the receive loss required to achieve the desirable OLR, based on the RLR 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

Column **h** shows the stability loss, which is the sum of the 4-wire transmit and receive losses.

NOTE 1: Losses have been selected as multiples of 3 dB, in the assumption that this may make implementation easier.

NOTE 2: It is not possible to achieve the optimum iSLR of 8 dB for connections from analogue networks due to the potential for DTMF signalling overload.

NOTE 3: The OLR values shown in the table are as perceived by the listener, i.e. this is shown as a one-way connection.

Annex A (informative): Loss and level planning and loudness ratings

This tutorial has been taken from the North American Voice Gateway Transmission Requirements standard, but it is believed that this information will be useful in the Pan-European context.

A.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 FCC Rules Part 68 regarding the prevention of network harm.

A.2 Send and receive levels

The objective of a telephone connection is to simulate a 1 metre free air path between two talkers. This simulation involves several objective and subjective factors that are not present in the 1 metre 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

A.3 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 A.1. As the acoustic and electrical units are both relative levels in dBs, the conversion factors are also in dBs.

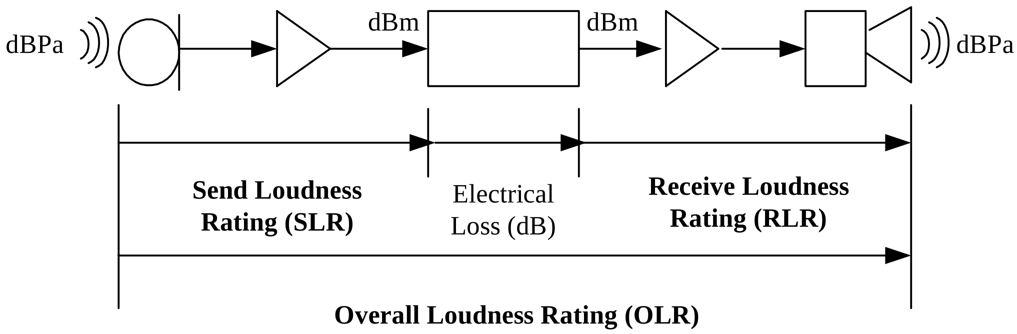

Figure A.1: Terminal loudness ratings

The loudness ratings of the three telephone ports defined in the present document are shown in table A.1.

Table A.1: Telephone Loudness Ratings

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

NOTE 1: The OPS loudness ratings are representative of 2500-type analogue 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 ANSI/TIA/EIA-470-B for further details.

NOTE 2: The ONS loudness ratings are representative of 2500-type analogue telephones operating on short loops with the typical current-limited battery feed and 600 ohm impedance characteristics of voice gateway ONS ports. See ANSI/TIA/EIA-470-B for further details.

NOTE 3: The DGS loudness ratings of SLR = 8 dB and RLR = 2 dB conform to the requirements specified in ANSI/TIA/EIA-810-A.

A.3.1 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 A.2 for set-to-set calls within a voice gateway.

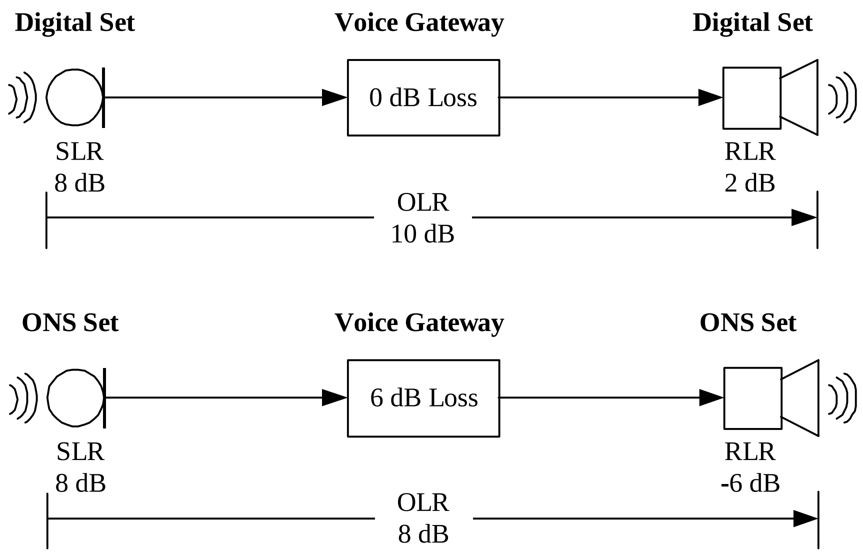

Figure A.2: Telephone-to-telephone overall loudness ratings

A.3.2 Optimum overall loudness ratings

Figure A.3 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.

