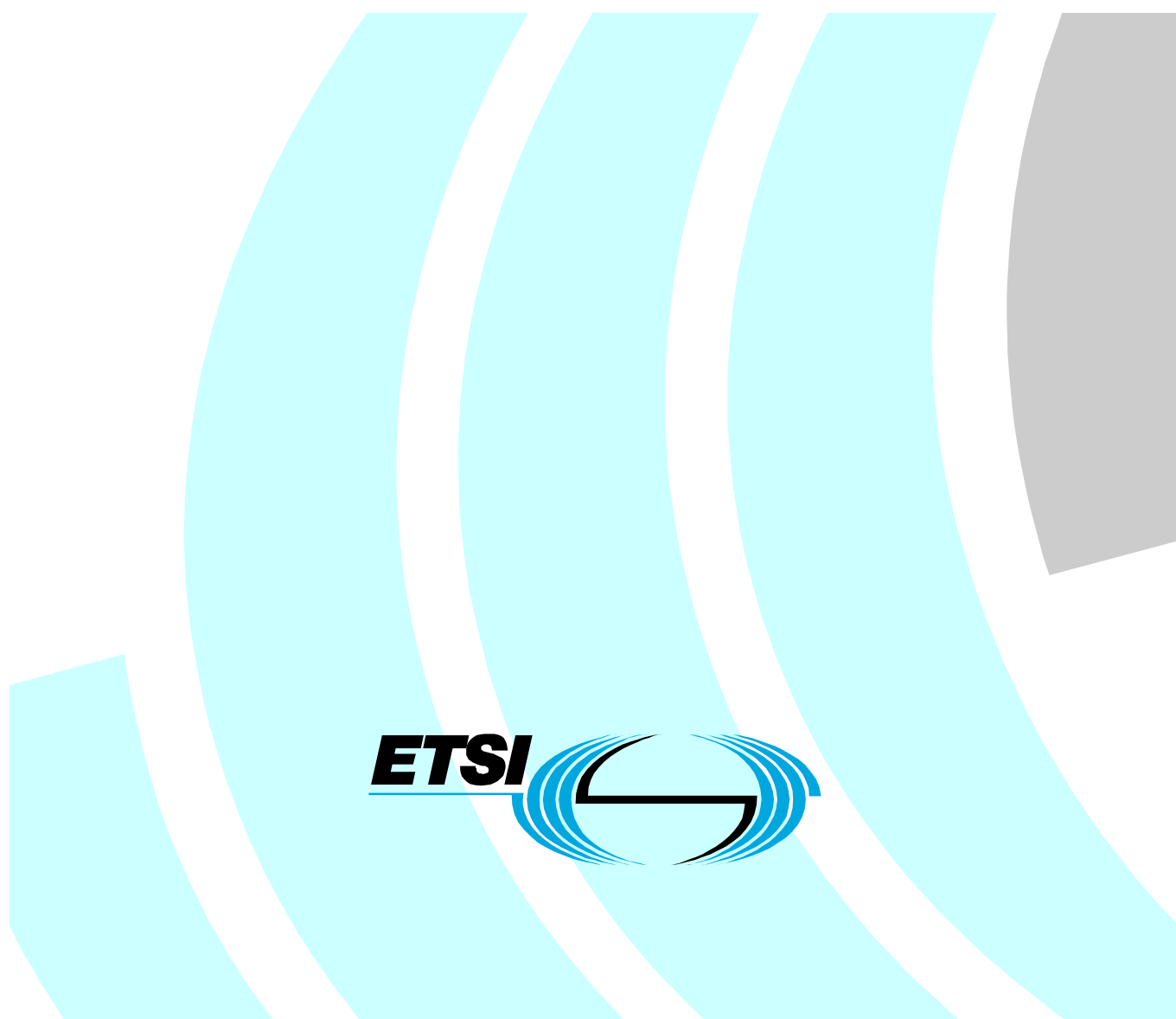


**Speech Processing, Transmission and Quality Aspects (STQ);  
Harmonized Pan-European/North-American approach to loss  
and level planning for voice gateways to IP based networks**

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Reference

RES/STQ-00032

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Keywords

QoS, quality, speech, transmission, voice

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## Foreword

This ETSI Standard (ES) has been produced by ETSI Technical Committee Speech Processing, Transmission and Quality Aspects (STQ), 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 Europe.

The recommended loss values are based on:

- Pan-European analogue telephone sets as specified in TBR 038 [3];
- Digital sets as specified in ITU-T Recommendation P.310 [7]; and
- Typical European public network losses.

These loss values may not provide optimum performance in conjunction with network operators whose 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 in 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].

NOTE: The present document introduces two variants of the L2 interfaces for transmission planning purposes.

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 interfacing 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.

Referenced documents which are not found to be publicly available in the expected location might be found at <http://docbox.etsi.org/Reference>.

- [1] TIA-912: "Telecommunication s IP Telephony Equipment Voice Gateway Transmission Requirements".
- [2] ETSI ES 201 168 (V1.2.1): "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] ETSI TBR 038 (1988): "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".
- [4] ITU-T Recommendation G.100: "Definitions used in Recommendations on general characteristics of international telephone connections and circuits ".
- [5] ITU-T Recommendation G.101: "The transmission plan".
- [6] ITU-T Recommendation G.107: "The E-Model, a computational model for use in transmission planning".
- [7] ITU-T Recommendation P.310: "Transmission characteristics for telephone band (300-3 400 Hz) digital telephones".
- [8] ITU-T Recommendation G.111: "Loudness ratings (LRs) in an international connection".
- [9] ITU-T Recommendation G.121: "Loudness ratings (LRs) of national systems".

