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ETSI Standard

Speech and multimedia Transmission Quality (STQ); Transmission Requirements for IP-based Narrowband and Wideband Home Gateways and Other Media Gateways from a QoS Perspective as Perceived by the User

Reference
DES/STQ-00145

Keywords
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650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
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Foreword

This final draft ETSI Standard (ES) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ), and is now submitted for the ETSI standards Membership Approval Procedure.

Introduction

Traditionally, the analogue and digital telephones were interfacing switched-circuit 64 kbit/s PCM networks. With the fast growth of IP networks, packet-switched networks (VoIP) interfacing PSTN networks and mobile networks, as well as different types of IP-terminals, are being rapidly introduced. Different types of gateways are used to interconnect to such IP networks. Since the IP networks will be in many cases interworking with the traditional PSTN and private networks, many of the basic transmission requirements have to be harmonized between these different types of network from an end-to-end perspective, including specifications for the edge points.

The present document covers IP-based narrowband and wideband home gateways and other media gateways. It aims to enhance the interoperability and end-to-end quality.

In contrast to other standards which define minimum performance requirements, it is the intention of the present document to specify gateway equipment requirements which enable manufacturers and service providers to enable end-to-end speech performance as perceived by the user. These requirements are absolutely necessary to ensure a good quality, but they are not sufficient. They have to be combined with requirements (and associated relevant measurement methods) for other elements in the transmission chain (core IP network, PSTN, terminals), as well as for the whole mouth-to-ear transmission path.

1 Scope

The present document provides speech transmission performance requirements for narrowband and wideband media gateways from a QoS perspective as perceived by the user. Media gateways can be network or home based, they may include a transcoding function. The present document covers the following types of IP-based media gateways:

- ATA (Analogue Terminal Adapter), home gateway IP to POTS
- ITA (ISDN Terminal Adapter), home gateway IP to ISDN
- IAD (Integrated Access device), home router including ATA or ITA
- Network based ATA and ITA
- Carrier grade media gateway, network gateway IP to TDM
- IP-to-IP media gateway, network gateway with transcoding and/or other media processing

DECT interfaces of media gateways are excluded from the present document and should be measured according to the relevant DECT standards.

Interfaces of media gateways used together with terminals as a system (i.e. connected via Ethernet or with a proprietary interface) are excluded in the present document and should be measured according to the relevant terminal standard.

If a media gateway includes more than one interface type (e.g. POTS and ISDN), each interface has to be dealt with differently.

The requirements available in the present document will ensure a high compatibility with IP-and TDM-based fixed and wireless terminals and networks, including DECT and mobile terminals.

It is the aim to optimize interoperability, the listening and talking quality and the conversational performance. Related requirements and test methods are defined in the present document.

The present document does not apply to media gateways with 4-wire analogue interfaces.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the reference document (including any amendments) applies.

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

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are necessary for the application of the present document.

[1] ETSI EN 300 726: "Digital cellular telecommunications system (Phase 2+) (GSM); Enhanced Full Rate (EFR) speech transcoding (GSM 06.60)".

[2] ETSI TS 126 171: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); AMR speech codec, wideband; General description (3GPP TS 26.171 version 6.0.0 Release 6)".

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

[4] ITU-T Recommendation G.108: "Application of the E-model: A planning guide".

[5] ITU-T Recommendation G.109: "Definition of categories of speech transmission quality".

[6] ITU-T Recommendation G.100.1: "The use of the decibel and of relative levels in speechband telecommunications".

[7] ITU-T Recommendation G.111: "Loudness Ratings (LRs) in an international connection".

[8] ITU-T Recommendation G.122: "Influence of national systems on stability and talker echo in international connections".

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

[10] ITU-T Recommendation G.723.1: "Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s".

[11] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

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

[13] ITU-T Recommendation G.729.1: "G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729".

[14] ITU-T Recommendation G.1020: "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".

[15] ITU-T Recommendation P.50: "Artificial voices".

[16] ITU-T Recommendation P.340: "Transmission characteristics and speech quality parameters of hands-free terminals".

[17] ITU-T Recommendation P.501: "Test signals for use in telephonometry".

[18] ITU-T Recommendation P.502: "Objective test methods for speech communication systems using complex test signals".

[19] ITU-T Recommendation P.862: "Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs".

[20] ISO 3 (1973): "Preferred numbers - Series of preferred numbers".

[21] ITU-T Recommendation P.800.1: "Mean Opinion Score (MOS) terminology" .

[22] ETSI TS 102 971: "Access and Terminals (AT); Public Switched Telephone Network (PSTN); Harmonized specification of physical and electrical characteristics of a 2-wire analogue interface for short line interface".

[23] ETSI ES 201 970: "Access and Terminals (AT); Public Switched Telephone Network (PSTN); Harmonized specification of physical and electrical characteristics at a 2-wire analogue presented Network Termination Point (NTP)".

[24] ITU-T Recommendation G.168: "Digital network echo cancellers".

[25] ITU-T Recommendation P.863: " Perceptual objective listening quality assessment '.

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

[27] ITU-T Recommendation G.722.1: "Low-complexity coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss".

[28] ITU-T Recommendation G.722.2: "Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)".

[29] ITU-T Recommendation P.862.1: "Mapping function for transforming P.862 raw result scores to MOS-LQO".

2.2 Informative references

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] ETSI EG 202 396-1: "Speech and multimedia Transmission Quality (STQ); Speech quality performance in the presence of background noise; Part 1: Background noise simulation technique and background noise database".

[i.2] ETSI EG 202 425: "Speech Processing, Transmission and Quality Aspects (STQ); Definition and implementation of VoIP reference point".

[i.3] ETSI EG 202 396-3: "Speech and multimedia Transmission Quality (STQ); Speech Quality performance in the presence of background noise Part 3: Background noise transmission - Objective test methods".

[i.4] IETF RFC 2833: "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[i.5] IETF RFC 4733: "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

[i.6] ETSI TR 102 927: "Speech and multimedia Transmission Quality (STQ); Packet Loss Concealment (PLC) performance measurement setup for home networks".

3 Definitions and abbreviations

3.1 Definitions

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

0dBr point: reference point always located at the digital side of the gateway, for IP-IP gateways located at the input of the MGW under test

NOTE: See ITU-T Recommendation G.100.1 [6].

2-wire interface: in the context of the present document, the telephony analogue interface over 2-wires used in the local loop

4-wire interface: in the context of the present document, a 4-wire digital interface with separate channels for both directions, irrespective of the physical transmission technology

codec: combination of an analogue-to-digital encoder and a digital-to-analogue decoder operating in opposite directions of transmission in the same equipment

Composite Source Signal (CSS): signal composed in time by various signal elements

MGW with 2-wire interface: MGW with an analogue 2-wire interface (ATA)

MGW with 4-wire interface: MGW with only 4-wire interfaces, e.g. ITA, IP-to-IP and wireless access points

nominal setting of the volume control: when a receive volume control is provided, the setting which is closest to the nominal RLR of 2 dB

receive direction: the direction from packet switched interfaces towards a synchronous interface (e.g. ISDN, analogue) or between two packet switched interfaces (for media gateways with packet switched transport on only one side)

NOTE: For media gateways with packet switched transport on both sides (IP-to-IP-MGW), the requirements of the receive direction have to be applied in both directions.

receive interface: interface in the measurement setup, where a receive signal is injected and/or a send signal is measured.

send direction: direction from a synchronous interface (e.g. ISDN, analogue) towards a packet switched interface (for media gateways with packet switched interface on only one side)

NOTE: For media gateways with packet switched interfaces on both sides the requirements of the send direction are not relevant.

send interface: interface in the measurement setup, where a send signal is injected and/or a receive signal is measured

3.2 Abbreviations

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

ATA	Analogue Terminal Adapter
CLR	Circuit Loudness Rating
CSS	Composite Source Signal
EL	Echo Loss
IAD	Integrated Access device
ITA	ISDN Terminal Adapter
JLR	Junction Loudness Rating
MGW	Media GateWay
MOS-LQOy	Mean Opinion Score - Listening Quality Objective

NOTE: See ITU-T Recommendation P.800.1 [21].

NLP	Non Linear Processor
PCM	Pulse Code Modulation
PESQ™	Perceptional Evaluation of Speech Quality™
PLC	Packet Loss Concealment
PN	Pseudo-random Noise
POI	Point Of Interconnect
PSTN	Public Switched Telephone Network
QoS	Quality of Service
TCN	Trace Control for Netem™
VAD	Voice Activity Detection

4 General considerations

4.1 Default Coding Algorithm

Narrowband VoIP gateways shall support the coding algorithm according to ITU-T Recommendation G.711 [9] (both µ-law and A-law). VoIP gateways may support other coding algorithms.

Wideband VoIP gateways shall support the coding algorithm according to ITU-T Recommendation G.722 [26]. VoIP gateways may support other coding algorithms.

NOTE: Associated Packet Loss Concealment (PLC) e.g. as defined in ITU-T Recommendation G.711 [9] appendix I should be used.

4.2 End-to-end considerations

In order to achieve a desired end-to-end speech transmission performance (mouth-to-ear) it is recommended that the general rules of transmission planning are carried out with the E-model of ITU-T Recommendation G.107 [3]; this includes the a-priori determination of the desired category of speech transmission quality as defined in ITU-T Recommendation G.109 [5].

While, in general, the transmission characteristics of single circuit-oriented network elements, such as switches or terminals can be assumed to have a single input value for the planning tasks of ITU-T Recommendation G.108 [4], this approach is not applicable in packet based systems and thus there is a need for the transmission planner's specific attention.

In particular the decision as to which delay measured according to the present document is acceptable or representative for the specific configuration is the responsibility of the individual transmission planner.

ITU-T Recommendation G.108 with its amendments [4] provides further guidance on this important issue.

The following optimum parameters from a users' perspective need to be considered:

- Minimized delay in send and receive direction.
- Optimum loudness Rating (JLR).
- Compensation for network delay variation.
- Packet loss recovery performance.
- Maximized echo loss.
- Immunity to false detection of DTMF in speech signal.

4.3 Parameters to be investigated

4.3.1 Applicability of parameters to different MGWs

Table 1: Parameter applicability

	2-wire home and network MGW	4-wire MGW (excl. IP-to-IP MGW)	4-wire MGW (IP-to-IP-MGW)	wireless home MGW
6.2 Codec independent parameters				
6.2.1 Send frequency response	M	M	NA	M
6.2.2 Circuit Loudness Rating in Send	M	M	NA	M
6.2.3 Linearity Range for CLR(SND)	M	M	NA	M
6.2.4 Send Distortion	M	M	NA	M
6.2.5 Spurious Out-of-Band Signals in Send direction	M	NA	NA	NA
6.2.6 Send Noise	M	M	NA	M
6.2.7 Receive frequency response	M	M	MM	M
6.2.8 Circuit Loudness Rating in Receive	M	M	MM	M
6.2.9 Linearity Range for CLR(RCV)	M	M	MM	M
6.2.10 Receive Distortion	M	M	MM	M
6.2.11 Out-of-Band Signals in Wideband to Narrowband Transcoding	NA	M	M	M
6.2.12 Spurious Out-of-band Signals Narrowband to Wideband Transcoding	NA	M	M	M
6.2.13 Minimum activation level and sensitivity in Receive direction	FFS	FFS	FFS	FFS
6.2.14 Receive Noise	M	M	MM	M
6.2.15 Double Talk Performance				
6.2.15.1 Attenuation Range in Send Direction during Double Talk	M	M	M	M
6.2.15.2 Attenuation Range in Receive Direction during Double Talk	M	M	M	M
6.2.15.3 Detection of Echo Components during Double Talk	M	M	M	M
6.2.15.4 Minimum activation level and sensitivity of double talk detection	FFS	FFS	FFS	FFS
6.2.16 Switching characteristics				
6.2.16.1 Activation in Send Direction	M	M	NA	M
6.2.16.2 Activation in Receive Direction	M	M	M	M

	2-wire home and network MGW	4-wire MGW (excl. IP-to-IP MGW)	4-wire MGW (IP-to-IP-MGW)	wireless home MGW
6.2.16.3 Silence Suppression and Comfort Noise Generation	FFS	FFS	FFS	FFS
6.2.17 Background Noise Performance				
6.2.17.1 Performance in send direction in the presence of background noise	M	M	MM	M
6.2.17.2 Quality of Speech with Background Noise	M	M	MM	M
6.2.17.3 Quality of Background Noise Transmission (with Far End Speech)	M	M	MM	M
6.2.17.4 Quality of Background Noise Transmission (with Near End Speech)	M	M	MM	M
6.2.18 Quality of echo cancellation				
6.2.18.2 Echo Performance acc. To G.168	M	M	NA	M
6.2.18.3 TCLw	M	M	NA	M
6.2.18.4Temporal echo effects	M	M	NA	M
6.2.18.5 Spectral Echo Attenuation	M	M	NA	M
6.2.18.6 Occurrence of Artefacts	FFS	FFS	NA	FFS
6.2.19 Variant Impairments; Network dependant				
6.2.19.1 Clock accuracy send	M	M	MM	M
6.2.19.2 Clock accuracy receive	M	M	MM	M
6.2.19.3 Send delay variation	M	M	MM	M
6.2.20 Immunity to DTMF false detection in send direction	M	M	MM	M
6.3 Codec Specific Requirements				
6.3.1 Send Delay	M	M	NA	M
6.3.2 Receive Delay	M	M	NA	M
6.3.3 Delay for IP-to-IP MGW	NA	NA	MM	NA
6.3.4 Objective Listening Speech Quality MOS-LQO in Send direction	M	M	M	M
6.3.5 Objective Listening Speech Quality MOS-LQO in Receive direction	M	M	M	M
6.3.5.1 Efficiency of Packet Loss Concealment (PLC)	FFS	FFS	FFS	FFS
6.3.5.2 Efficiency of Delay Variation Removal	FFS	FFS	FFS	FFS
M: Mandatory MM: Mandatory for both interfaces of the MGW NA: Not applicable FFS: For further study				

5 Test equipment

5.1 IP half channel measurement adaptor

The IP half channel measurement adaptor is described in EG 202 425 [i.2]. Such an apparatus is required to code and insert audio signals into IP packets send to the IP receive interface of the gateway under test, as well as to capture and decode audio signals constituting the payload of IP packets received from the IP sending interface of the gateway under test.