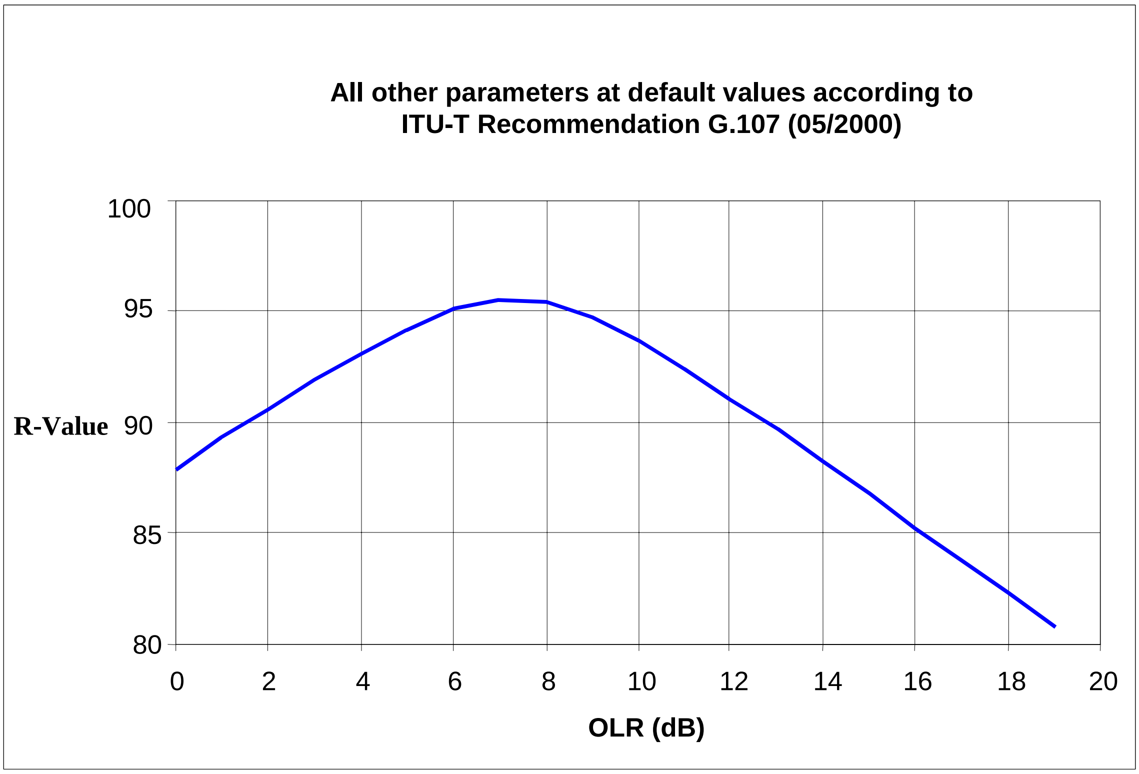

Figure A.3: E-Model Optimum Overall Loudness Rating

The recommended value for the Overall Loudness Rating (OLR) for standard applications of 3,1 kHz handset telephony is 10 dB (ITU-T Recommendation G.111).

Investigations have shown the optimum OLR to be somewhat lower than 10 dB. for connections free from echo and sidetone problems. However, lower values of OLR increase the risk of perceiving echo and too low OLR values may also (if present for many years) increase the risk of hearing damage for telephone operators.

Lower speech levels corresponding to an OLR somewhat higher than 10 dB is not as critical as high speech levels as long as the deviation from 10 dB is moderate. Somewhat too low speech levels will "only" lead to small deterioration of the perceived speech quality.

The recommended value of 10 dB is therefore justified based on the above discussion, and also considering the relatively flat speech quality curve around the optimum value of 7 dB.

A.3.3 Deviations from recommended OLR

Connections with very low OLR values (down to OLR = -5 dB) have been in use, e.g. in Europe for internal connection between telephone sets of the same PBX.

For the development of an appropriate migration strategy from too low to recommended OLR it should be taken into consideration that, customers may react strongly on abrupt changes in OLR, see ITU-T Recommendations G.101 [6], clause 4.2 and G.111 clause 3.2 for further guidance.