---

## 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)

**receive loudness rating:** electrical/acoustic conversion characteristic of the terminating equipment

**loudness rating:** send and receive loudness ratings are collectively referred to as the loudness rating of a telephone

**equivalent loudness rating:** equivalent loudness rating of a interface is the send loudness rating or receive loudness rating of the terminal connected to that interface, plus any gain or loss in the connection between the terminal and the interface

**IP send loudness rating:** specific designation for the equivalent send loudness rating at the reference point

**Reference level point:** VG-to-IP network connection point (also termed the zero-level point)

**reference level:** IP send loudness rating 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.729A (see bibliography).

**telephony terminal:** 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 each loss is defined for a complete path between two acoustical interfaces

**half-channel loss plan:** loss plan where each loss is defined for a path between an acoustical interface and an (electrical) standard reference point

**interface:** Pan-European term for the input or output circuit connections

**port:** North American term for the input or output circuit connections

NOTE: The terms interface and port are used interchangeably in the present document

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

NOTE: It may provide interfaces to analog and digital voice terminals and access to public and private switched telephone networks and WANs.

## 3.2 Abbreviations

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

ELR	Equivalent Loudness Rating
ESLR	Equivalent Send Loudness Rating
iSLR	IP Send Loudness Rating
LR	Loudness Rating
OLR	Overall Loudness Rating
RLR	Receive Loudness Rating
SLR	Send Loudness Rating
VG	Voice Gateway
ZLP	Zero-Level Point

---

## 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 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.

VGs can also connect to existing analogue and TDM based digital networks. In North America these connections require losses to be defined on a port-to-port basis for technical and sometimes regulatory reasons, and full-channel loss plans are required for VGs.

### 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.

NOTE: Clause 5 of the present document describes the half-channel loss implementation in more detail.

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 1/3 octave band. See ITU-T Recommendation G.100 [4].

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 VG interface (it can be assumed that a digital interface would have a LR of 8 dB and 2 dB). The ELR of a interface is the SLR or RLR of the terminal connected to that interface, plus any gain or loss in the connection between the terminal and the interface.

EXAMPLE: An analogue telephone with an LR of 3 dB and -8 dB, connected to an L2 interface of a VG via a 2-wire line with 3 dB loss, would have an ELR of 6dB and -5 dB as shown in figure 2.

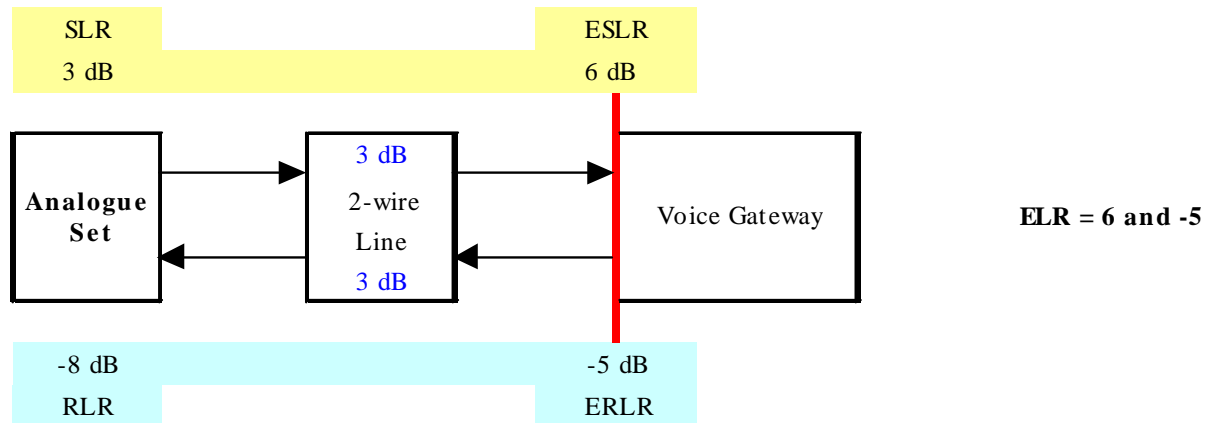


Figure 1: Equivalent loudness rating example

### 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 VG-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.

## 4.5 Reference level

The reference level for IP telephony is an iSLR of 8 dB.

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

### 4.5.1 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.

## 4.6 Harmonized half-channel loss and level plans

The present document gives the requirements for the harmonized half-channel loss and level plans 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 ELR of each interface;

NOTE: The ELR of an interface with no circuit loss between the terminal and the interface is equivalent to the terminal LR.

- the transmit losses required to achieve an iSLR of 8 dB for each interface;
- the receive losses required to achieve a satisfactory OLR for each receiving end point.

## 4.7 Regional full-channel loss and level plans

Annexes B and C of the present document provide the full-channel loss and level plans for two regions, the Pan-European region and the North American region.

Each full-channel plan defines:

- the ELR of each interface;
- the interface-to-interface loss for each interface to every other interface;
- The OLR from each interface to every other interface based on the LRs or ELRs, and the corresponding interface-to-interface loss.

---

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

As stated in clause 4.2, 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.

The half-channel loss plan operates on the following principles:

- 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.