5.2 Environmental conditions for tests

The following conditions shall apply for the testing environment:

a) Ambient temperature: 15 °C to 35 °C (inclusive);

b) Relative humidity: 5 % to 85 %;

c) Air pressure: 86 kPa to 106 kPa (860 mbar to 1 060 mbar).

5.3 Accuracy of measurements and test signal generation

Unless specified otherwise, the accuracy of measurements made by test equipment shall be equal to or better than:

Table 2: Measurement Accuracy

Item	Accuracy
Electrical signal level	±0,2 dB for levels ≥ -50 dBV ±0,4 dB for levels < -50 dBV
Frequency	±0,2 %
Time	±0,2 %

Unless specified otherwise, the accuracy of the signals generated by the test equipment shall be better than:

Table 3: Accuracy of test signal generation

Quantity	Accuracy
Electrical excitation levels	±0,4 dB across the whole frequency range.
Frequency generation	±2 % (see note)
Time	±0,2 %
Specified component values	±1 %
NOTE: This tolerance may be used to avoid measurements at critical frequencies, e.g. those due to sampling operations within the terminal under test.	

If the equipment is powered by other means and those means are not supplied as part of the apparatus, all tests shall be carried out within the power supply limit declared by the supplier. If the power supply is a.c. the test shall be conducted within ±4 % of the rated frequency.

5.4 Network impairment simulation

At least one set of requirements is based on the assumption of an error free packet network, and at least one other set of requirements is based on a defined simulated malperformance of the packet network.

An appropriate network simulator has to be used, for example Netem™.

The key points of Netem™ can be summarized as follows:

- Netem™ is part of most Linux™ distributions, it only has to be switched on, when compiling a kernel. With Netem™, there are the same possibilities as with Nistnet™, there can be generated loss, duplication, delay and jitter (and the distribution can be chosen during runtime). Netem™ can be run on a Linux-PC™ running as a bridge or a router (Nistnet™ only runs on routers).

- With an amendment of Netem™, TCN (Trace Control for Netem™) which was developed by ETH Zurich™, it is even possible, to control the behaviour of single packets via a trace file. So it is for example possible to generate a single packet loss, or a specific delay pattern. This amendment is planned to be included in new Linux Kernels™, nowadays it is available as a patch to a specific kernel and to the iproute2 tool (iproute2 contains Netem™).

- It is not advised to define specific distortion patterns for testing in standards, because it will be easy to adapt devices to these patterns (as it is already done for test signals). But if a pattern is unknown to a manufacturer, the same pattern can be used by a test lab for different devices and gives comparable results. It is also possible to take a trace of Nistnet™ distortions, generate a file out of this and playback the exact same distortions with Netem™.

6 Requirements and associated Measurement Methodologies

Differences between different media gateway types are dealt with in the respective requirements.

In the case of IP-IP MGW packet based interfaces are provided at both sides of the gateway. Therefore the receive requirements apply, for both interfaces.

NOTE 1: In general the test methods as described in the present document apply. If alternative methods exist they may be used if they have been proven to give the same result as the method described in the standard.

NOTE 2: Due to the time variant nature of IP connections delay variation may impair the measurements. In such cases the measurement has to be repeated until a valid measurement result is achieved.

6.1 Test setup

The preferred way of testing a gateway is to connect its interfaces to network simulators with exact defined settings and access points. The test sequences are fed in electrically, using a reference codec or using the direct signal processing approach.

When VoIP runs on the gateway under test only in conjunction with a registration by an application server (e.g. SIP proxy), the network simulator may need to provide also the registration functionality.

Alternatively, if for the IP-interfaces another technology than Ethernet is used (for instance DSL access, it may be necessary to add additional equipment in the test setup for connecting the measurement equipment (e.g. a DSLAM, if the IP-interface works over DSL). There should be no speech signal processing in this additional equipment (the media payload has to be passed transparent through this equipment, while e.g. header manipulation is allowed). The influence of this additional equipment (delay and eventually delay variation) has to be taken in account for the measurements.

NOTE 1: It is up to the testlab to identify potential time invariances or non linearities in the network used for interconnection and to take those effects into account properly.

With this setup it is possible to measure the parameters listed in the present document over a whole network, if the behaviour of the network is known.

In the present document, the terms "send" and "receive" can be found in the pictures of the relevant test setup.

When a coder with variable bit rate is used for testing the MGW parameters, the bit rate recognized giving the best characteristics and/or the ones commonly used should be selected, e.g.:

- AMR-NB (TS 126 171 [2]): 12,2 kbit/s.
- AMR-WB (G. 722.2 [28]): 12,65 kbit/s.
- ITU-T Recommendation G.729.1 [13]: 32 kbit/s.

NOTE 2: Although packet capturing and network simulation in figures 1 to 4 are shown in one box they may be separate devices.

6.1.1 Setup for Media Gateways with 4-wire interface

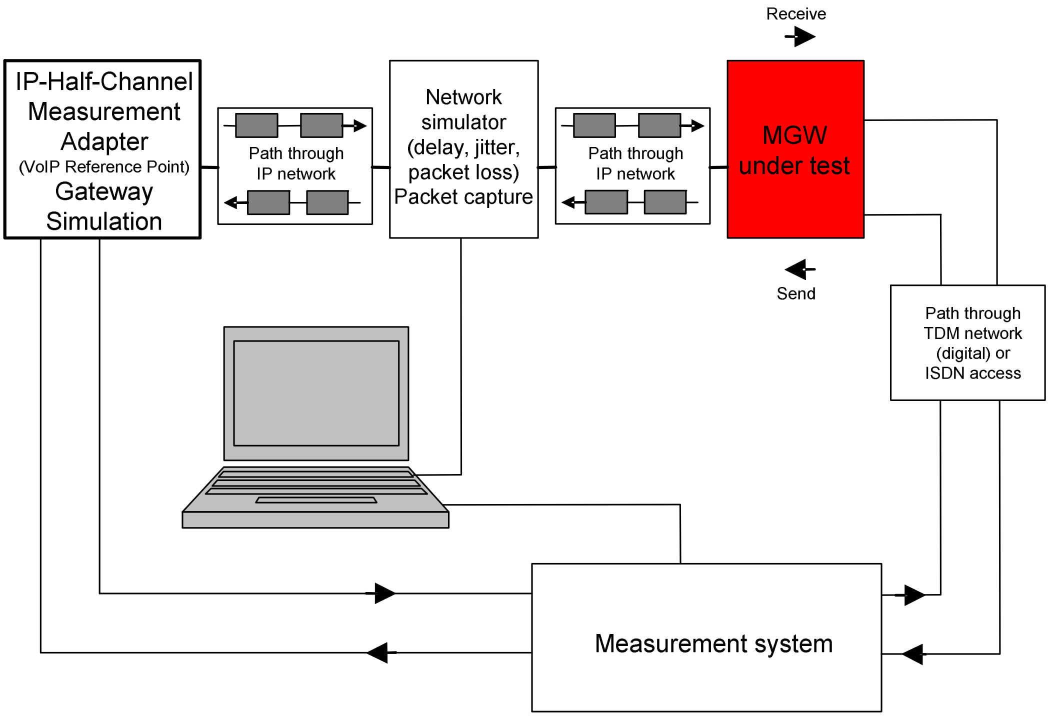

Figure 1: Half channel measurement for MGW with 4-wire interface

6.1.2 Setup for Media Gateways with 2-wire interface

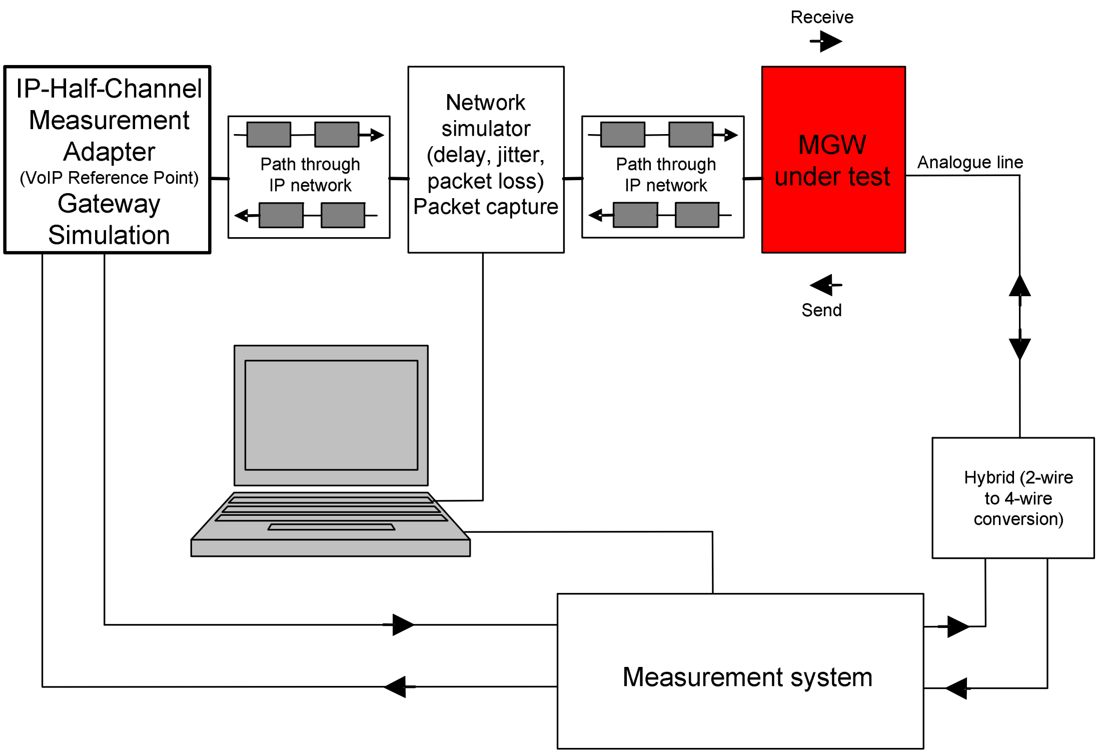

Figure 2: Half channel measurement for MGW with 2-wire interface

6.1.3 Setup for Media Gateways with Wireless Access

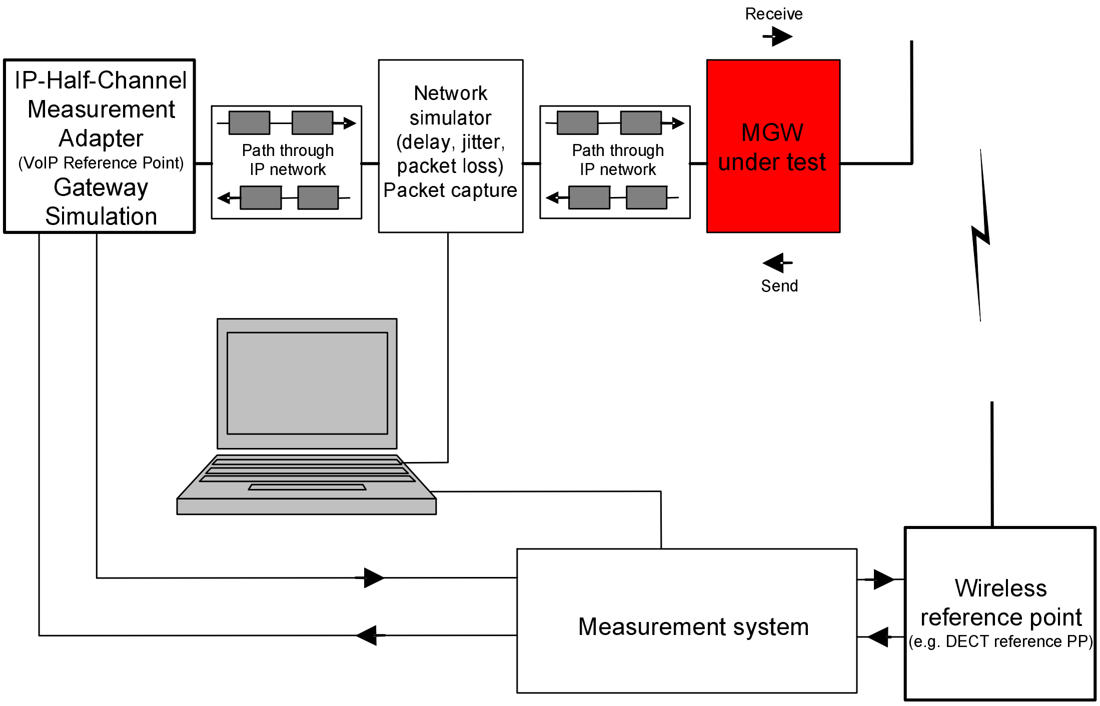

Figure 3: Half channel measurement for MGW with wireless access

6.1.4 Setup for IP-to-IP Media Gateways

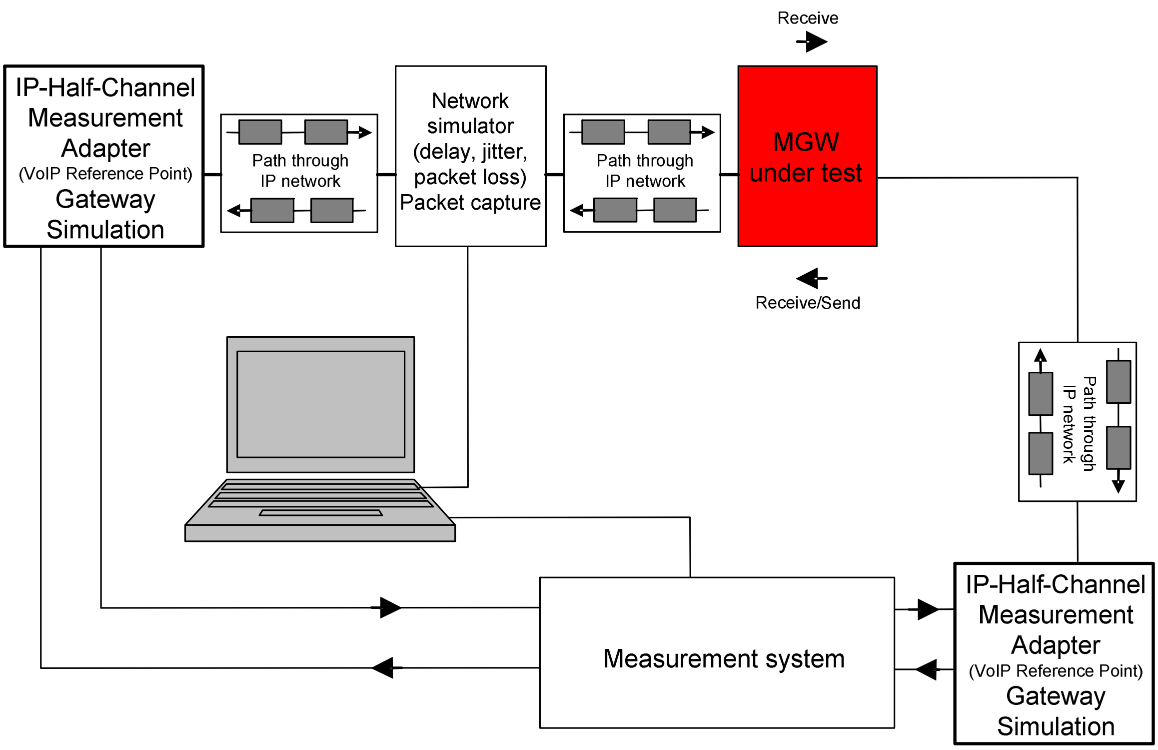

Figure 4: Half channel measurement for IP-to-IP MGW

NOTE 1: For measuring both directions of an IP-to-IP MGW, the network simulator can be moved to the other side of the MGW.