Connections via tandem analogue trunks may require higher insertion loss for stability and echo control reasons and are not likely to achieve an OLR close to 10 dB. Each decibel of increase in OLR (if applied to both directions of transmission) will account twice for the Talker Echo Loudness Rating (TELR), thus improving the suppression of any echo which may occur.

A.3.4 Network interface loudness ratings

The network interface loudness ratings are derived from the combination of terminal loudness ratings and nominal network losses. The example in figure A.4 shows the derivation of the network interface SLR & RLR for the ATT analogue network interface:

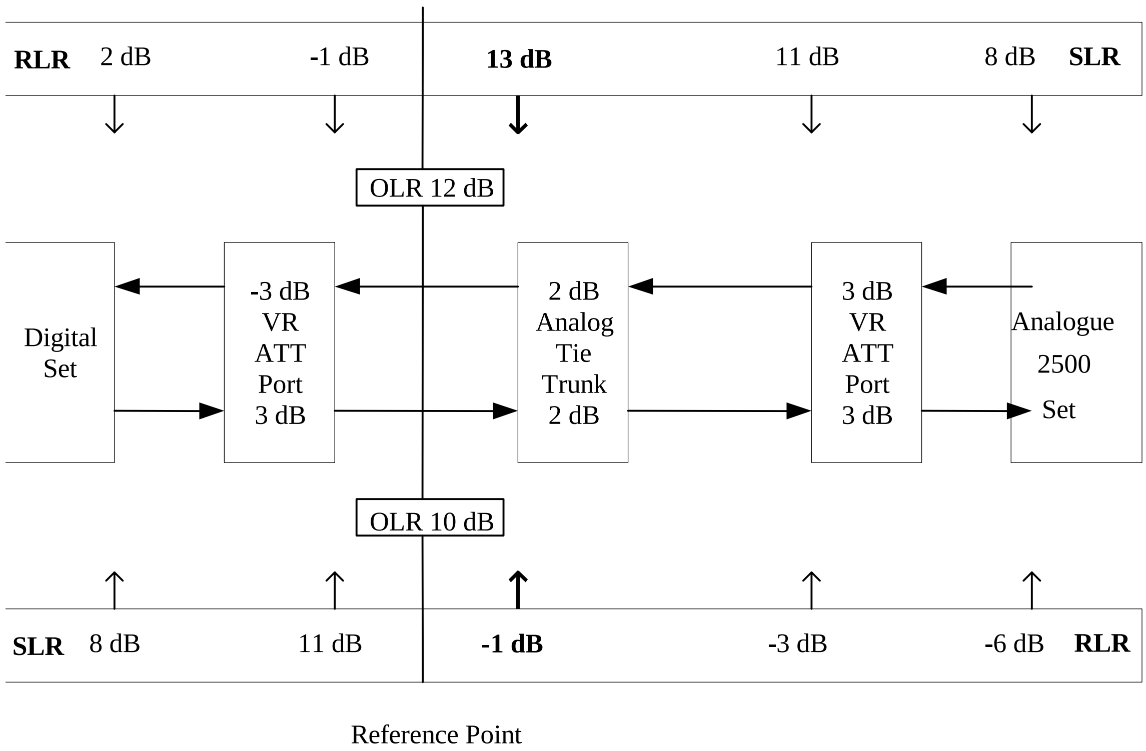

Figure A.4: ATT network interface loudness ratings

A.4 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 dB, loss = -9 dB).

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 +7 dBm, 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.

Annex B (informative): The Pan-European ELR and loss plan derivations

Annex B provides a spreadsheet for use with the present document as a means of calculation.

Annex B is contained in an Excel file (annex B.xls) contained in archive es_202020v010101m0.zip) which accompanies the present document.

Annex C (informative): The North American loss and level plan

C.1 Overview

The North American Voice Gateway loss and level plan is a sub-section of the Voice Gateway Transmission Plan. The Voice Gateway Transmission Plan in turn is related to, but independent from, the transmission section of ANSI/TIA/EIA-464-C "Requirements for PBX switching Equipment".

The full-channel loss plan defines the port-to-port losses for a matrix of three telephone types, four trunk types, and an IP network connection. Figure C.1 defines the connections and shows their relationships.