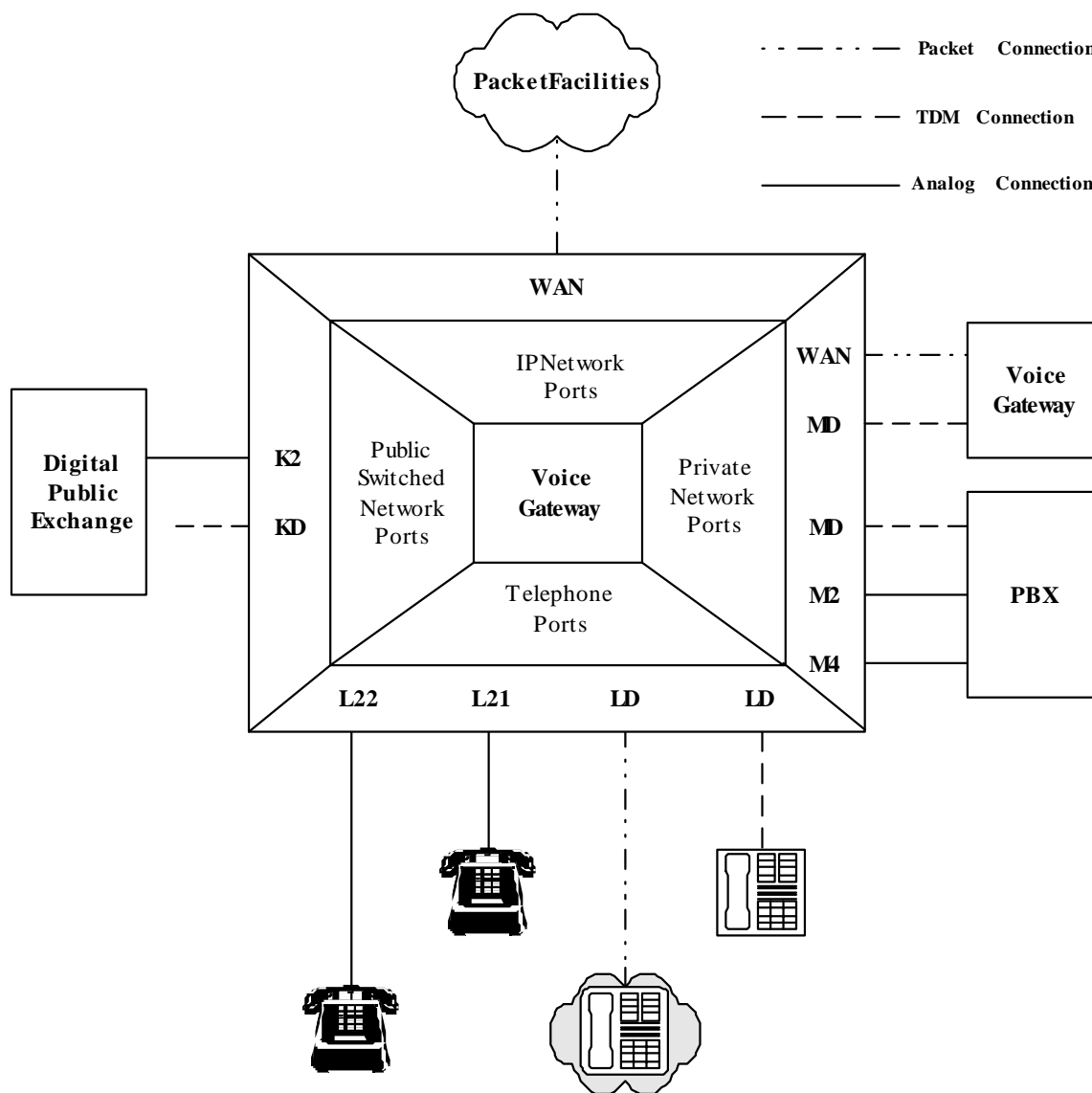
The advantage of this approach is that neither entity requires knowledge of the other, and loss planning becomes a local issue.

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

### 5.1.1 Voice Gateway Connections

Figure 2 illustrates the connection types for a typical Pan-European VG 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 interface losses to be applied.



**Figure 2: Pan-European Voice Gateway Connections**

NOTE: Figure 2 is based in part on figures 1 and 2 of ES 201 168 [2].

## 5.1.2 Interface Descriptions

The following descriptions were adapted in part from clause 3.1.9 of ES 201 168 [2].

### Telephone Interfaces

#### L2 - 2-wire Analogue

The L2 interface provides for the connection of 2-wire analogue extension lines and will carry signals such as speech, voice-band analogue data and DTMF signals, etc. In addition, the interface L2 provides for ordinary functions such as direct current (DC) feeding, DC signalling, ringing, etc.

NOTE 1: The L2 interface is subdivided into short lines and long lines for transmission planning purposes. These are designated L21 and L22 respectively, and are equivalent to the North American ONS and OPS ports.

#### LD - Digital

The LD interface provides for the connection of digital terminals that conform to the LR requirements of ITU-T Recommendation P.310 (see clause 2).

NOTE 2: This definition differs from the one given in ES 201 168, by also including system specific terminals.

### Public Network Interfaces

#### K2 - 2-wire Analogue

The K2 interface provides for the connection of 2-wire analogue subscriber lines between a VG and a public exchange.

#### KD - Digital

The KD interface provides for the connection of a digital access to the public switched network.

### Private Network Interfaces

#### M2 - 2-wire Analogue

The M2 interface provides for the connection to 2-wire analogue circuits (e.g. leased lines) between a VG and a PBX.

#### M4 - 4-wire Analogue

The M4 interface provides for the connection to 4-wire analogue circuits (e.g. leased lines) between a VG and a PBX.

NOTE 3: The M4 interface is used as the generic private network analogue interface for loss planning purposes. The required losses for the M4 and M2 interfaces are the same.

#### MD - Digital

The interface MD provides for connection to a digital inter-VG or PBX circuit.

NOTE 4: The MD interface shares the KD designation for loss planning purposes (they are both digital connection with zero loss).

### Packet Network Interfaces

#### WAN – Wide Area Network

A WAN interface connects from VGs to packet-based wide area networks.

NOTE 5: The term WAN is used in the same context as the public switched network, in that it represents connections between geographically separated VGs. It should be noted that WAN and LAN are synonymous from a transmission perspective.

### 5.1.3 Voice Gateway Half-Channel Loss Plan

Table 1 shows the VG 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		Assumed OLR
L21	3	5	8		-8	10	10		10
L22	6	2	8		-5	7	10		10
LD	8	0	8		2	0	10		10
WAN	8	0	8		2	0	10		10
KD	8	0	8		2	0	10		10
K2	16	-6	10		5	-1	14		12
M4	10	-2	8		4	-2	10		10

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

Column **b** shows the VG 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 VG 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 1: It is not possible to achieve the optimum iSLR of 8 dB for connections from analogue networks due to the potential for loop instability.