NOTE 2: If the network simulator is moved to the other side of the IP-to-IP MGW the measurements to be conducted in receive are to be conducted in this scenario on the other channel (send in the precious scenario).

6.1.5 Test Signal Levels

Unless specified otherwise, the applied test signal level at the digital inputs shall be -16 dBm0. For analogue inputs (2-wire) the applied test signal level should be -16 dBm for home MGWs and -19 dBm for network MGWs at the media gateway interface.

NOTE: For analogue inputs of network MGWs, an attenuation of 3 dB for the simulation of an access line is taken into account for measurements with nominal levels.

6.1.6 Background noise simulation

Background noise signals used for testing should be recorded according to the description in EG 202 396-1 [i.1].

EG 202 396-1 [i.1] contains a description of the recording arrangement for realistic background noises, a description of the setup for a loudspeaker arrangement suitable to simulate a background noise field in a lab-type environment and a database of realistic background noises, which can be used for testing the terminal performance with a variety of different background noises.

In order to create a representative electrical test signal for the MGW containing speech at a nominal level mixed with the amount of background noise picked up by a terminal, the setup in EG 202 396-1 [i.1] is used. The terminal is connected to a reference interface providing nominal properties for the electrical interface as used by the terminal. The signal (speech plus noise) is recorded at this interface and inserted through the appropriate reference interfaces as described in clauses 6.1.1 to 6.1.4 in such a way that the signal level and spectral content delivered to the MGW under test is equivalent to the one it would have seen if the terminal was connected directly. Either terminals considered to be representative for the type of terminal attached to the MGW are used or individual terminals are used.

NOTE: Due to terminal geometry, microphone technique and signal processing used in the phone the signals acquired at the electrical interface may highly vary. As a consequence the performance of the MGW may highly depend on the terminal connected.

The following noises of EG 202 396-1 [i.1] shall be used.

Recording in pub	Pub_Noise_binaural	30 s	L: 77,8 dB(A) R: 78,9 dB(A)	Binaural
Recording at sales counter	Cafeteria_Noise_binaural	30 s	L: 68,4 dB(A) R: 67,3 dB(A)	Binaural
Recording in business office	Work_Noise_Office_Callcener_binaural	30 s	L: 56,6 dB(A) R: 57,8 dB(A)	Binaural

6.2 Coding independent parameters

6.2.1 Send Frequency response

Requirement

The frequency response for 4-wire MGW shall be according to tables 4 and 5:

Table 4: Send frequency response for 4-wire MGW

Frequency	Upper Limit	Lower Limit
100 Hz	1 dB	
300 Hz	1 dB	-1 dB
3 400 Hz	1 dB	-1 dB
4 000 Hz	1 dB	

Table 5: Send frequency response for wideband 4-wire MGW

Frequency	Upper Limit	Lower Limit
100 Hz	1 dB	
200 Hz	1 dB	1 dB
7 000 Hz	1 dB	1 dB
8 000 Hz	1 dB	

NOTE: The frequency response characteristics requirements apply to codecs having flat response characteristics. If a codec with non-flat characteristics is used the requirement has to be corrected by the ideal response characteristics of this codec.

The frequency response for 2-wire MGW shall be according to table 6 (see [22]) for both home and network MGWs:

Table 6: Send frequency response for narrowband 2-wire MGW

Frequency	Upper Limit	Lower Limit
0 Hz	1 dB	
300 Hz	1 dB	-1 dB
3 400 Hz	1 dB	-1 dB
4 000 Hz	1 dB	

Measurement Method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [15]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501 [17] shall be used. The test signal duration shall be 20 s (10 s female, 10 s male voice). The test signal level is averaged over the complete test signal sequence.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [20] for frequencies from 100 Hz to 4 kHz (8 kHz for wideband 4-wire MGWs) inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dB.

6.2.2 Circuit Loudness Rating in Send

Requirement

The nominal value of Circuit Loudness Rating in Send (CLR) for MGWs with 4-wire interface shall be:

- CLR(SND) = 0 dB ± 1 dB

The nominal value of Circuit Loudness Rating in Send (CLR) for MGWs with 2-wire interface shall be (in accordance with [22] and [23]):

- CLR(SND) = 3 dB ± 1 dB (Home MGW)
- CLR(SND) = 0 dB ± 1 dB (Network MGW)

Measurement Method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [15]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501 [17] shall be used. The test signal duration shall be 20 s (10 s female, 10 s male voice). The test signal level is averaged over the complete test signal sequence.

For narrowband MGWs, the send sensitivity shall be calculated as average loss in the frequency range from 300 Hz to 3 400 Hz (ITU-T Recommendation G.111, Annex A [7]).

For wideband MGWs, the send sensitivity shall be calculated as average loss in the frequency range from 100 Hz to 7 000 Hz.

The sensitivity is expressed in terms of dB.

6.2.3 Linearity Range for CLR (SND)

Requirement

For MGW with 4-wire interface, the send sensitivity determined with input levels between -36 dBm0 and -11 dBm0 shall not differ by more than ±0,5 dB from the send sensitivity determined with an input level of -16 dBm0. For the input level of -6 dBm0 a limit of +2/-1 dB applies.

Table 7: Linearity range for CLR (SND) for MGW with 4-wire interface

Linearity range of CLR: ΔCLR = CLR - CLR@-16 dBm0			
Input Level	**Target ΔCLR**	**Upper limit**	**Lower limit**
-36 dBm0	0	0,5 dB	-0,5 dB
-31 dBm0	0	0,5 dB	-0,5 dB
-26 dBm0	0	0,5 dB	-0,5 dB
-21 dBm0	0	0,5 dB	-0,5 dB
-16 dBm0	0	0 dB	0 dB
-11 dBm0	0	0,5 dB	-0,5 dB
-6 dBm0	0	2 dB	-1 dB

For Home-MGW with 2-wire interface, the send sensitivity determined with input levels between -33 dBm and -8 dBm shall not differ by more than ±0,5 dB from the send sensitivity determined with an input level of -16 dBm. For the input level of -3 dBm a limit of +2/-1 dB applies.

Table 8: Linearity range for CLR (SND) for Home-MGW with 2-wire interface

Linearity range of CLR: ΔCLR = CLR - CLR@-16 dBm0			
Input Level	**Target ΔCLR**	**Upper limit**	**Lower limit**
-33 dBm	0	0,5 dB	-0,5 dB
-28 dBm	0	0,5 dB	-0,5 dB
-23 dBm	0	0,5 dB	-0,5 dB
-18 dBm	0	0,5 dB	-0,5 dB
-13 dBm	0	0 dB	0 dB
-8 dBm	0	0,5 dB	-0,5 dB
-3 dBm	0	2 dB	-1 dB

For Network-MGW with 2-wire interface, the send sensitivity determined with input levels between -40 dBm and -6 dBm shall not differ by more than ±0,5 dB from the send sensitivity determined with an input level of -20 dBm. For the input level of -6 dBm a limit of +2/-1 dB applies.

Table 9: Linearity range for CLR (SND) for Network-MGW with 2-wire interface

Linearity range of CLR: ΔCLR = CLR - CLR@-16 dBm0			
Input Level	**Target ΔCLR**	**Upper limit**	**Lower limit**
-40 dBm	0	0,5 dB	-0,5 dB
-35 dBm	0	0,5 dB	-0,5 dB
-30 dBm	0	0,5 dB	-0,5 dB
-25 dBm	0	0,5 dB	-0,5 dB
-20 dBm	0	0,5 dB	-0,5 dB
-15 dBm	0	0 dB	0 dB
-10 dBm	0	0,5 dB	-0,5 dB
-6 dBm	0	2 dB	-1 dB

NOTE 1: It is assumed that the variation of gain is mostly codec independent. In case codec specific requirements are needed this is found in the codec specific section.

NOTE 2: The broader tolerance range at the highest input level takes into account, that at this level clipping will occur.

Measurement Method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [15]. If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [17] shall be used. The test signal level is averaged over the complete test signal sequence.

For narrowband MGWs, the send sensitivity shall be calculated as average loss in the frequency range from 300 Hz to 3 400 Hz (ITU-T Recommendation G.111, Annex A [7]).

For wideband MGWs, the send sensitivity shall be calculated as average loss in the frequency range from 100 Hz to 7 000 Hz.

6.2.4 Send Distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask:

Table 10: Send distortion for narrowband MGW

Frequency	Signal to distortion ratio limit, receive
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Table 11: Send distortion for wideband MGW

Frequency	Signal to distortion ratio limit, receive
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Measurement Method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz for narrowband, additionally 2 000 Hz for wideband.

An artificial voice according to ITU-Recommendation P.50 [15] or a speech like test signal as described in ITU-T Recommendation P.501 [17] can be used for activation.

Measurement are made at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz.

The signal to harmonic distortion ratio is measured selectively up to 3,15 kHz for narrowband MGW, up to 6,3 kHz for wideband MGW.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

6.2.5 Spurious Out-of-Band Signals in Send direction

Requirement

For MGW with 2-wire interface:

- The level of any image frequency produced in the time slot corresponding to the input connection should be at least 25 dB below the level of the test signal.

Measurement Method

A sine-wave signal in the range from 4,6 kHz to 72 kHz applied to the 2-wire interface of an input connection at a level of −25 dBm0. The level of any image frequency produced in the time slot corresponding to the input connection is measured and referred to the level of the test signal.

6.2.6 Send Noise

Requirement

For MGW with 4-wire interface:

- The maximum noise level produced by the MGW under silent conditions in the send direction shall not exceed -64 dBm0p for narrowband MGWs and -68 dBm0 (A) for wideband MGWs.
- No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

For MGW with 2-wire interface:

- The maximum noise level produced by the MGW under silent conditions in the send direction shall not exceed -67 dBm0p, see [22].
- No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement Method

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals (CSS) as described in ITU-T Recommendation P.501 [17]. The activation signal level shall be -16 dBm0. The activation signal level is averaged over the complete activation signal sequence. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation.

The send noise is measured at the receive interface in the frequency range from 100 Hz to 4 kHz for narrowband MGWs, from 100 Hz to 8 kHz for wideband MGWs. The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 second. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The noise level is measured in dBm0p for narrowband MGW, in dBm0 (A) for wideband MGW.

6.2.7 Receive Frequency Response

The frequency response for 4-wire MGW shall be according to tables 9 and 10:

Table 12: Receive frequency response for 4-wire MGW

Frequency	Upper Limit	Lower Limit
100 Hz	1 dB	
300 Hz	1 dB	-1 dB
3 400 Hz	1 dB	-1 dB
4 000 Hz	1 dB	

Table 13: Receive frequency response for wideband 4-wire MGW

Frequency	Upper Limit	Lower Limit
100 Hz	1 dB	
200 Hz	1 dB	-1 dB
7 000 Hz	1 dB	-1 dB
8 000 Hz	1 dB	

The frequency response for 2-wire MGW shall be according to table 14 (see [22]) 3 for both home and network MGWs:

Table 14: Receive frequency response for narrowband 2-wire MGW

Frequency	Upper Limit	Lower Limit
0 Hz	**1 dB**	
300 Hz	1 dB	-1 dB
3 400 Hz	1 dB	-1 dB
4 000 Hz	1 dB	

The frequency response for 2-wire MGW shall be according to [22] for both home and network MGWs.

Measurement Method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [15]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501 [17] shall be used. The test signal duration shall be 20 s (10 s female, 10 s male voice). The test signal level is averaged over the complete test signal sequence.

Measurements shall be made at one twelfth-octave intervals as given by the R.40 series of preferred numbers in ISO 3 [20] for frequencies from 100 Hz to 4 kHz (8 kHz for wideband 4-wire MGWs) inclusive. For the calculation the averaged measured level at each frequency band is referred to the averaged test signal level measured in each frequency band.

The sensitivity is expressed in terms of dB.

6.2.8 Circuit Loudness Rating in Receive

The nominal value of Circuit Loudness Rating in Receive (CLR) for MGWs with 4-wire interface shall be:

- CLR(RCV) = 0 dB ± 1 dB

The nominal value of Circuit Loudness Rating in Receive (CLR) for MGWs with 2-wire interface shall be (in line with [22] and [23]):

- CLR(RCV) = 10 dB ± 1 dB (Home MGW)
- CLR(RCV) = 7 dB ± 1 dB (Network MGW)

NOTE: CLR (RCV) for MGW with 2-wire interface can differ from the above recommended value due to national transmission plans.