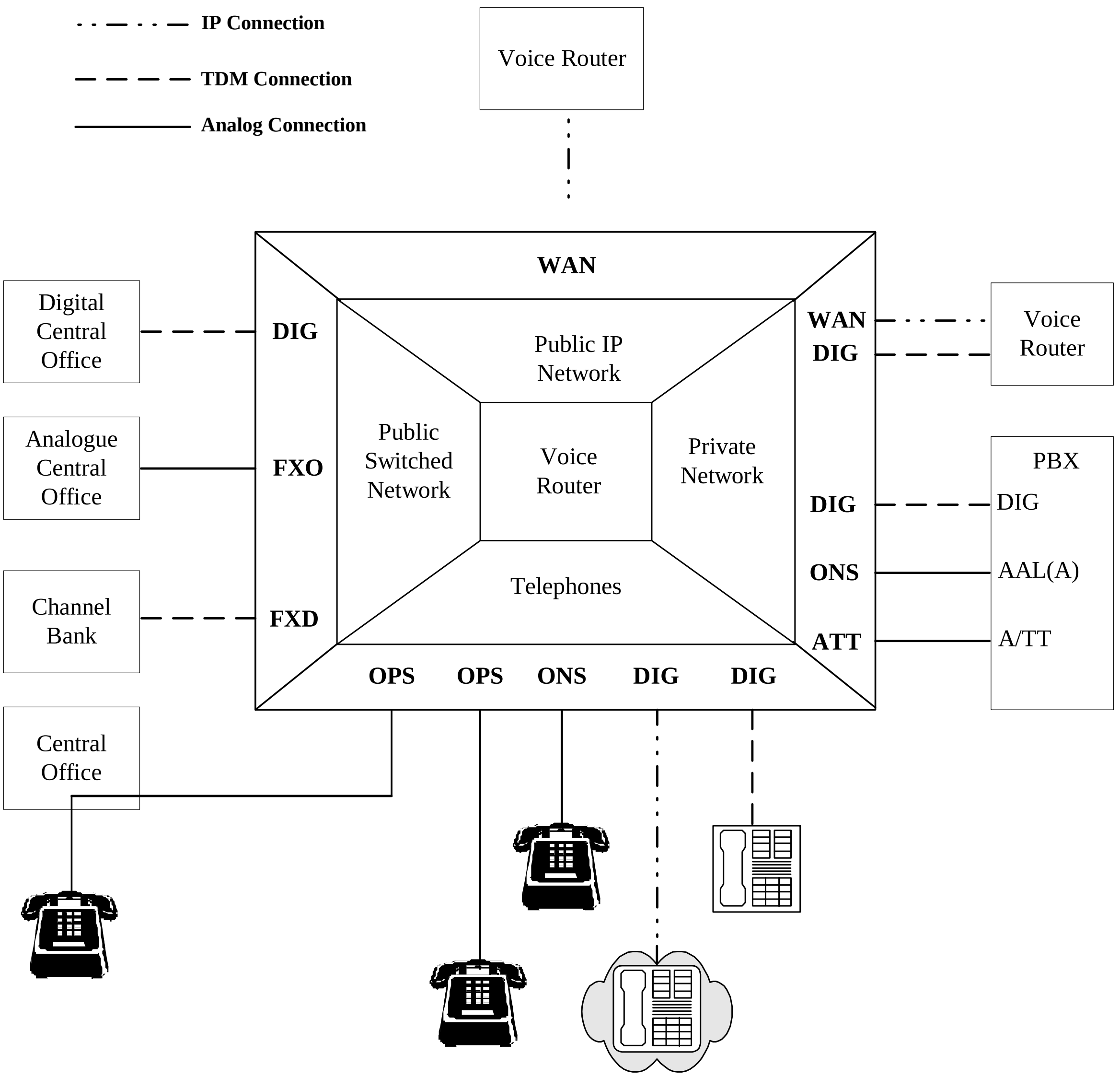

Figure C.1: Voice gateway connection definitions

C.2 Port descriptions

C.2.1 ONS – On Premise Station

An ONS interface is used for standard analogue 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.

C.2.2 OPS – Off Premise Station

An OPS interface is used for standard analogue telephones, representative of 2500-type telephones, located off the 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 analogue 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.

C.2.3 DGS – Digital Station

A DGS interface is used for all digital telephones, i.e. based on both TDM and packet transmission.

C.2.4 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 FCC Part 68.308(b)(5)(i).