NOTE 2: 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

### 5.2.1 Voice Gateway Connections

Figure 3 illustrates the connection types for a typical North American 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.

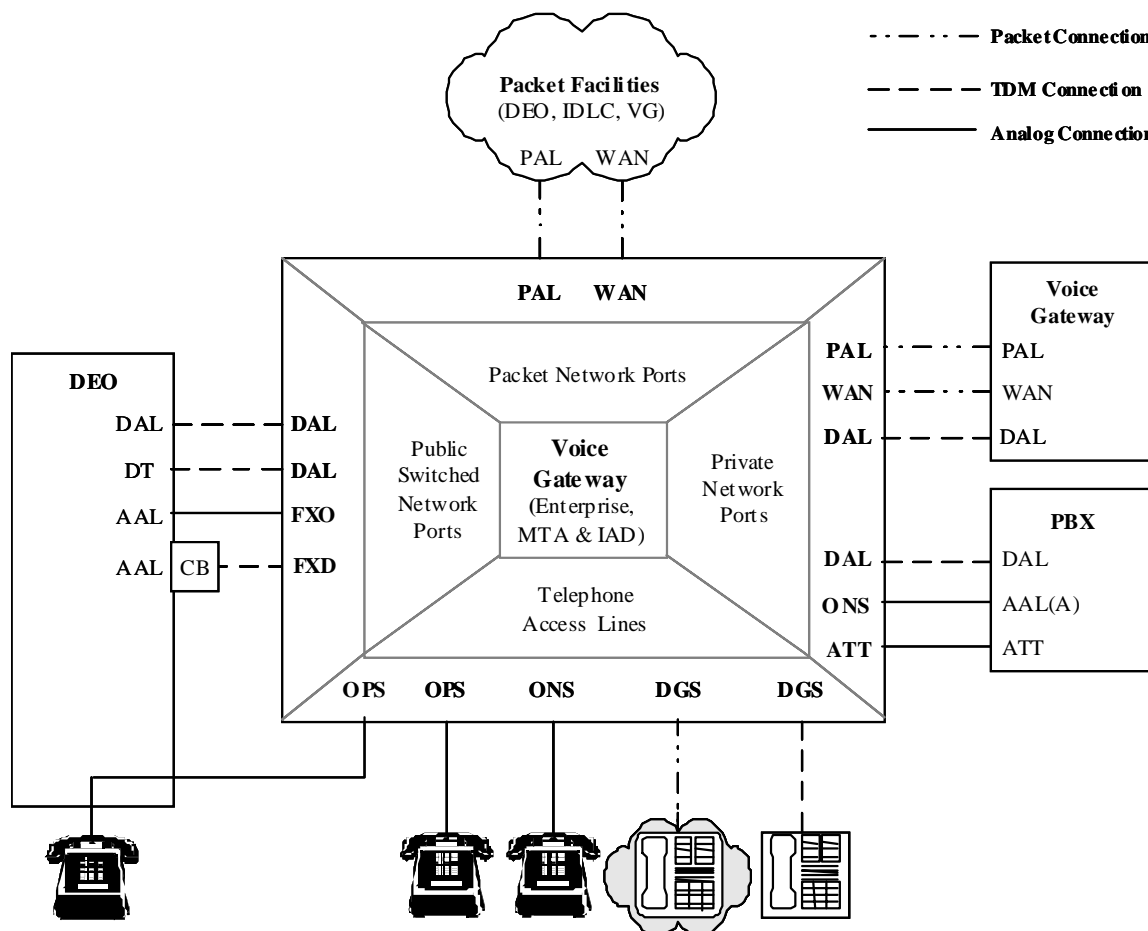


Figure 3: North American Voice Gateway Connections

## 5.2.2 Port Descriptions

### **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 VG, and is the direct equivalent of the PBX ONS connection. The connection loss from the station to the VG is typically low. The term FXS is sometimes used in place of ONS.

### **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 VG, and is the direct equivalent of the PBX OPS connection. The connection loss from the station to the VG is typically significant. This port is also used for analog two-wire connections to remote VGs, PBXs and Key Systems, via a local DEO. The term FXS is sometimes used in place of OPS.

### **DGS - Digital Station**

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

NOTE 1: This includes digital telephones based on both TDM and packet transmission.

### **WAN - Wide Area Network**

A WAN interface connects from Enterprise VGs to packet-based wide area networks. The transmission path within the WAN is entirely digital. The only distinction between a WAN port and a PAL port is that the PAL port supports less bandwidth.

NOTE 2: The term WAN is used in the same context as PSTN, in that it represents connections between geographically separated VGs. It should be noted that WAN and LAN are synonymous from a transmission perspective.

### **PAL - Packet Access Line**

A PAL interface connects from MTA or IAD VGs to packet-based wide area networks. The transmission path within the PAL is entirely digital. The only distinction between a WAN port and a PAL port is that the PAL port supports less bandwidth.

### **DAL - Digital Access Line**

A DAL interface connects to all TDM-based digital 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.

### **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).

### **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 VG, as the channel bank is located close to the central office.

### **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 VG 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.

### 5.2.3 Voice Gateway Half-Channel Loss Plan

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

**Table 2: Voice gateway half-channel loss plan**

	WAN/PAL Zero-Level Point ↓			d	e	f	g
	a	b	c				
	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
WAN	8	0	8	2	0	10	10
PAL	8	0	8	2	0	10	10
DAL	8	0	8	2	0	10	10
FXO	20	-6	14	6	0	20	14
FXD	17	-3	14	3	3	20	14
ATT	13	0	13	-1	3	15	10

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

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/PAL).

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 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 is based on one in the North American Voice Gateway Transmission Requirements standard, modified for 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.