Measurement Method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [15]. If the signal to noise ratio in the high frequency domain is not sufficient Composite Source Signal (CSS) as defined in ITU-T Recommendation P.501 [17] shall be used. The test signal duration shall be 20 s (10 s female, 10 s male voice). The test signal level is averaged over the complete test signal sequence.

For narrowband MGWs, the receive sensitivity shall be calculated as average loss in the frequency range from 300 Hz to 3400 Hz (ITU-T Recommendation G.111, Annex A [7]).

For wideband MGWs, the receive sensitivity shall be calculated as average loss in the frequency range from 100 Hz to 7 000 Hz.

The sensitivity is expressed in terms of dB.

6.2.9 Linearity Range for CLR (RCV)

Requirement

For MGW with 4-wire interface, the receive sensitivity determined with input levels between -36 dBm0 and -11 dBm0 shall not differ by more than ±0,5 dB from the receive sensitivity determined with an input level of -16 dBm0. For the input sound pressure level of -6 dBm0 a limit of +2/-1 dB applies.

Table 15: Linearity range for CLR (RCV) for Home-MGW with 4-wire interface

Linearity range of CLR: ΔCLR = CLR - CLR@-16 dBm0			
Input Level	**Target ΔCLR**	**Upper limit**	**Lower limit**
-36 dBm0	0	0,5 dB	-0,5 dB
-31 dBm0	0	0,5 dB	-0,5 dB
-26 dBm0	0	0,5 dB	-0,5 dB
-21 dBm0	0	0,5 dB	-0,5 dB
-16 dBm0	0	0 dB	0 dB
-11 dBm0	0	0,5 dB	-0,5 dB
-6 dBm0	0	2 dB	-1 dB

For Home-MGW with 2-wire interface, the send sensitivity determined with input levels between -36 dBm and -16 dBm0 shall not differ by more than ±0,5 dB from the send sensitivity determined with an input level of -16 dBm0. For the input level of -3 dBm a limit of +2/-1 dB applies.

Table 16: Linearity range for CLR (RCV) for Home-MGW with 2-wire interface

Linearity range of CLR: ΔCLR = CLR - CLR@-16 dBm0			
Input Level	**Target ΔCLR**	**Upper limit**	**Lower limit**
-36 dBm	0	0,5 dB	-0,5 dB
-31 dBm	0	0,5 dB	-0,5 dB
-26 dBm	0	0,5 dB	-0,5 dB
-21 dBm	0	0,5 dB	-0,5 dB
-16 dBm	0	0 dB	0 dB
-11 dBm	0	0,5 dB	-0,5 dB
-6 dBm	0	2 dB	-1 dB

For Network-MGW with 2-wire interface, the send sensitivity determined with input levels between -40 dBm and -6 dBm shall not differ by more than ±0,5 dB from the send sensitivity determined with an input level of -20 dBm. For the input level of -6 dBm a limit of +2/-1 dB applies.

Table 17: Linearity range for CLR (RCV) for Network-MGW with 2-wire interface

Linearity range of CLR: ΔCLR = CLR - CLR@-16 dBm0			
Input Level	**Target ΔCLR**	**Upper limit**	**Lower limit**
-40 dBm	0	0,5 dB	-0,5 dB
-35 dBm	0	0,5 dB	-0,5 dB
-30 dBm	0	0,5 dB	-0,5 dB
-25 dBm	0	0,5 dB	-0,5 dB
-20 dBm	0	0,5 dB	-0,5 dB
-15 dBm	0	0 dB	0 dB
-10 dBm	0	0,5 dB	-0,5 dB
-6 dBm	0	2 dB	-1 dB

NOTE 1: It is assumed that the variation of gain is mostly codec independent. In case codec specific requirements are needed this is found in the codec specific section.

NOTE 2: The broader tolerance range at the highest input level takes into account, that at this level clipping will occur.

Measurement Method

The test signal to be used for the measurements shall be the artificial voice according to ITU-T Recommendation P.50 [15]. If the signal to noise ratio in the high frequency domain is not sufficient CSS as defined in ITU-T Recommendation P.501 [17] shall be used. The test signal level is averaged over the complete test signal sequence.

For narrowband MGWs, the receive sensitivity shall be calculated as average loss in the frequency range from 300 Hz to 3 400 Hz (ITU-T Recommendation G.111, Annex A [7]).

For wideband MGWs, the receive sensitivity shall be calculated as average loss in the frequency range from 100 Hz to 7 000 Hz.

6.2.10 Receive Distortion

Requirement

The ratio of signal to harmonic distortion shall be above the following mask:

Table 18: Receive distortion for narrowband MGW

Frequency	Signal to distortion ratio limit, receive
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Table 19: Receive distortion for wideband MGW

Frequency	Signal to distortion ratio limit, receive
315 Hz	26 dB
400 Hz	30 dB
500 Hz	30 dB
800 Hz	30 dB
1 kHz	30 dB
2 kHz	30 dB
NOTE: Limits at intermediate frequencies lie on a straight line drawn between the given values on a linear (dB ratio) - logarithmic (frequency) scale.	

Measurement Method

The signal used is an activation signal followed by a sine-wave signal with a frequency at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz and 1 000 Hz for narrowband, additionally 2 000 Hz for wideband.

An artificial voice according to ITU-Recommendation P.50 [15] or a speech like test signal as described in ITU-T Recommendation P.501 [17] can be used for activation.

Measurement are made at 315 Hz, 400 Hz, 500 Hz, 630 Hz, 800 Hz, 1 000 Hz and 2 000 Hz.

The signal to harmonic distortion ratio is measured selectively up to 3,15 kHz for narrowband MGW, up to 6,3 kHz for wideband MGW.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

6.2.11 Out-of-Band Signals Wideband to Narrowband Transcoding

For MGW with 4-wire interface, this parameter has to be measured only, if the MGW supports a mode, where one side is wideband and the other side narrowband (transcoding wideband to narrowband). This measurement is only applicable for transcoding from wideband to narrowband.

Requirement

With any signal above 4,6 kHz and up to 8 kHz applied with a level of -16 dBm0, the level of any image frequency shall be below the level obtained for the reference signal by at least the amount (in dB) specified in table 20.

Table 20: Out-of-band signal limit, receive

Frequency	Minimum attenuation
4,6 kHz	30 dB
8 kHz	40 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

Measurement Method

For a correct activation of the system, an artificial voice according to ITU-Recommendation P.50 [15] or a speech like test signal as described in ITU-T Recommendation P.501 [17] shall be used for activation. Level of this activation signal shall be -16 dBm0.

For the test, an out-of-band signal shall be provided as a frequency band signal centred on 4,65 kHz, 5 kHz, 6 kHz, 6,5 kHz, 7 kHz and 7,5 kHz respectively. The level of any image frequencies at the digital interface shall be measured.

The levels of these signals shall be -16 dBm0.

The complete test signal is constituted by t1 ms of in-band signal (reference signal), t2 ms of out-of-band signal and another time t1 ms of in-band signal (reference signal).

The observation of the output signal on the first and second in-band signals permits the control if the set is correctly activated during the out-of-band measurement. This measurement shall be performed during t2 period.

A value of 250 ms is suggested for t1.

T2 depends on the integration time of the analyser, typically less than 150 ms.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

6.2.12 Spurious Out-of-band Signals Narrowband to Wideband Transcoding

For MGW with 4-wire interface, this parameter has to be measured only, if the MGW supports a mode, where one side is wideband and the other side narrowband (transcoding wideband to narrowband). This measurement is for transcoding from narrowband to wideband.

If the gateway provides wideband extension techniques this measurement is not applicable.

Requirement

Any spurious out-of-band image signals in the frequency range from 4,6 kHz to 8 kHz measured selectively shall be lower than the in-band level measured with a reference signal. The minimum level difference between the reference signal level and the out-of-band image signal level shall be as given in the table 21:

Table 21: Spurious out of band signal limits, receive

Frequency	Minimum attenuation
4,6 kHz	35 dB
8 kHz	45 dB
NOTE: The limits for intermediate frequencies lie on a straight line drawn between the given values on a linear (dB) - logarithmic (kHz) scale.	

Measurement Method

The test arrangement is according to clause 6.1.

The signal used is an activation signal followed by a sine-wave signal. For input signals at the frequencies 500 Hz, 1 000 Hz, 2 000 Hz and 3 150 Hz applied at the level of -16 dBm0, the level of spurious out-of-band image signals at frequencies up to 8 kHz is measured selectively at measurement point.

An artificial voice according to ITU-Recommendation P.50 [15] or a speech like test signal as described in ITU-T Recommendation P.501 [17] can be used for activation. The level of this activation signal is -16 dBm0.

NOTE: Depending on the type of codec the test signal used may need to be adapted.

6.2.13 Minimum Activation Level and Sensitivity in Receive Direction

For further study.

6.2.14 Receive Noise

Requirement

For MGW with 4-wire interface: The maximum noise level produced by the MGW under silent conditions in the receive direction shall not exceed -64 dBm0p for narrowband MGWs and -68 dBm0 (A) for wideband MGWs.

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

For MGW with 2-wire interface:

The maximum noise level produced by the MGW under silent conditions in the send direction shall not exceed -70 dBm0p, see [22].

No peaks in the frequency domain higher than 10 dB above the average noise spectrum shall occur.

Measurement Method

For the actual measurement no test signal is used. In order to reliably activate the terminal an activation signal is introduced before the actual measurement. The activation signal shall be a sequence of 4 composite source signals (CSS) as described in ITU-T Recommendation P.501 [17]. The activation signal level shall be -16 dBm0 for 4-wire MGW, -16 dBm for 2-wire home-GW and -19 dBm for 2-wire network MGW. The activation signal level is averaged over the complete activation signal sequence. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used for activation.

The receive noise is measured at the send interface in the frequency range from 100 Hz to 4 kHz for narrowband MGWs, from 100 Hz to 8 kHz for wideband MGWs. The analysis window is applied directly after stopping the activation signal but taking into account the influence of all acoustical components (reverberations). The averaging time is 1 second. The test house has to ensure (e.g. by monitoring the time signal) that during the test the terminal remains in activated condition. If the terminal is deactivated during the measurement, the measurement time has to be reduced to the period where the terminal remains in activated condition.

The noise level is measured in dBm0p for narrowband MGW, in dBm0 (A) for wideband MGW.

6.2.15 Double Talk Performance

During double talk the speech is mainly determined by 2 parameters: impairment caused by echo during double talk and level variation between single and double talk (attenuation range).

In order to guarantee sufficient quality under double talk conditions the Talker Echo Loudness Rating should be high and the attenuation inserted should be as low as possible. MGWs which do not allow double talk in any case should provide a good echo attenuation which is realized by a high attenuation range in this case.

The most important parameters determining the speech quality during double talk are (see ITU-T Recommendations P.340 [16] and P.502 [18]):

- Attenuation range in send direction during double talk $A_{H,S,dt}$.
- Attenuation range in receive direction during double talk $A_{H,R,dt}$.
- Echo attenuation during double talk.

The double talk performance may be highly influenced by the performance of the echo canceller, especially by the NLP implementation. The double talk performance should be checked by using the relevant echo paths as described in clause 6.2.18.1.

6.2.15.1 Attenuation Range in Send Direction during Double Talk $A_{H,S,dt}$

Requirement

Based on the level variation in send direction during double talk $A_{H,S,dt}$ the behaviour of the MGW can be classified according to table 22.

For media gateway Type 1 is required.

Table 22

Category (according to ITU-T Recommendation P.340 [16])	**1**	**2a**	**2b**	**2c**	**3**
	Full Duplex Capability	*Partial Duplex Capability*			*No Duplex Capability*
$A_{H,S,dt}$ ***[dB]***	*≤ 3*	*≤ 6*	*≤ 9*	*≤ 12*	*> 12*

In general table 22 provides a quality classification of MGWs regarding double talk performance. However, this does not mean that a MGW which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement Method

The test signal to determine the attenuation range during double talk is shown in figure 5. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction.

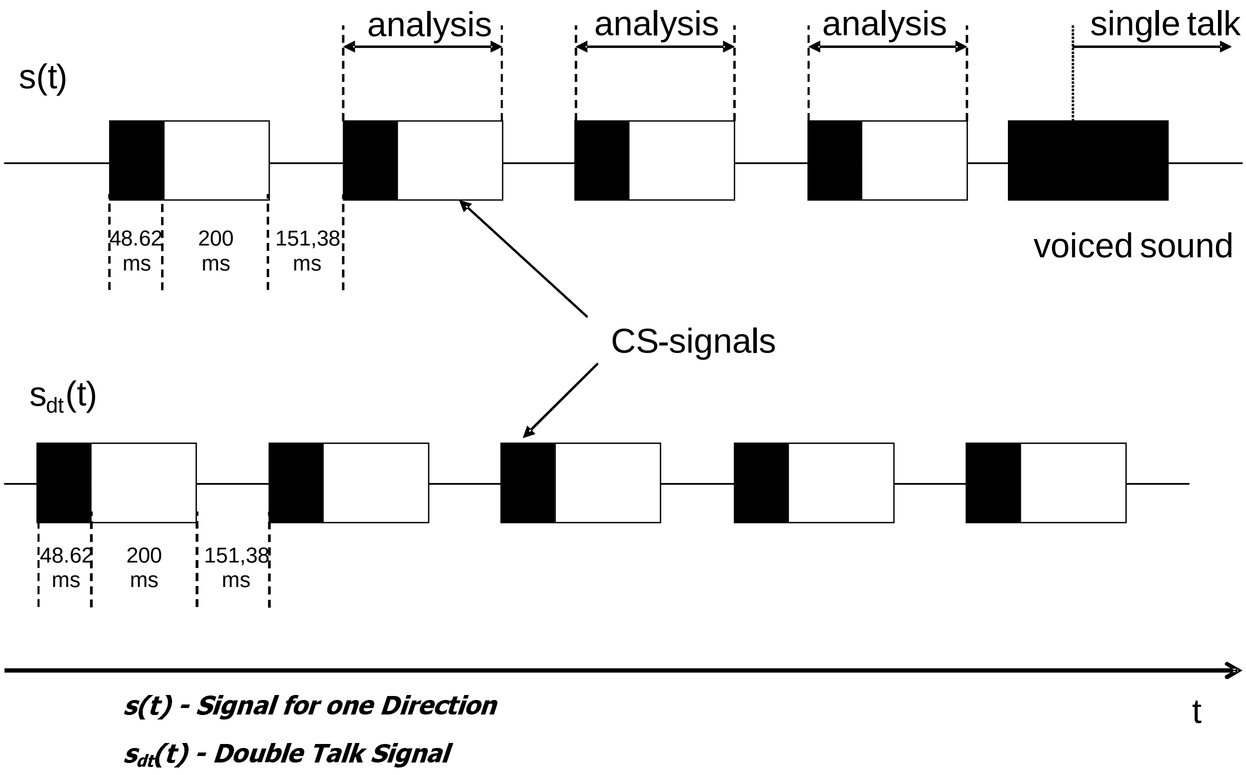

Figure 5: Double Talk Test Sequence with overlapping CS signals in send and receive direction

Figure 5 indicates that the sequences overlap partially. The beginning of the CS sequence (voiced sound, black) is overlapped by the end of the pn-sequence (white) of the opposite direction. During the active signal parts of one signal the analysis can be conducted in send and receive direction. The analysis times are shown in figure 6 as well. The test signals are synchronized in time at the send interface directly at the MGW (this may be different to the send interface used for measurement). The delay of the test arrangement should be constant during the measurement.