C.2.5 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 FCC Part 68.308(b)(5)(i).

C.2.6 FXO – Foreign Exchange Office

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

C.2.7 FXD – Foreign Exchange Digital

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

C.2.8 ATT – Analogue Tie Trunk

An ATT interface is used for four-wire analogue private network connections 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.

C.3 The North American full-channel loss plan

C.3.1 Port-to-port loss table interpretation

In table C.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 1 B 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.

Table C.1: Voice gateway full-channel loss plan (North America)

			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: To prevent potential DTMF overload when the voice gateway is located less than 2 km from the central office, it is recommended that the FXD/WAN setting be used instead of the FXO/WAN setting. The FXD setting introduces an additional 3 dB loss in each direction.

C.3.2 Port-to-port loudness ratings table interpretation

Table C.2 is provided to show the relationship between OLR and SLR, voice gateway loss, and RLR.

For example, for an ONS to OPS connection, the ONS SLR is 8 dB, the voice gateway loss is 3 dB (from 1B in table C.2), and the OPS RLR is –3 dB. The overall loudness ratings is therefore 8 + 3 – 3 = 8 dB.

NOTE: Again, this is a loss plan, therefore negative values denote gain; e.g., -3 indicates 3 dB gain.

Table C.2: Voice gateway loudness ratings (North America)

			ONS	**OPS**	**DGS**	**WAN**	**DAL**	**FXO**	**FXD**	**ATT**
		RLR	-6	-3	2	2	2	3	0	-1
	SLR	**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 (SLR & RLR) and loss values 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 line loss.

Annex D (informative): The Pan-European loss and level plan

Historically, separate national transmission plan 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 [6], 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 no 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 E (informative): Bibliography

ITU-T Recommendation P.79: "Calculation of loudness ratings for telephone sets".

ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

ITU-T Recommendation G.712: "Transmission performance characteristics of pulse code modulation channels".

ITU-T Recommendation G.713: "Performance characteristics of PCM channels between 2-wire interfaces at voice frequencies".

ITU-T Recommendation G.714: "Separate performance characteristics for the encoding and decoding sides of PCM channels applicable to 4-wire voice-frequency interfaces".

ITU-T Recommendation G.715: "Separate performance characteristics for the encoding and decoding side of PCM channels applicable to 2-wire interfaces".

ITU-T Recommendation G.720: "Characterization of low-rate digital voice coder performance with non-voice signals".

ITU-T Recommendation G.721: "32 kbit/s adaptive differential pulse code modulation (ADPCM)".

ITU-T Recommendation G.722: "7 kHz audio-coding within 64 kbit/s".

ITU-T Recommendation G.723: "Extensions of Recommendation G.721 adaptive differential pulse code modulation to 24 and 40 kbit/s for digital circuit multiplication equipment application".

ITU-T Recommendation G.724: "Characteristics of a 48-channel low bit rate encoding primary multiplex operating at 1544 kbit/s".

ITU-T Recommendation G.725: "System aspects for the use of the 7 kHz audio codec within 64 kbit/s".

ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)".

ITU-T Recommendation G.727: "5-, 4-, 3- and 2-bit/sample embedded adaptive differential pulse code modulation (ADPCM)".

ITU-T Recommendation G.728: "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

ITU-T Recommendation G.729: "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)".

Federal Communications Commission (FCC) 47, CFR 68.500: "Code of Federal Regulations (USA); Title 47 Telecommunication; Chapter 1 Federal Communications Commission, Part 68 Connection of Terminal Equipment to the Telephone Network".

NOTE: This document (Federal Communications Commission (FCC) 47, CFR 68.500 can be obtained from:
Superintendent of Documents
Washington DC 20402
United States
Tel: +1 202 512 18003

ANSI/TIA/EIA-470-B: "Telecommunications - Telephone Terminal Equipment - Performance and Compatibility Requirements for Telephone Sets with Loop Signaling (ANSI/TIA/EIA-470-B-97)".

ANSI/TIA/EIA-810-A: "Telecommunications-Telephone Terminal Equipment-Transmission Requirements for Narrowband (ANSI/TIA/EIA-810-A-2000)".

ANSI/TIA/EIA TSB32-A: "Overall Transmission Plan Aspects for Telephony in a Private Network".

ANSI/TIA/EIA-464-C (draft): "Telecommunications - Multiline Terminal Systems - Requirements for PBX Switching Equipment".

History

Document history		
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