---

### 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.

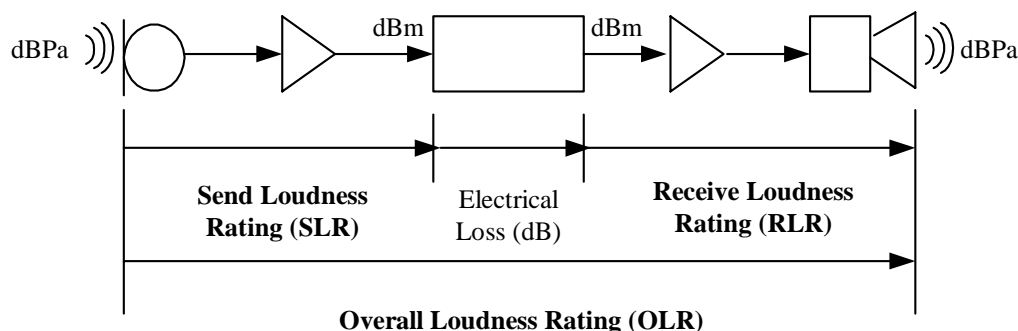
The send and receive levels of a telephone relate the conversion of acoustic pressure to electrical power and vice versa. The acoustic pressure units are in dBPa (Pascals), and the electrical power units are in dB mW.

Pressure is measured in Newtons per square meter (Pascals), and the relationship between dB SPL and dB Pa is shown in the following table.

dB SPL	dB Pa	Parameter
94	0	One Pascal
89,3	-4,7	Average speech level at the MRP
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
L21	3	-8	1
L22	6	-5	2
LD	8	2	3

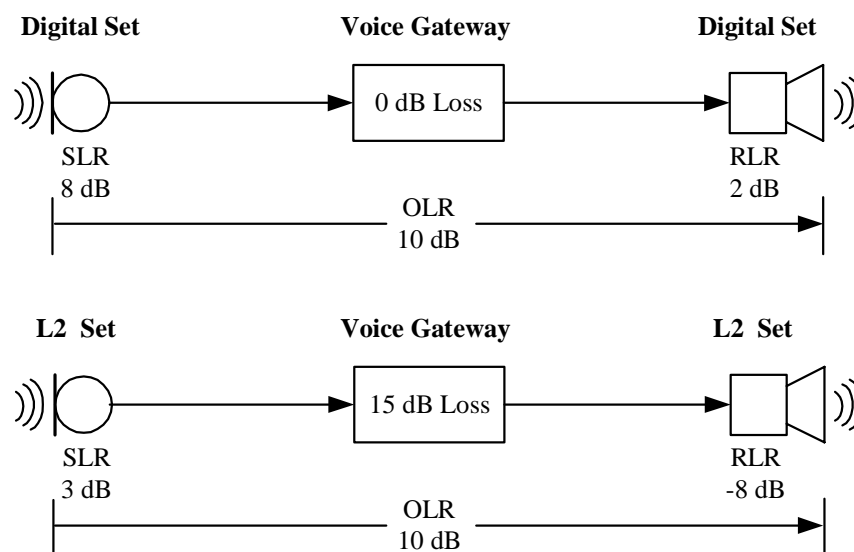
NOTE 1: The L21 loudness ratings are representative of TBR 038 [3] type analogue telephones operating on short lines with the typical battery feed and  $270 \Omega + (750 \Omega \parallel 150 \text{ nF})$  impedance characteristics of VG L2 ports.

NOTE 2: The L22 loudness ratings are representative of TBR 038 [3] type analogue telephones operating on nominal 2,5 km lines with 3 dB of loss and the typical battery feed and  $270 \Omega + (750 \Omega \parallel 150 \text{ nF})$  impedance characteristics of VG L2 ports.

NOTE 3: The LD loudness ratings of SLR = 8 dB and RLR = 2 dB conform to the requirements specified in ITU-T Recommendation P.310 [7]

### Overall loudness ratings

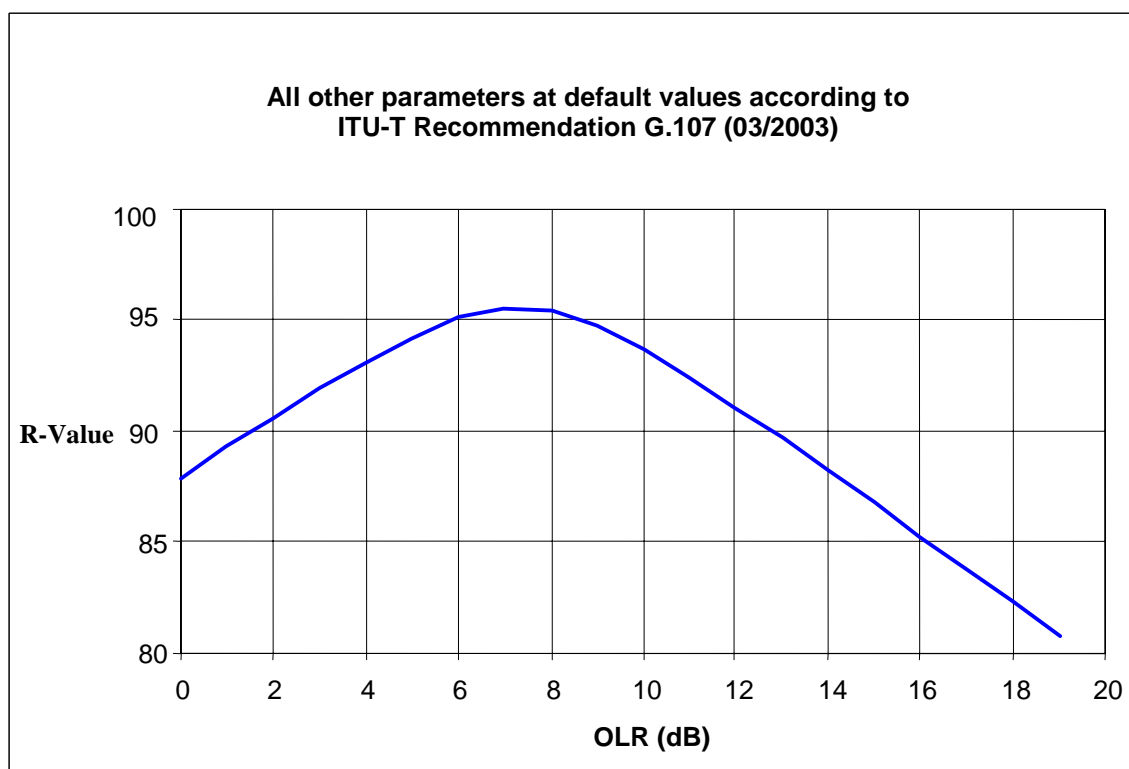
The Overall Loudness Rating (OLR) of a connection is the sum of the sending terminal SLR, any system or network loss, and the receiving terminal RLR. This is illustrated in figure A.2 for set-to-set calls within a VG.