NOTE: The length of voiced sound of the double talk signal is achieved by repeating one period of the voiced sound for double talk according to ITU-T Recommendation P.501 [17] 10 times and cutting off the initial 3,3 ms of the period of the first voiced sound.

The settings for the test signals are as follows.

Table 23

	Receive Direction (sdt(t))	Send Direction (s(t)) 4-wire MGW	Send Direction (s(t)) 2-wire home MGW	Send Direction (s(t)) 2-wire network MGW
Pause Length between two Signal Bursts	151,38 ms	151,38 ms	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original Pause length of 101,38 ms)	-16 dBm0	-16 dBm0	-16 dBm	-19 dBm
Active Signal Parts	-14,7 dBm0	-14,7 dBm0	-14,7 dBm	-17,7 dBm

The test arrangement is according to clause 6.1.

When determining the attenuation range in send direction the signal measured at the receive interface is referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in send direction until its complete activation (during the pause in the receive channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

6.2.15.2 Attenuation Range in Receive Direction during Double Talk $A_{H,R,dt}$

Requirement

Based on the level variation in receive direction during double talk $A_{H,R,dt}$ the behaviour of the MGW can be classified according to table 24.

For media gateway Type 1 is required.

Table 24

Category (according to ITU-T Recommendation P.340 [16])	1	2a	2b	2c	3
	Full Duplex Capability	*Partial Duplex Capability*			*No Duplex Capability*
$A_{H,R,dt}$ [dB]	*≤ 3*	*≤ 5*	*≤ 8*	*≤ 10*	*> 10*

In general this table provides a quality classification of terminals regarding double talk performance. However, this does not mean that a terminal which is category 1 based on the double talk performance is of high quality concerning the overall quality as well.

Measurement Method

The test signal to determine the attenuation range during double talk is shown in figure 5. A sequence of uncorrelated CS signals is used which is inserted in parallel in send and receive direction. The test signals are synchronized in time at the send interface directly at the MGW (this may be different to the send interface used for measurement). The delay of the test arrangement should be constant during the measurement.

The settings for the test signals are as follows.

Table 25

	Receive Direction (s(t))	Send Direction (s(t)) 4-wire MGW	Send Direction (s(t)) 2-wire home GW	Send Direction (s(t)) 2-wire network MGW
Pause Length between two Signal Bursts	151,38 ms	151,38 ms	151,38 ms	151,38 ms
Average Signal Level (Assuming an Original pause Length of 101,38 ms)	-16 dBm0	-16 dBm0	-16 dBm	-19 dBm
Active Signal Parts	-14,7 dBm0	-14,7 dBm0	-14,7 dBm	-17,7 dBm

The test arrangement is according to clause 6.1.

When determining the attenuation range in receive direction the signal measured at the send interface referred to the test signal inserted.

The level is determined as level vs. time from the time domain. The integration time of the level analysis is 5 ms. The attenuation is determined from the level difference measured at the beginning of the double talk always with the beginning of the CS-signal in receive direction until its complete activation (during the pause in the sending channel). The analysis is performed over the complete signal starting with the second CS-signal. The first CS-signal is not used for the analysis.

6.2.15.3 Detection of Echo Components during Double Talk

Requirement

Echo Loss (EL) during double talk is the echo suppression provided by the MGW during double talk measured at the receive interface.

For media gateway Type 1 is required.

NOTE: The echo attenuation during double talk is based on the parameter Talker Echo Loudness Rating (TELRdt). It is assumed that the terminal at the opposite end of the connection provides nominal Loudness Rating (SLR + RLR = 10 dB).

Under these conditions the requirements given in table 26 are applicable (more information can be found in annex A of the ITU-T Recommendation P.340 [16]).

Table 26

Category (according to ITU-T Recommendation P.340 [16])	1	2a	2b	2c	3
	Full Duplex Capability	*Partial Duplex Capability*			*No Duplex Capability*
Echo Loss [dB]	*≥27*	*≥23*	*≥17*	*≥11*	*< 11*

Measurement Method

The test arrangement is according to clause 6.1.

The double talk signal consists of a sequence of orthogonal signals which are realized by voice-like modulated sine waves spectrally shaped similar to speech. The measurement signals used are shown in figure 6. A detailed description can be found in ITU-T Recommendation P.501 [17].

The signals are fed simultaneously in send and receive direction. The level in send direction is -16 dBm0 for 4-wire MGW, -16 dBm for 2-wire home MGW and -19 dBm0 for 2 wire network MGW (nominal level), the level in receive direction is -16 dBm0 (nominal level).

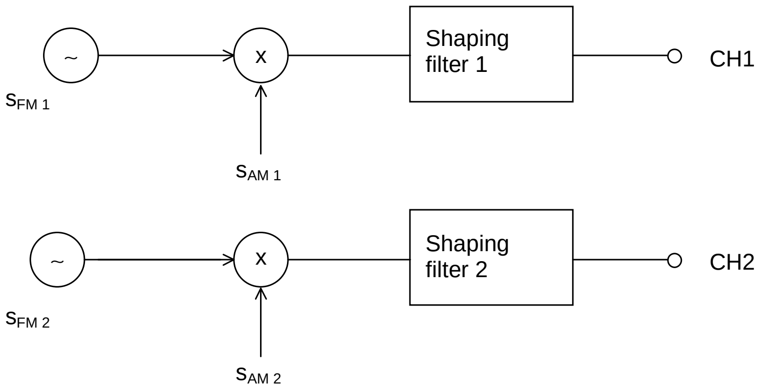

Figure 6: Measurement signals

$$s_{FM1,2}(t) = \sum A_{FM1,2} * \cos(2\pi t n * F_{01,2})\text{ ; n= 1, 2, etc.} \quad (1)$$

$$s_{AM1,2}(t) = A_{AM1,2} * \cos(2\pi t F_{AM1,2}); \quad (2)$$

The settings for the signals are as follows.

Table 27: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves for narrowband MGW

Receive Direction				Send Direction		
f_m[Hz]	$f_{mod(fm)}$ [Hz]	F_{am}[Hz]		f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
250	±5	3		270	±5	3
500	±10	3		540	±10	3
750	±15	3		810	±15	3
1 000	±20	3		1 080	±20	3
1 250	±25	3		1 350	±25	3
1 500	±30	3		1 620	±30	3
1 750	±35	3		1 890	±35	3
2 000	±40	3		2 160	±35	3
2 250	±40	3		2 400	±35	3
2 500	±40	3		2 900	±35	3
2 750	±40	3		3 150	±35	3
3 000	±40	3		3 400	±35	3
3 250	±40	3		3 650	±35	3
3 500	±40	3		3 900	±35	3
3 750	±40	3				
NOTE: Parameters of the Shaping Filter: Low Pass Filter, 5 dB/oct.						

Table 28: Parameters of the two Test Signals for Double Talk Measurement based on AM-FM modulated sine waves for wideband MGW

Receive Direction				Send Direction		
f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]		f_m[Hz]	$f_{mod(fm)}$[Hz]	F_{am}[Hz]
125	±2,5	3		150	±2,5	3
250	±5	3		270	±5	3
500	±10	3		540	±10	3
750	±15	3		810	±15	3
1 000	±20	3		1 080	±20	3
1 250	±25	3		1 350	±25	3
1 500	±30	3		1 620	±30	3
1 750	±35	3		1 890	±35	3
2 000	±40	3		2 160	±35	3
2 250	±40	3		2 400	±35	3
2 500	±40	3		2 650	±35	3
2 750	±40	3		2 900	±35	3
3 000	±40	3		3 150	±35	3
3 250	±40	3		3 400	±35	3
3 500	±40	3		3 650	±35	3
3 750	±40	3		3 900	±35	3
4 000	±40	3		4 150	±35	3
4 250	±40	3		4 400	±35	3
4 500	±40	3		4 650	±35	3
4 750	±40	3		4 900	±35	3
5 000	±40	3		5 150	±35	3
5 250	±40	3		5 400	±35	3
5 500	±40	3		5 650	±35	3
5 750	±40	3		5 900	±35	3
6 000	±40	3		6 150	±35	3
6 250	±40	3		6 400	±35	3
6 500	±40	3		6 650	±35	3
6 750	±40	3		6 900	±35	3
7 000	±40	3				
NOTE: Parameters of the Shaping Filter: f ≥ 250 Hz: Low Pass Filter, 5 dB/oct; f < 250 Hz,: High Pass Filter.						

The test signal is measured at the receive interface. The measured signal consists of the double talk signal which was fed in at the send interface and the echo signal. The echo signal is filtered by comb filter using mid-frequencies and bandwidth according to the signal components of the signal in receive direction (see ITU-T Recommendation P.501 [17]). The filter will suppress frequency components of the double talk signal.

In each frequency band which is used in receive direction the echo attenuation can be measured separately. The requirement for category 1 is fulfilled if in any frequency band the echo signal is either below the signal noise or below the required limit. If echo components are detectable, the classification is based on the table 26. The echo attenuation is to be achieved for **each individual frequency band** according to the different categories.

6.2.15.4 Minimum Activation Level and Sensitivity of Double Talk Detection

For further study.

6.2.16 Switching Characteristics

NOTE: Additional requirements may be needed in order to further investigate the effect of NLP implementations on the users' perception of speech quality.

6.2.16.1 Activation in Send Direction

The activation in send direction is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in send direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the send interface of the MGW.

Requirements

- For 4-wire MGW the minimum activation level $L_{S,min}$ shall be ≤ -32 dBm0.
- For 2-wire home MGW the minimum activation level $L_{S,min}$ shall be ≤ -32 dBm.
- For 2-wire network MGW the minimum activation level $L_{S,min}$ shall be ≤ -35 dBm.

The built-up time $T_{r,S,min}$ (measured with minimum activation level) should be ≤ 15 ms.

Measurement Method

The structure of the test signal is shown in figure 7. The test signal consists of CSS components according to ITU-T Recommendation P.501 [17] with increasing level for each CSS burst.

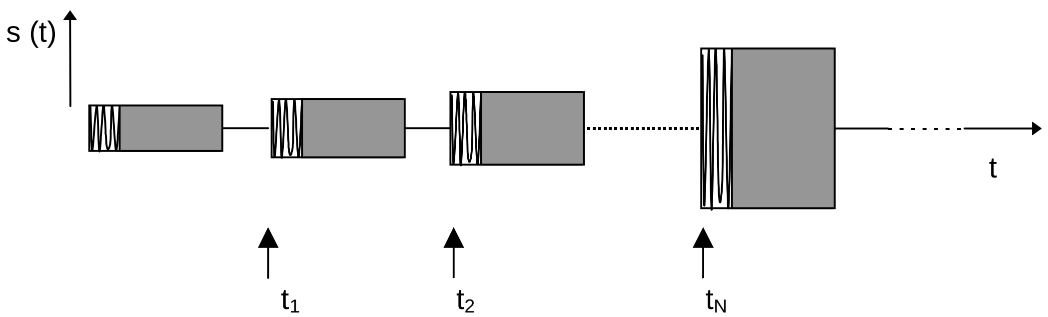

Figure 7: Test Signal to Determine the Minimum Activation Level and the Built-up Time

The settings of the test signal are as follows.

Table 29

	CSS Duration/ Pause Duration	**Level of the first CS Signal (active Signal Part at the MRP) (see note)**	**Level Difference between two Periods of the Test Signal**
CSS to Determine Switching Characteristic **in Send Direction**	~250 ms / ~450 ms	-34,3 dBm0 (4-wire MGW) -37,3 dBm (2-wire home MGW) -34,3 dBm (2-wire network MGW)	1 dB
NOTE: The level of the active signal part corresponds to an average level of -36 dBm0 (4-wire MGW) at the send interface of the MGW for the CSS according to ITU-T Recommendation P.501 [17] assuming a pause of about 100 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The test arrangement is described in clause 6.1.

The level of the transmitted signal is measured at the receive interface. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE: If the measurement using the CS-Signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one syllable word instead of the CS-Signal. The word used should be of similar duration, the average level of the word should be adapted to the CS-signal level of the according CS-burst.

6.2.16.2 Activation in Receive Direction

The activation in receive direction is mainly determined by the built-up time $T_{r,S,min}$ and the minimum activation level ($L_{S,min}$). The minimum activation level is the level required to remove the inserted attenuation in receive direction during idle mode. The built-up time is determined for the test signal burst which is applied with the minimum activation level.

The activation level described in the following is always referred to the test signal level at the receive interface of the MGW.

Requirements

The minimum activation level $L_{S,min}$ shall be ≤ -32 dBm0.
The built-up time $T_{r,S,min}$ (measured with minimum activation level) should be ≤ 15 ms.

Measurement Method

The structure of the test signal is shown in figure 7. The test signal consists of CSS components according to ITU-T Recommendation P.501 [17] with increasing level for each CSS burst.

The settings of the test signal are as follows.