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

### 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 1 are at or above an R-Value of 90, which puts them in the "very satisfied" category.



**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 [[8]).

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.

### 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 Recommendation G.101 [5], clause 4.2 and G.111 [[8] 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.

### 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 A.4 shows the derivation of the network interface ESLR & ERLR for the K2 analogue network interface.

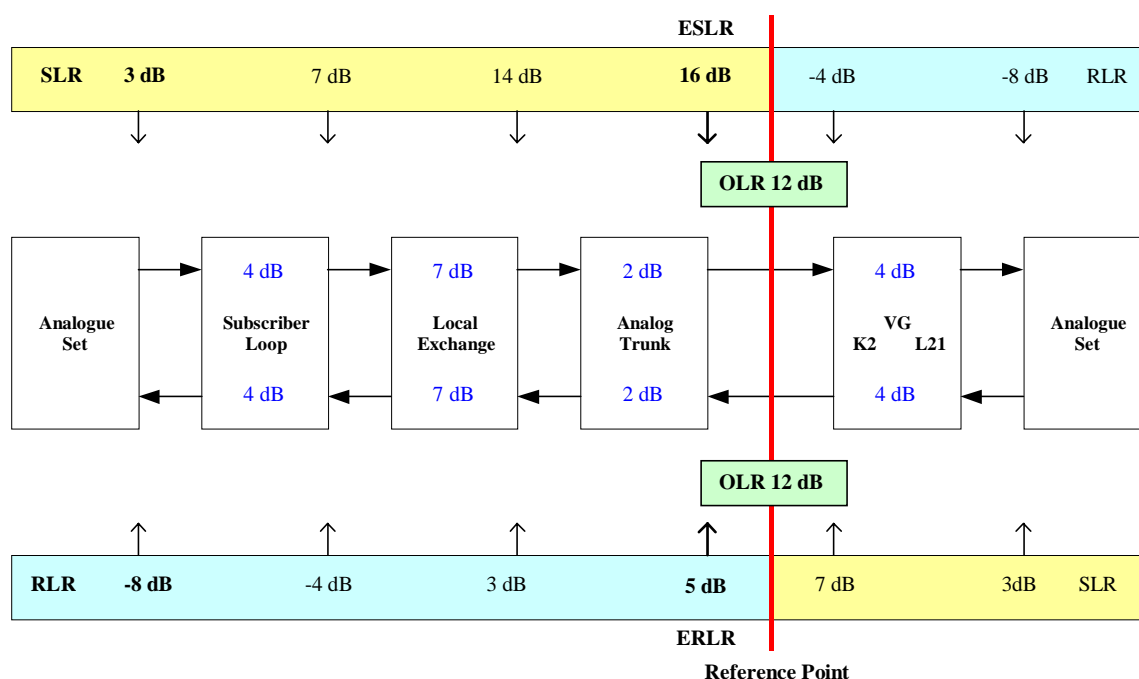


Figure A.4: K2 network interface equivalent loudness ratings

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## A.4 Interface-to-interface loss allocation

NOTE 1: The following discussion is only relevant to the North American full-channel loss plan. The use of the half-channel transmit and receive losses for the Pan-European full-channel plan ensures optimum levels at the zero-level point.

It should be noted, that the actual allocation of the interface-to-interface 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 2: 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 89 dB SPL, the average power level at the ZLP would be approximately -2 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 +8 dBm, resulting in clipping.

Conversely, an ONS input loss of 9 dB, and an OPS output loss of -6 dB, would result in lower power at the ZLP, and a reduction in the signal-to-noise ratio.

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

### B.1 Overview

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 [5], G.111 [[8] and G.121 [9]. Hence, there was no reason to issue a Pan-European Loss and Level Plan.

Regulatory treatment of a telephony connection in Europe consists 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. To this end a Pan-European full-channel loss plan has been developed using the ELR values derived for the half-channel loss plan. This full-channel plan is has been developed to assist manufacturers in achieving satisfactory voice performance, and is not a regulatory requirement.

The full-channel loss plan analyses showed that, unlike the North American case, the same transmit and receive losses could be used for both the half-channel and full-channel plans, and hence these losses could be fixed.

Annex B.2 provides information how the Pan-European ELRs were derived.

Annex B.3 contains the Pan-European full-channel loss plan.

Annex B.4 is the Excel spreadsheet used to derive the Pan-European ELRs, and the half-channel and full-channel loss plans.

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### B.2 The Pan-European ELR and loss plan derivations

Clause B.4 provides a spreadsheet for use with the present document as a means of calculation.

The ELR values derived in this spreadsheet are nominal values based on a theoretical European model. The ELR values are necessary inputs to the Pan-European half-channel loss plan in order to determine the required loss or gain required in the VG to achieve the desired iSLR of 8 dB.

The nominal telephone loudness ratings of 3 and -8 are taken from TBR 038 [3].

The nominal 4 dB line loss and 7 dB local exchange Rx pad values are taken from a background paper on TBR 038 [3].

Another paper derives the nominal 4 dB line loss as 3 km of 0,5 mm cable with a typical loss of 1,25 dB per km. (see Paper 2 in the bibliography).

The L22 ELR derivation assumes a 2,5 km line with a loss of 3 dB (equivalent to the North American off-premise station).

Information on European dBr levels was derived from ES 201 168 [2].

This annex also provides examples of the impact of various national set loudness ratings and PSTN losses on the ELR derivations. ELR derivations for the UK and the Netherlands were done as examples of two countries at the opposite ends of the European SLR range, and French ELR derivation was done as an example of an RLR at the top end of the range. These examples demonstrate that the application of the Pan-European loss plan should result in satisfactory OLR performance in most regions in Europe.

## B.3 The Pan-European full-channel loss plan

### Full-channel loss plan interpretation

Table B.1 shows the interface-to-interface loss in the Pan-European full-channel loss plan. 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 interface-to-interface loss of 12 dB from the L21 interface to the L22 interface, and co-ordinate 2A indicates a 12 dB loss in the other direction, from the L22 interface to the L21 interface.

NOTE 1: This is a loss plan, therefore negative values denote gain; e.g., -6 indicates 6 dB gain.

**Table B.1: Voice gateway full-channel loss plan (Pan-European)**

			A	B	C	D	E	F	G
			L21	L22	LD	WAN	KD	K2	M4
		Loss (dB)	↑	↑	↑	↑	↑	↑	↑
1	L21	→	15	12	5	5	5	4	3
2	L22	→	12	9	2	2	2	1	0
3	LD	→	10	7	0	0	0	-1	-2
4	WAN	→	10	7	0	0	0	-1	-2
5	KD	→	10	7	0	0	0	-1	-2
6	K2	→	4	1	-6	-6	-6	-7	-8
7	M4	→	8	5	-2	-2	-2	-3	-4

NOTE 2: The interface-to-interface losses are the sum of the transmit and receive half-channel losses defined in clause 5.1.3, table 1.

### Full-channel loudness ratings table interpretation

Table B.2 is provided to show the relationship between OLR, ESLR, ERLR and interface-to-interface loss. For example, for an L21 to L22 connection, the L21 ESLR is 3 dB, the VG loss is 12 dB (from 1B in table B.1), and the L22 ERLR is -5 dB. The overall loudness ratings is therefore  $3 + 12 - 5 = 10$  dB.

**Table B.2: Voice gateway full-channel loudness ratings (Pan-European)**

			L21	L22	LD	WAN	KD	K2	M4
		ERLR	-8	-5	2	2	2	5	4
		ESLR	OLR	↑	↑	↑	↑	↑	↑
L21	3	→	10	10	10	10	10	12	10
L22	6	→	10	10	10	10	10	12	10
LD	8	→	10	10	10	10	10	12	10
WAN	8	→	10	10	10	10	10	12	10
KD	8	→	10	10	10	10	10	12	10
K2	16	→	12	12	12	12	12	14	12
M4	10	→	10	10	10	10	10	12	10

## B.4 Calculation spreadsheet

Clause B.4 is contained in an Excel file (ES 202 020 V1-2-1 Annex.xls contained in archivees\_202020v010201p0.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 TIA-912 [1] which in turn is related to, but independent from, the transmission section of ANSI/TIA/EIA-464-C.

### C.2 The North American full-channel loss plan

#### Enterprise, MTA, and IAD Voice Gateways

The North American full-channel loss plan differentiates between Enterprise VGs and Integrated Access Device (IAD) and Multimedia Terminal Adaptor (MTA) VGs. An enterprise VG typically provides the same full switching capabilities that a PBX would, while an IAD or MTA would primarily provide network access only with little or no local switching. Two loss plans are therefore provided, one for enterprises VGs and one for IAD and MTA VGs.

NOTE 1: Unlike the Pan-European case, the same Enterprise VG transmit and receive losses could not be used for both the half-channel and full-channel plans, and hence the full-channel losses have to be changed on a per-connection basis.

#### Port-to-port loss table interpretation

In table C.2.1, 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 2: This is a loss plan, therefore negative values denote gain; e.g., -3 indicates 3 dB gain.

**Table C.1: Enterprise 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 (dB) (see note 1)		↑	↑	↑	↑	↑	↑	↑	↑
1	ONS →	6	3	0	0	0	0	3	3
2	OPS →	3	0	-3	-3	-3	-3	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	0	3	3
6	FXO →	0	0	-6 (see note 2)	-6 (see note 2)	-6 (see note 2)	0	0	0
7	FXD →	3	0	-3 (see note 2)	-3 (see note 2)	-3 (see note 2)	0	0	0
8	ATT →	3	0	0	0	0	0	0	0

NOTE 1: Losses have been selected as multiples of 3 dB, assuming that this simplifies the implementation.  
NOTE 2: There is a potential risk of DTMF overload if an FXO analog trunk is connected to DGS, WAN or DAL and the VG is located less than 2 km from the DEO due to the gain added in the A/D direction. In these cases it is recommended that the FXD to DGS, WAN or DAL setting be used instead because the FXD setting introduces 3 dB less gain in one direction and 3 dB more loss in the other direction.

### Port-to-port loudness ratings table interpretation

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

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

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

**Table C.2: Enterprise voice gateway loudness ratings (North America)**

			ONS	OPS	DGS	WAN	DAL	FXO (see note 1)	FXD (see note 1)	ATT (see note 2)
	(dB)	ERLR	-6	-3	2	2	2	6	3	-1
	ESLR	OLR	↑	↑	↑	↑	↑	↑	↑	↑
ONS	8	→	8	8	10	10	10	14	14	10
OPS	11	→	8	8	10	10	10	14	14	13
DGS	8	→	11	11	10	10	10	14	14	10
WAN	8	→	11	11	10	10	10	14	14	10
DAL	8	→	11	11	10	10	10	14	14	10
FXO (see note 1)	20	→	14	17	16	16	16	26	23	19
FXD (see note 1)	17	→	14	14	16	16	16	23	20	16
ATT (see note 2)	13	→	10	10	15	15	15	19	16	12

NOTE 1: The loudness ratings for the FXO and FXD ports include a nominal 6 dB DEO loss.  
NOTE 2: The loudness ratings for the ATT port include a nominal 2 dB trunk loss.