Table 30

	CSS Duration/ Pause Duration	**Level of the first CS Signal (active Signal Part at the MRP)** (see note)	**Level Difference between two Periods of the Test Signal**
CSS to Determine Switching Characteristic **in Receive Direction**	~250 ms / ~450 ms	-34,3 dBm0	1 dB
NOTE: The level of the active signal part corresponds to an average level of -36 dBm0 at the receive interface of the MGW for the CSS according to ITU-T Recommendation P.501 [17] assuming a pause of about 100 ms.			

It is assumed that the pause length of about 450 ms is longer than the hang-over time so that the test object is back to idle mode after each CSS burst.

The test arrangement is described in clause 6.1.

The level of the transmitted signal is measured at the send interface. The measured signal level is referred to the test signal level and displayed vs. time. The levels are calculated from the time domain using an integration time of 5 ms.

The minimum activation level is determined from the CSS burst which indicates the first activation of the test object. The time between the beginning of the CSS burst and the complete activation of the test object is measured.

NOTE: If the measurement using the CS-Signal does not allow to clearly identify the minimum activation level, the measurement may be repeated by using a one syllable word instead of the CS-Signal. The word used should be of similar duration, the average level of the word should be adapted to the CS-signal level of the according CS-burst.

6.2.16.3 Silence Suppression and Comfort Noise Generation

To provide optimum speech quality as perceived by end users, these speech processing features should be avoided in the transmission path.

For further study.

6.2.17 Background Noise Performance

To provide optimum speech quality as perceived by end users, noise cancelling should be avoided in the transmission path. The preferred location for the implementation of such feature is the terminal.

However, the requirement and measurement methods exposed below are meant to ensure good performance of the media gateway echo canceller in the presence of background noise.

6.2.17.1 Performance in Send Direction in the Presence of Background Noise

Requirement

The level of comfort noise, if implemented, shall be within in a range of +2 and -5 dB compared to the original (transmitted) background noise. The noise level is calculated with psophometric weighting.

NOTE 1: It is advisable that the comfort noise matches the original signal as good as possible (from a perceptional point of view).

NOTE 2: Input for further specification necessary (e.g. on temporal matching).

The spectral difference between comfort noise and original (transmitted) background noise shall be within the mask given through straight lines between the breaking points on a logarithmic (frequency) - linear (dB sensitivity) scale as given in table 31.

Table 31: Requirements for Spectral Adjustment of Comfort Noise (Mask) for narrowband MGW

Frequency	Upper Limit	Lower Limit
200 Hz	12 dB	-12 dB
800 Hz	12 dB	-12 dB
800 Hz	10 dB	-10 dB
2 000 Hz	10 dB	-10 dB
2 000 Hz	6 dB	-6 dB
4 000 Hz	6 dB	-6 dB
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Table 32: Requirements for Spectral Adjustment of Comfort Noise (Mask) for wideband MGW

Frequency	Upper Limit	Lower Limit
200 Hz	12 dB	-12 dB
800 Hz	12 dB	-12 dB
800 Hz	10 dB	-10 dB
2 000 Hz	10 dB	-10 dB
2 000 Hz	6 dB	-6 dB
4 000 Hz	6 dB	-6 dB
8 000 Hz	6 dB	-6 dB
NOTE: All sensitivity values are expressed in dB on an arbitrary scale.		

Measurement Method

The background noise simulation as described in clause 6.1 is used.

First the background noise transmitted in send is recorded at the POI for a period of at least 20 s.

In a second step a test signal is applied in receive direction consisting of an initial pause of 10 s and a periodical repetition of the Composite Source Signal in receive direction (duration 10 s) with nominal level to enable comfort noise injection simultaneously with the background noise. For the measurement the background noise sequence has to be started at the same point as it was started in the previous measurement. Alternatively other speech like test signals (e.g. artificial voice) with the same signal level can be used.

The transmitted signal is recorded in send direction at the POI.

The power density spectra measured in send direction without far end speech simulation averaged between 10 s and 20 s is referred to the power density spectrum measured in send direction determined during the period with far end speech simulation in receive direction averaged between 10 s and 20 s. Level and spectral differences between both power density spectra are analysed and compared to the requirements.

6.2.17.2 Quality of Speech with Background Noise

Generally this measurement is performed in send direction. For IP-IP MGW the measurement is made in both directions.

Requirement

Speech Quality for wideband systems can be tested based on EG 202 396-3 [i.3]. The test method is applicable for narrowband (100 Hz to 4 kHz) and wideband (100 Hz to 8 kHz) transmission systems. The test method described leads to three MOS-LQO quality numbers:

For narrowband MGW:

- N-MOS-LQOn: Transmission quality of the background noise narrowband
- S-MOS-LQOn: Transmission quality of the speech narrowband
- G-MOS-LQOn: Overall transmission quality narrowband
- For the background noises defined in clause 6.1 the following requirements apply:
- N-MOS-LQOn ≥ 3,5
- S-MOS-LQOn ≥ 3,5
- G-MOS-LQOn ≥ 3,5

For wideband MGW:

- N-MOS-LQOw: Transmission quality of the background noise wideband
- S-MOS-LQOw: Transmission quality of the speech wideband

- G-MOS-LQOw: Overall transmission quality wideband
- For the background noises defined in clause 6.1 the following requirements apply:
- N-MOS-LQOw ≥ 3,5
- S-MOS-LQOw ≥ 3,5
- G-MOS-LQOw ≥ 3,5

NOTE: It is recommended to test the MGW performance with other types of background noises if the terminal is likely to be exposed to other noises than specified in clause 6.1.

Measurement Method

The speech plus background noise signal is generated as described in clause 6.1 is used. The test arrangement is described in clause 6.1.

The background noise should be applied for at least 5 s in order to adapt noise reduction algorithms in advance to the test.

The near end speech signal consists of 8 sentences of speech (2 male and 2 female talkers, 2 sentences each). Appropriate speech samples can be found in ITU-T Recommendation P.501 [17]. The preferred language is English since the objective method was validated with English language in narrowband.

Three signals are required for the tests:

1) The clean speech signal is used as the undisturbed reference (see [27]).
2) The previously recorded speech plus undisturbed background noise signal (see clause 6.1.6) is used as the unprocessed speech plus noise reference.
3) The send signal, recorded at the output of the gateway send path is used as the degraded signal.

N-MOS-LQOn, S-MOS LQOn and G-MOS LQOn are calculated as described in EG 202 396-3 [i.3].

6.2.17.3 Quality of Background Noise Transmission (with Far End Speech)

Requirement

The test is carried out applying the Composite Source Signal in receive direction. During and after the end of Composite Source Signal bursts (representing the end of far end speech simulation) the signal level in send direction should not vary more than 10 dB (during transition to transmission of background noise without far end speech). The measurement is conducted for all types of background noise as defined in clause 6.1.

Measurement Method

The test arrangement is according to clause 6.1.

The background noises are generated as described in clause 6.1.

First the measurement is conducted without inserting the signal at the far end. At least 10 s of noise is analysed. The background signal level versus time is calculated using a time constant of 35 ms. This is the reference signal.

In a second step the same measurement is conducted but with inserting the CS-signal at the far end. The exactly identical background noise signal is applied. The background noise signal must start at the same point in time which was used for the measurement without far end signal. The background noise should be applied for at least 5 seconds in order to allow adaptation of the noise reduction algorithms. After at least 5 seconds a Composite Source Signal according to ITU-T Recommendation P.501 [17] is applied in receive direction with duration of ≥ 2 CSS periods. The test signal level is -16 dBm0 at the receive interface.

The send signal is recorded at the output of the gateway send path. The test signal level versus time is calculated using a time constant of 35 ms.

The level variation in receive direction is determined during the time interval when the CS-signal is applied and after it stops. The level difference is determined from the difference of the recorded signal levels vs. time between reference signal and the signal measured with far end signal.

6.2.17.4 Quality of Background Noise Transmission (with Near End Speech)

Requirement

The test is carried out applying a simulated speech signal in send direction. During and after the end of the simulated speech signal (Composite Source Signal bursts) the signal level in send direction should not vary more than 10 dB.

Measurement Method

The test arrangement is according to clause 6.1.

The background noises are generated as described in clause 6.1. The background noise should be applied for at least 5 s in order to allow adaptation of the noise reduction algorithms.

The near end speech is simulated using the Composite Source Signal according to ITU-T Recommendation P.501 [17] with duration of ≥ 2 CSS periods. The CSS plus the background noise are pre-recorded as described in clause 6.1.

The send signal is recorded at the electrical reference point. The test signal level versus time is calculated using a time constant of 35 ms.

First the measurement is conducted without inserting the signal at the near end. The signal level is analysed vs. time. In a second step the same measurement is conducted but with inserting the CS-signal at the near end. The level variation is determined by the difference between the background noise signal level without inserting the CS-signal and the maximum level of the noise signal during and after the CS-bursts in send direction.

6.2.18 Quality of Echo Cancellation

Echo measurements apply only for narrowband media gateways. In wideband it must be assumed that all terminals provide a sufficient echo cancellation so no further echo cancellation is required in the media gateway.

In IP-to-IP MGWs there should not be any echo cancellation.

6.2.18.1 Echo paths

For narrowband 2-wire MGW the following echopath simulations are used:

- Different DECT phone echo paths
- Different analogue phone echo paths
- Different analogue hands-free echo paths

For narrowband 4-wire MGW the following echopath simulations are used:

- Different ISDN phone echo paths
- Different ISDN phone hands-free echo paths
- Different DECT phone echo paths
- Infinite echo loss
- Different analogue terminal echo paths
- Different analogue terminal hands-free echo paths

If the narrowband 4-wire mediagateway is connected to a TDM network the additional delay introduced by the TDM network has to be included in the echopath.

No explicit echo paths are given in the present document. The types of echo paths used is subject to be agreed upon with the network operator or the test house.

NOTE: A DECT phone echo path impulse response is provided in Annex A and can be used in combination with appropriate hybrid echo simulation or artificial echo loss simulation in case of an ISDN fixed part.

6.2.18.2 Echo Performance According to ITU-T Recommendation G.168

The tests of the echo canceller cancellation performance are performed according to ITU-T Recommendation G.168 [24].

The following Tests of ITU-T Recommendation G.168 apply:

- Test 2A
- Test 2C
- Test 4
- Test 5
- Test 9

For all tests the NLP remains enabled, tests requiring the deactivation of the NLP are not applicable.

The performance has to checked with all different echo paths relevant for the media gateway as described in clause 6.2.18.1.

Requirement

See ITU-T Recommendation G.168.

Measurement Method

The test arrangement is according to clause 6.1.

See ITU-T Recommendation G.168 [24].

6.2.18.3 Terminal Coupling Loss (TCLw)

Requirement

The TCLw provided by the gateway in conjunction with typical echo paths as described in clause 6.2.18.1 shall be ≥ 55 dB.

Measurement Method

The test arrangement is according to clause 6.1.

The appropriate echo paths as described in clause 6.2.18.1 are used for the tests.

The attenuation from electrical reference point input to electrical reference point output shall be measured using a speech like test signal.

Before the actual test a training sequence consisting of 10 s male artificial voice followed by 10 s female artificial voice according to ITU-T Recommendation P.50 [15] is applied. The training sequence level shall be -16 dBm0 in order not to overload the codec.

The test signal following immediately the training sequence is a PN sequence complying with ITU-T Recommendation P.501 [17] with a length of 4 096 points (for the 48 kHz sampling rate) and a crest factor of 6 dB. The length of the complete test signal composed of at least four sequences of CSS shall be at least one second (1,0 s). The test signal level is -3 dBm0 (from 50 Hz to 4 kHz). The low crest factor is achieved by random alternation of the phase between -180° and 180°.

The TCLw is calculated according to ITU-T Recommendation G.122 [8], clause B.4 (trapezoidal rule). For the calculation the averaged measured echo level at each frequency band is referred to the averaged test signal level measured in each frequency band. For the measurement a time window has to be applied adapted to the duration of the actual pn-sequence of the test signal (200 ms) choosing the pn-sequence of the third CSS.

6.2.18.4 Temporal Echo Effects

Requirement

This test is intended to verify that the system will maintain sufficient echo attenuation during single talk. The measured echo attenuation during single talk should not decrease by more than 6 dB from the maximum TCL measured during the test.

Measurement Method

The test arrangement is according to clause 6.1.

The test signal consists of periodically repeated Composite Source Signal according to ITU-T Recommendation P.501 [17] with an average level of -5 dBm0 as well as an average level of -25 dBm0. The echo signal is analysed during a period of at least 2,8 s which represents 8 periods of the CS signal. The integration time for the level analysis shall be 35 ms, the analysis is referred to the level analysis of the reference signal. The TCL variation is compared to the maximum TCL achieved in the test.

The measurement result is displayed as attenuation vs. time. The exact synchronization between input and output signal has to be guaranteed.

NOTE 1: In addition tests with more speech like signals should be made, e.g. ITU-T Recommendation P.50 [15] to see time variant behaviour of EC. However, for such tests the simple broadband attenuation based test principle as described above cannot be applied due to the time varying spectral content of the speech like signals.

NOTE 2: The analysis is conducted only during the active signal part, the pauses between the Composite Source Signals are not analysed. The analysis time is reduced by the integration time of the level analysis (35 ms).

6.2.18.5 Spectral Echo Attenuation

Requirement

The echo attenuation vs. frequency shall be below the tolerance mask given in table 33.

Table 33: Echo attenuation limits for narrowband MGW

Frequency	Limit
100 Hz	-20 dB
200 Hz	-30 dB
300 Hz	-38 dB
800 Hz	-34 dB
1 500 Hz	-33 dB
2 600 Hz	-24 dB
4 000 Hz	-24 dB
NOTE 1: All sensitivity values are expressed in dB on an arbitrary scale. NOTE 2: The limit at intermediate frequencies lies on a straight line drawn between the given values on a log (frequency) - linear (dB) scale.	

During the measurement it should be ensured that the measured signal is really the echo signal and not the Comfort Noise which possibly may be inserted in send direction in order to mask the echo signal.

Measurement Method

The test arrangement is according to clause 6.1.

Before the actual measurement a training sequence is fed in consisting of 10 seconds CS signal according to ITU-T Recommendation P.501 [17]. The level of the training sequence is -16 dBm0.

The test signal consists of a periodically repeated Composite Source Signal. The measurement is carried out under steady-state conditions. The average test signal level is -16 dBm0, averaged over the complete test signal. 4 CS signals including the pauses are used for the measurement which results in a test sequence length of 1,4 s. The power density spectrum of the measured echo signal is referred to the power density spectrum of the original test signal. The analysis is conducted using FFT analysis with 8 k points (48 kHz sampling rate, Hanning window).

The spectral echo attenuation is analysed in the frequency domain in dB.

6.2.18.6 Occurrence of Artefacts

For further study.

6.2.19 Variant Impairments; Network dependant

6.2.19.1 Clock Accuracy Send

Requirement

The clock drift in send direction between the MGW and the IP reference interface shall be less than 40 ppm under ideal network conditions.

NOTE: The clock accuracy does not cover all possible network configurations. Especially it is not sufficient for data transmission or distributed TDM PBX where synchronisation is required.

Measurement Method

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyze clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1,2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level at the MGW interface is -16 dBm0 for 4-wire MGW, -16 dBm for 2-wire home MGW and -19 dBm0 for 2-wire network MGW.

A cross correlation analysis versus time is carried out over the whole 120 s sequence between the received and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock drift within an analysis time range of at least 60 s is calculated as follows:

$$clock\ drift\ [ppm] = \frac{delay\ drift\ [s]}{analysis\ duration\ [s]} \cdot 1 \cdot 10^6 \qquad (3)$$

6.2.19.2 Clock Accuracy Receive

Requirement

The clock drift in receive direction between the IP reference interface and the MGW shall be less than 40 ppm under ideal network conditions.

Measurement Method

A sequence of CS signals (active signal length = 250 ms) is repeated for 120 s in order to analyze clock accuracy and any other time-variant delay. The pause length between two CS bursts is 100 ms and 1,2 s after every fourth burst in order to simulate a speech pause, which may lead to buffer adjustments. The test signal level at the IP reference interface is -16 dBm0.

A cross correlation analysis versus time is carried out over the whole 120 s sequence between the received and the original test signal. The duration of the measurement (120 s) is indicated on the x-axis, the result of the cross correlation analysis (delay) is plotted on the y-axis.

The resulting clock drift within an analysis time range of at least 60 s is calculated as follows:

$$clock\ drift\,[ppm] = \frac{delay\ drift\,[s]}{analysis\ duration\,[s]} \cdot 1 \cdot 10^{6} \qquad (4)$$

6.2.19.3 Send Delay Variation

Requirement

The measured maximum delay variation in send direction of the MGW under test should be less than 5 ms.

NOTE: Any delay variation introduced in send direction will lead to potentially increased delay due to increased de-jitter buffer at the far end terminal.

Measurement Method

The RTP data stream in send direction should be monitored with a tap or a switch providing a monitoring port, positioned at the location of the network impairment simulator (see clause 6.1).
The test arrangement is according to clause 6.1.

The monitoring time should be 60 s. A speech signal according to P.50 [15] is played back in send direction using a nominal network level of -16 dBm0 for 4-wire MGW, -16 dBm for 2-wire home MGW and -19 dBm0 for 2-wire network MGW. This speech signal is only necessary to make sure, RTP is played out, and even in the case VAD is active.

The delay variation for each packet D(i) compared to the first packet of the analysis period is calculated:

$$D(i) = \Delta t_{eff(i)} - \Delta t_{exp(i)} \qquad (5)$$

With:

- $\Delta t_{exp(i)}$ = the expected time between packet i and the first packet based on RTP timestamp information; and
- $\Delta t_{eff(i)}$ = the effective time between packet i and the first packet.

Maximum delay variation = MAX(|D(i)|)

6.2.19.4 Delay versus Time Receive

For further study.

6.2.19.5 Quality of Jitter buffer adjustment

For further study.

6.2.20 Immunity to DTMF False Detection in Send Direction

On the TDM interface of a gateway, the incoming audio signal may be analysed in order to detect DTMF tones to be transmitted in a separate way than speech, on the IP network (see RFC 2833 [i.4], RFC 4733 [i.5]). It can occur that some parts of speech signal are analysed by the gateway as DTMF and therefore processed as such. The result of it is that the far end listener will hear a tone instead of a syllable of a word. This must be avoided.

Other aspects of DTMF transmission performance are not impacting speech quality.

Requirement

No more than 5 false detections of DTMF must be reported on duration of 30 minutes.

Measurement Method

The test arrangement is according to clause 6.1.

The test signal to be used for the measurements shall be composed of speech with a relatively high speech activity ratio. The resulting signal, captured in the IP network, can be analysed in real time or after recording. The number of DTMF tones present (i.e. detected during this analysis, in conformance with DTMF signals specifications, in terms of frequencies, levels and durations) in this signal is counted and reported.

The test signal used is found in Annex B.

6.3 Codec Specific Requirements

6.3.1 Send Delay

For a MGW, send delay is defined as the one-way delay from a synchronous interface (e.g. ISDN, analogue) towards a packet switched interface. The total send delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements from the TDM input reference point to the packet output reference point shown in figure A.1 in ITU-T Recommendation G.1020 [14], respectively.

The send delay T(s) is defined as follows:

$$T(s) = T(ps) + T(la) + T(aif) + T(asp) \quad (6)$$

Where:

- T(ps) = packet size = N × T(fs).
- N = number of frames per packet.
- T(fs) = frame size of encoder.
- T(la) = look-ahead of encoder.
- T(aif) = air interface framing.
- T(asp) = allowance for signal processing.

The additional delay required for IP packet assembly and presentation to the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time usually will be very small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE 1: The size of T(aif) is for further study.

Requirement

The allowance for signal processing shall be T(asp) < 10 ms.

NOTE 2: With the knowledge of the codec specific values for T(fs) and T(la) the requirements for send delay for any type of coder and any packet size T(ps) can easily be calculated by equation 6. Tables 30 and 31 provide requirements calculated accordingly for frequently used codecs and packet sizes.

Table 34

Codec	N	T(fs) in ms	T(ps) in ms	T(la) in ms	T(aif) in ms	T(asp) in ms	T(s) Requirement in ms
ITU-T Recommendation G.711 [9]	80	0,125	10	0	0	10	< 20
ITU-T Recommendation G.711 [9]	160	0,125	20	0	0	10	< 30
ITU-T Recommendation G.729 [12]	1	10	10	5	0	10	< 25
ITU-T Recommendation G.729 [12]	2	10	20	5	0	10	< 35
ITU-T Recommendation G.723.1 [10] (5,3 kbit/s and 6,3 kbit/s)	1	30	30	7,5	0	10	< 47,5

Table 35

Codec	N	T(fs) in ms	T(ps) in ms	T(la) in ms	T(aif) in ms	T(asp) in ms	T(s) Requirement in ms
G.722 [26]	80	0,0625	10	0	0	10	< 20,0625
	160	0,0625	20	0	0	10	< 30,0625
G.722.1 [27]	1	20	10	5	0	10	To be completed
	2	20	20	5	0	10	To be completed
L16-256	160	0,0625	10	0	0	10	< 20,0625

Measurement Method

The test signal to be used for the measurements shall be a Composite Source Signal (CSS) as described in ITU-T Recommendation P.501 [17]. The test signal consists of the voiced part as described in ITU-T Recommendation P.501 [17] followed by a pseudo random noise sequence with a periodicity of minimum 500 ms. The test signal level shall be -16 dBm0, measured at the electrical test point. The test signal level is averaged over the complete test signal sequence.

NOTE 3: If the expected delay is higher than 500 ms a pseudo random sequence with a higher periodicity should be used.

The delay is calculated using the cross correlation function between the signal at the output and the signal at the input. The cross correlation analysis has to be chosen in such a way that the maximum delay of 500 ms can be analysed. The measurement is corrected by the delay introduced by the test equipment.

The delay is expressed in ms, determined from the maximum of the cross correlation function.

NOTE 4: Delay may be time variant. Therefore constant monitoring of the actual delay may be required when evaluating the range of delay which can be observed in a given connection. The test setup should take into account either real network conditions or the tools needed to simulate typical causes for time variant delay (e.g. packet loss) during the measurement period. Other methods like running cross correlation or delay estimation procedures e.g. used in PESQ (ITU-T Recommendation P.862 [19]) may be used.

6.3.2 Receive delay

For a MGW, receive delay is defined as the one-way delay from a packet based interface towards a synchronous interface (e.g. ISDN, analogue). The total receive delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements from the packet input reference point to the TDM output reference point shown in figure A.2 of ITU-T Recommendation G.1020 [14], respectively.

The receive delay T(r) is defined as follows:

$$T(r) = T(fs) + T(aif) + T(jb) + T(plc) + T(asp) \quad (7)$$

Where:

- T(fs) = frame size of encoder.
- T(aif) = air interface framing.
- T(jb) = jitter buffer size.
- T(plc) = PLC buffer size.
- T(asp) = allowance for signal processing.

The additional delay required for IP packet dis-assembly and presentation from the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time usually will be very small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE 1: The size of T(aif) is for further study.

Requirements

The allowance for signal processing shall be T(asp) < 10 ms.

The additional delay introduced by the jitter buffer shall be T(jb) ≤ 10 ms.

For Coders without integrated PLC the additional PLC buffer size shall be T(plc) < 10 ms.

For Coders with integrated PLC the additional PLC buffer size shall be T(plc) = 0 ms.

NOTE 2: With the knowledge of the codec specific values for T(fs) and T(la) the requirements for receive delay for any type of coder and any packet size T(ps) can easily be calculated by equation 7. The tables 32 and 33 provide requirements calculated accordingly for some frequently used codecs and packet sizes as an example.

Table 36

Codec	N	T(fs) in ms	T(aif) in ms	T(jb) in ms	T(plc) in ms	T(asp) in ms	T(r) in ms	T(r) Requirement in ms
ITU-T Recommendation G.711 [9]	80	0,125	0	10	10	10	< 30,125	< 31
ITU-T Recommendation G.711 [9]	160	0,125	0	10	10	10	< 30,125	< 31
ITU-T Recommendation G.729 [12]	1	10	0	10	0	10	< 30	< 30
ITU-T Recommendation G.729 [12]	2	10	0	10	0	10	< 30	< 30
ITU-T Recommendation G.723.1 (5,3 kbit/s and 6,3 kbit/s) [10]	1	30	0	10	0	10	< 50	< 50
NOTE 1: T(ps) = packet size = N × T(fs). NOTE 2: N = number of frames per packet.								

Table 37

Codec	N	T(fs)	T(fi)	T(aif)	T(jb)	T(plc)	T(asp)	T(r) Requirement
G.722 [26]	80	0,0625	0	0	10	10	10	< 30,0625
G.722 [26]	160	0,0625	0	0	10	10	10	< 30,0625
G.722.1 [27]	1	20	0	0	10	0	10	< 40
G.722.1 [27]	2	20	0	0	10	0	10	< 40
L 16-256	160	0,0625	0	0	10	10	10	< 30,0625
NOTE 1: T(ps) = packet size = N × T(fs). NOTE 2: N = number of frames per packet.								

NOTE 3: These requirements are based on the lowest possible delay values which can be expected under ideal network conditions. Caution should be exercised to ensure that the terminal is operated under optimum conditions in order to avoid adverse effects, e.g. network conditions, settings and memory effects of the terminal jitter buffer.

Measurement Method

The test signal to be used for the measurements shall be a Composite Source Signal (CSS) as described in ITU-T Recommendation P.501 [17]. The test signal consists of the voiced part as described in ITU-T Recommendation P.501 [17] followed by a pseudo random noise sequence with a periodicity of minimum 500 ms. The test signal level shall be -16 dBm0, measured at the electrical test point. The test signal level is averaged over the complete test signal sequence.

NOTE 4: If the expected delay is higher than 500 ms a pseudo random sequence with a higher periodicity should be used.

The delay is calculated using the cross correlation function between the signal at the input and the signal at the output. The cross correlation analysis has to be chosen in such a way that the maximum delay of 500 ms can be analysed. The measurement is corrected by the delay introduced by the test equipment.

The delay is expressed in ms, determined from the maximum of the cross correlation function.

NOTE 5: Delay may be time variant. Therefore constant monitoring of the actual delay may be required when evaluating the range of delay which can be observed in a given connection. The test setup should take into account either real network conditions or the tools needed to simulate typical causes for time variant delay (e.g. packet loss) during the measurement period. Other methods like running cross correlation or delay estimation procedures e.g. used in PESQ (ITU-T Recommendation P.862 [19]) may be used.

6.3.3 Delay for IP-to-IP MGW

For a MGW with packet switched interfaces on both sides, the delay requirement has to be measured for each direction. The total delay per direction is the upper bound on the mean delay and takes into account the delay contributions of all of the elements from the packet input reference point to a hypothetical points corresponding to the TDM output reference point shown in figure A.2 and from a hypothetical points corresponding to the TDM input reference point to the packet output reference point in figure A.1 of ITU-T Recommendation G.1020 [14], respectively.

NOTE: An IP-to-IP MGW includes a decoding and a coding function in series.

The delay requirements should be taken out of tables 34 (table 35 for wideband) and 36 (table 37 for wideband). The delay should be the sum of the codec specific value from table 34 (or 35) and the codec specific value from table 36 (or 37) minus signal processing time and PLC time:

$$T(IP\text{-}IP) = T(s) + T(r) - 2 \times T(asp) - T(plc) \qquad (8)$$

where T(s) and T(r) have to be chosen according to the used codecs on both sides of the MGW.

EXAMPLE: For a transcoding function G.711 [9] to G.729 [12] with 10ms audio per packet on both sides, the delay should be less than:

- T(IP-IP, G.711 [9] to G.729 [12]) < 31 ms + 25 ms – 2 x 10 ms – 10 ms = 26 ms

- T(IP-IP, G.711 [9] to G.729 [12]) < 20 ms + 30 ms – 2 x 10 ms – 0 ms = 30 ms

6.3.4 Objective Listening Speech Quality MOS-LQO in Send direction

The listening speech quality tests are conducted under clean network conditions.