**Table C.3: MTA/IAD voice gateway loudness ratings (North America)**

		A	B	C	D
		ONS	DGS	PAL	DAL
Loss (dB) (see note 1)		↑	↑	↑	↑
1	ONS	→	9	0 (see note 2)	0 (see note 2)
2	DGS	→	9	0	0
3	PAL	→	9	0	0
4	DAL	→	9	0	0

NOTE 1: Losses have been selected as multiples of 3 dB, assuming that this simplifies the implementation. For the optimum OLR of 10 dB, the "to ONS" value is actually 8 dB.  
NOTE 2: For MTAs/IADs with DTMF receivers that have issues with overload on very short ONS loops, an additional A/D loss of up to 2 dB may be inserted.

**Table C.4: MTA/IAD voice gateway loudness ratings (North America)**

		ONS	DGS	PAL	DAL
(dB)		-6	2	2	2
ESLR		↑	↑	↑	↑
ONS	8	→	11	10	10
DGS	8	→	11	10	10
PAL	8	→	11	10	10
DAL	8	→	11	10	10

## Annex D (informative): Relationship of channel based loss plans to Q.551

There are certain similarities in the use of the terms half-channel as used in the present document, and half-connection as used in ITU-T Recommendation Q.551 (see bibliography).

The following defines connections in the context of ITU-T Recommendation Q.551, and figure D.1 shows the relationship between the terms full and half connection, as defined in ITU-T Recommendation Q.551 (see bibliography), and full and half channel as defined in the present document.

**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

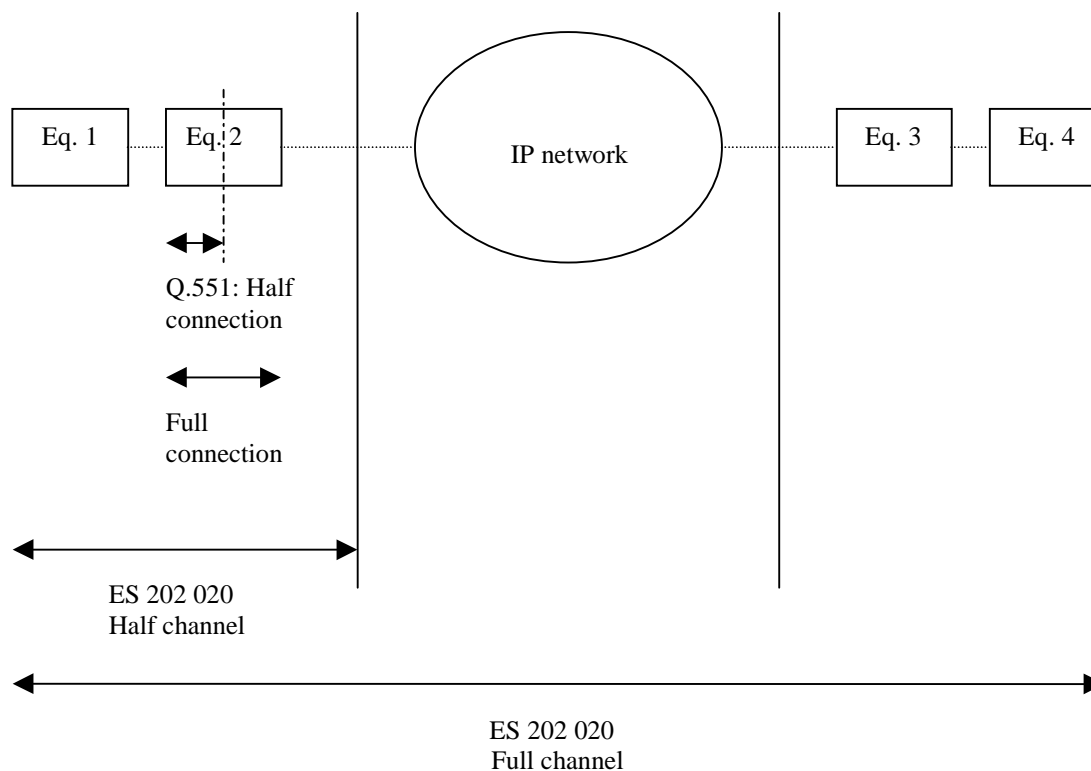
**half-connection:** bi-directional path comprised of an input connection and an output connection, both having the same exchange interface

**full-connection:** bi-directional path comprised of two half-connection, providing a complete connection through 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.



**Figure D.1: Connection versus channel**

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## 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.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.724: "Characteristics of a 48-channel low bit rate encoding primary multiplex operating at 1 544 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)".

ITU-T Recommendation G.729 Annex A: "Reduced complexity 8 kbit/s CS-ACELP speech codec".

TIA-968-A-2002: "Technical Requirements for Connection of Terminal Equipment to the Telephone Network".

TIA-968-A-1-2003: "Technical Requirements for Connection of Terminal Equipment to the Telephone Network".

NOTE: TIA-968-A and Annex A-1 replaced Federal Communications Commission (FCC) 47, CFR 68.500: "Part 68 Connection of Terminal Equipment to the Telephone Network" in the year 2002.

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)".

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

Directive 1999/5/EC of the European Parliament and of the Council of 9 March 1999 on radio equipment and telecommunications terminal equipment and the mutual recognition of their conformity (R&TTE Directive)

Paper 1: "A strategy for the harmonisation of telephone loudness requirements", Undated paper by W.J. Mellors.

Paper 2: "(GPT) - Personal communication", W. J. Mellors.

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## History

<b>Document history</b>		
V1.1.1	February 2002	Publication
V1.2.1	October 2003	Membership Approval Procedure    MV 20031205: 2003-10-07 to 2003-12-05