Requirements

For narrowband MGW the requirements for the listening speech quality are as follows:

Table 38

Speech coder	MOS-LQON	MOS-LQOS
ITU-T Recommendation G.711 [9]	> 4,2	> 3,4
ITU-T Recommendation G.729 [12]	> 3,8	> 2,9
ITU-T Recommendation G.723.1 [10]	> 3,5	> 2,7
ITU-T Recommendation G.726 @ 32 kbit/s [11]	> 3,9	> 3,1
GSM EFR [1] and AMR @ 12,2 kbit/s [2]	> 4,0	> 3,2
ITU-T Recommendation G.729.1 @ 8 kbit/s [13]	> 3,8	> 2,9

For wideband MGW the requirements for the listening speech quality are as follows:

Table 39

Speech coder	MOS-LQOS
ITU-T Recommendation G.722 [26]	> 4,0
G.729.1 @ 32 kbit/s [13]	> 4,2
G.722.1 [27]	> 4,0
L16-256	> 4,3
AMR-WB [28]	> 4,1

NOTE: P.863 [25] is using a superwideband scale instead of a mixed scale. Not sufficient experience is available so far with this method. Therefore the numbers given for MOS-LQOS are provisional and may be updated with a later revision of the document.

Measurement method

For narrowband media gateways MOS-LQON is measured either using ITU-T Recommendation P. 862 [19] with mapping as defined in P.862.1 [29] or ITU-T Recommendation P.863 [25] in narrowband mode.

For wideband media gateways or media gateways supporting both, narrowband and wideband mode MOS-LQOS is measured using ITU-T Recommendation P.863 [25] in superwideband mode.

6.3.5 Objective Listening Quality MOS-LQO in Receive direction

The listening speech quality tests are conducted under clean network conditions as well as with network impairments simulated. In addition to the listening speech quality tests the delay is measured.

The method for the assessment of MOS-LQO scores is either ITU-T Recommendations P.862 [19] or P.863 [25].

Requirements

The requirement for the listening speech quality and the delay under clean network conditions are as follows:

Table 40

Speech coder	MOS-LQON	MOS-LQOS
ITU-T Recommendation G.711 [9]	> 4,2	> 3,4
ITU-T Recommendation G.729 [12]	> 3,8	> 2,9
ITU-T Recommendation G.723.1 [10]	> 3,5	> 2,7
ITU-T Recommendation G.726 @ 32 kbit/s [11]	> 3,9	> 3,1
GSM EFR [1] and AMR @ 12,2 kbit/s [2]	> 4,0	> 3,2
ITU-T Recommendation G.729.1 @ 8 kbit/s [13]	> 3,8	> 2,9

For wideband MGW the requirements for the listening speech quality are as follows:

Table 41

Speech coder	MOS-LQOS
ITU-T Recommendation G.722 [26]	> 4,0
G.729.1 @ 32 kbit/s [13]	> 4,2
G.722.1 [27]	> 4,0
L16-256	> 4,3
AMR-WB [28]	> 4,1

NOTE 1: P.863 [25] is using a superwideband scale instead of a mixed scale. Not sufficient experience is available so far with this method. Therefore the numbers given for MOS-LQOS are provisional and may be updated with a later revision of the document.

Test method

For narrowband media gateways MOS-LQON is measured either using ITU-T Recommendation P. 862 [19] with mapping as defined in P.862.1 [29] or ITU-T Recommendation P.863 [25] in narrowband mode.

For wideband media gateways or media gateways supporting both, narrowband and wideband mode MOS-LQOS is measured using ITU-T Recommendation P.863 [25] in superwideband mode.

For the performance tests with network impairments the following settings are used.

Table 42: Network Conditions for Electrical-Electrical Measurements (Speech Samples)

Condition	Packet Loss (Equal)	Delay Variation
0 (see note 2) (VAD)	0	No
1	0	No
2	0	20 ms (see note 1)
3	1 %	No
4	1 %	20 ms (see note 1)
5	3 %	No
NOTE 1: Delay Variation produced with a Pareto-Distribution and r = 0,5. NOTE 2: VAD on, all other conditions (1-5) tested with VAD off. NOTE 3: For some network emulation tools, it is necessary to introduce a constant delay to offer the possibility to generate a delay variation distribution. This delay has to be subtracted from the measured delay before interpreting the results.		

Table 43: Requirements for ITU-T Recommendation G.711 [9] speech codecs

Condition	MOS-LQON	MOS-LQOS	Delay
0			
1	> 4,2	> 3,4	< 31 ms
2	> 4,0	> 3,1	< 51 ms
3	> 4,0	> 3,1	< 31 ms
4	> 4,0	> 3,1	< 51 ms
5	> 4,0	> 3,1	< 31 ms
NOTE: The settings are derived from the ones used in the ETSI Plugtest VoIP speech quality test events.			

Table 44: Requirements for ITU-T Recommendation G.729 [12] speech codecs

Condition	MOS-LQON	MOS-LQOS	Delay
1	> 3,8	> 2,9	< 30 ms
2	> 3,7	> 2,8	< 50 ms
3	> 3,7	> 2,8	< 30 ms
4	> 3,7	> 2,8	< 50 ms
5	> 3,3	> 2,3	< 30 ms

Table 45: Requirements for ITU-T Recommendation G.723.1 [10] speech codecs

Condition	MOS-LQON	MOS-LQOS	Delay
1	> 3,5	> 2,7	< 50 ms
2	> 2,8	> 2,5	< 70 ms
3	> 2,8	> 2,5	< 50 ms
4	> 2,8	> 2,5	< 70 ms
5	> 2,7	> 2,4	< 50 ms

Table 46: Requirements for ITU-T Recommendation G.722 [26] speech codecs

Condition	MOS-LQOS	Delay
1	> 4,0	< 30,0625 ms
2	> 3,8	< 50,0625 ms
3	> 3,8	< 30,0625 ms
4	> 3,8	< 50,0625 ms
5	> 3,6	< 30,0625 ms
NOTE: The settings are derived from the ones used in the ETSI Plugtest VoIP speech quality test events.		

Table 47: Requirements for ITU-T Recommendation G.722.1 [27] speech codecs

Condition	MOS-LQOS	Delay
1	> 4,0	< 40 ms
2	> 3,8	< 60 ms
3	> 3,8	< 40 ms
4	> 3,8	< 60 ms
5	> 3,8	< 40 ms

NOTE 2: P.863 [25] is using a superwideband scale instead of a mixed scale. Not sufficient experience is available so far with this method. Therefore the numbers given for MOS-LQOS are provisional and may be updated with a later revision of the present document.

6.3.5.1 Efficiency of Packet Loss Concealment (PLC)

A method for assessing the efficiency of packet loss concealment van be found in TR 102 927 [i.6].

Requirements are for further study.

6.3.5.2 Efficiency of Delay Variation Removal

For further study.

Annex A (informative): Impulse Response of a Narrowband and Wideband DECT PP

The following (embedded EXCEL-) table provides two impulse responses measured for a commercially available DECT PP in narrowband and wideband mode. They are used as the basis for echopath simulation of a typical DECT PP connected to an IAD. It should be noted that the impulse response only covers linear distortion; non-linear distortions as often found in portable phones today are not covered by impulse responses.

Furthermore the impulse responses have to be complemented with the impulse response of the fixed part. In case of an analogue fixed part a hybrid impulse response is used. Examples for narrowband impulse responses can be found in [24]. Alternatively individual responses can be used. In case of a digital fixed part the artificial echo loss of 24 dB is used.

The impulse responses contain 1 024 taps at a sampling frequency of 48 kHz (24 bit resolution). This results in an impulse response length of 21,3 ms. The impulse responses are given by the following coefficients:

Narrowband	Wideband
8721	4785
8687	26708
8737	32710
8856	18356
9025	-7482
9228	-28534
9447	-30970

The echopath simulation is separated in two parts. First the filter is scaled in such a way that 0 dB TCLw (in narrowband) and 0 dB echo attenuation (in wideband) is achieved. The required TCLw resp. Echo attenuation used finally for testing is adjusted separately. For echo canceller testing a TCLw resp. Echo attenuation of e.g. 35 dB is realistic and recommended.

The frequency characteristics of the echo path after adjusting the filter to 0 dB TCLw (narrowband) and 0 dB echo attenuation (wideband) is shown in Figures A.1 and A.2.

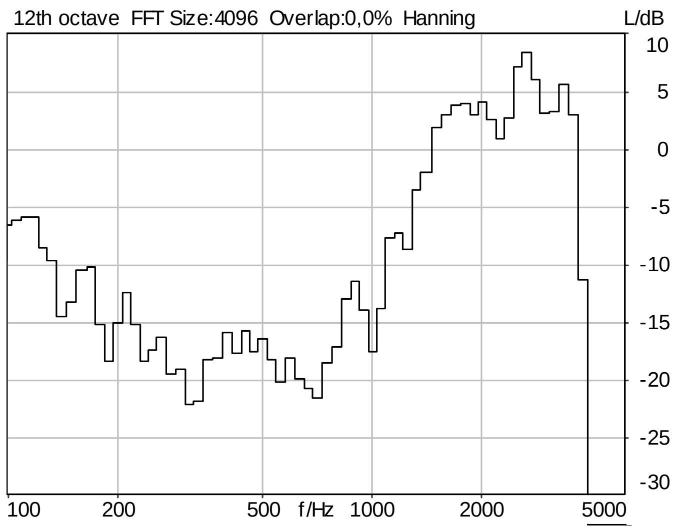

Figure A.1: ERL(f) for the narrowband echopath simulation

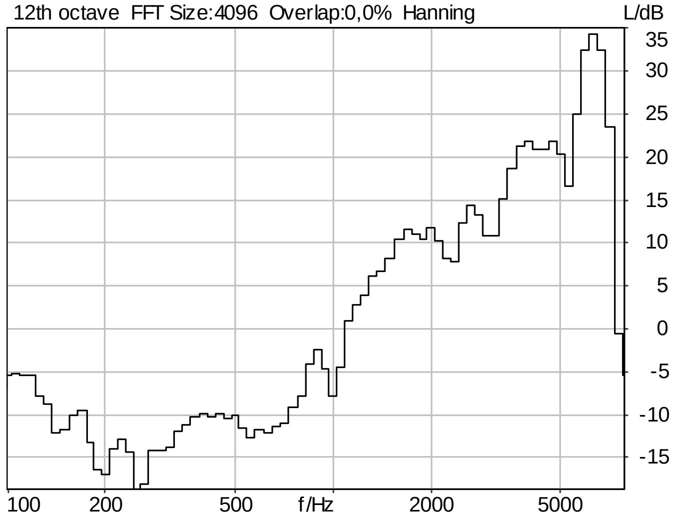

Figure A.2: ERL(f) for the wideband echopath simulation

In wideband the echo attenuation is calculated by calculating the level difference between electrical input and electrical output of the terminal. Since this calculation is based on speech and is dependent on the spectral energy distribution a CSS signal with speech like power density according to ITU-T Recommendation P.501 [17] is used. The spectral representation of this test signal in comparison to the spectra of English speech sentences as found in ITU-T Recommendation P.501 [17] is shown in Figure A.3.

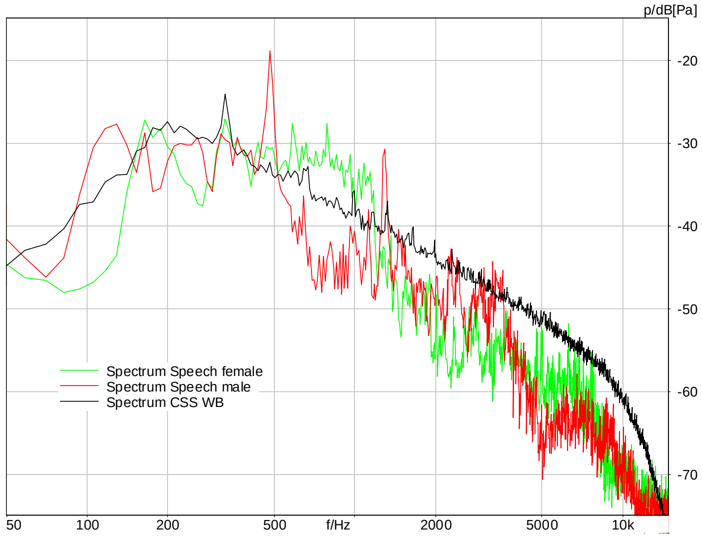

Figure A.3: CSS test signal (black) for determining the echo attenuation for the wideband echopath simulation

Annex B (normative): Test signal for immunity to DTMF false detection in send direction

The type of signal used is a multi speaker signal chosen for containing various frequency and critical for DTMF decoders. The signal is a recording of a polish theatre play used in a French standard for test of Telephone Answering Machines. The duration of the test signal is 38 minutes. For convenience the signal is divided into 94 parts of 25 s duration each, to be played in series.

The signal level has to be adjusted to -20 dBm0 RMS (which corresponds approximately to a speech level of -10 dBm0 when measured with a 20 s averaging).

The signal is applied to the input of MGW and production of DTMF codes is recorded.

The signal is part of the standard and can be downloaded from:

http://docbox.etsi.org/STQ/Open/ES%20202%20718%20Test%20signal/

Annex C (informative): Bibliography

ITU-T Recommendation P.56: "Objective measurement of active speech level".

ITU-T Recommendation P.64: "Determination of sensitivity/frequency characteristics of local telephone systems".

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

ITU-T Recommendation P.581: "Use of head and torso simulator (HATS) for hands-free terminal testing".

IEC 61260: "Electroacoustics - Octave-band and fractional-octave-band filters".

ETSI ES 202 739: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

ETSI TR 102 648-1: "Speech Processing, Transmission and Quality Aspects (STQ); Test Methodologies for ETSI Test Events and Results; Part 1: VoIP Speech Quality Testing".

ETSI EG 201 377-2: "Speech Processing, Transmission and Quality Aspects (STQ); Specification and measurement of speech transmission quality; Part 2: Mouth-to-ear speech transmission quality including terminals".

ETSI ES 201 235 (Parts 1 to 4): "Access and Terminals (AT); Specification of Dual-Tone Multi-Frequency (DTMF) Transmitters and Receivers".

History

Document history		
V1.1.1	July 2011	Membership Approval Procedure MV 20110927: 2011-07-29 to 2011-09-27