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Technical Report

Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points

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Contents

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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Speech and multimedia Transmission Quality (STQ).

Introduction

The present document is intended to fill a gap in a field where the industry has expressed an urgent need for standardized objectives. Based on the assumption that voice over IP services with the goal of users being satisfied or even very satisfied with the overall voice communication quality, the present document provides initial guidance on voice quality related parameters and respective objectives for interconnected networks.

For the time being the present document only covers fixed line IP access to the NGN core and simple Segment-connection scenarios.

Since further work is underway in this area, it is intended to update the present document in accordance with feedback from experience within the industry. The objective values given in the present document are provisional and may be revised.

The present document forms part of STQ's roadmap with respect to Quality aspects of NGN.

1 Scope

The present document provides guidance on the quality parameters that need to be considered at the Segment-connection of Voice over IP (VoIP) services and provides guidance on objectives for these parameters.

Inside the TISPAN NGN overall architecture (see figure 1), the present document cares only about the transport layer and its Segment-connection.

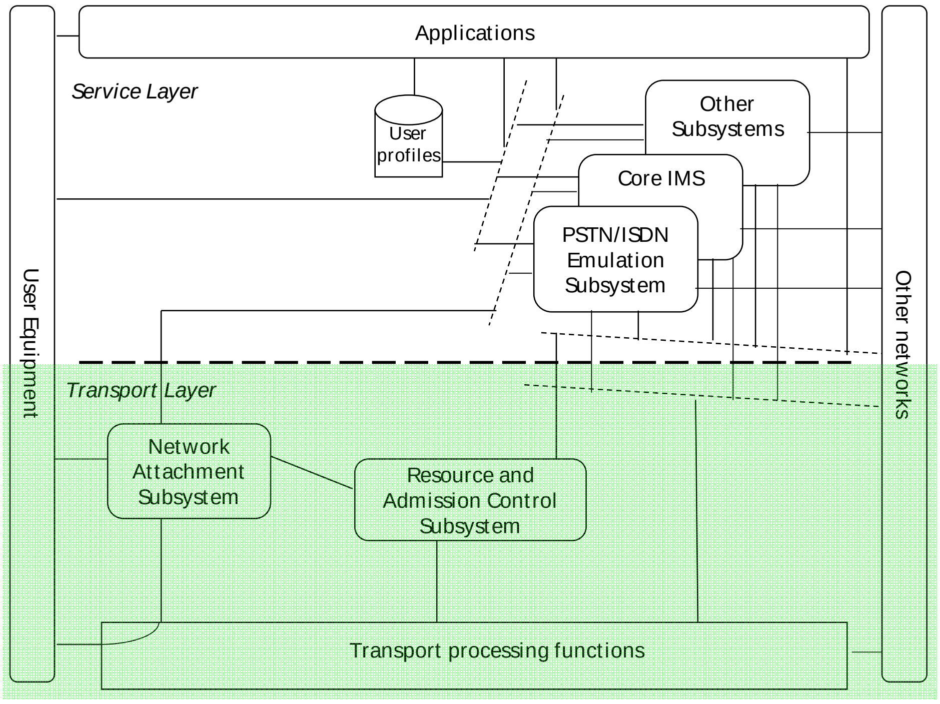

Figure 1: TISPAN NGN overall architecture (adopted from [i.14])

2 References

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2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

Not applicable.

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area**.** For non-specific references, the latest version of the referenced document (including any amendments) applies.

[i.1] ITU-T Recommendation Y.1540 (2002): "Internet protocol data communication service - IP packet transfer and availability performance parameters".

[i.2] ITU-T Recommendation Y.1541 (2006): "Network performance objectives for IP-based services".

[i.3] ITU-T Recommendation Y.1542 (2006): "Framework for achieving end-to-end IP performance objectives".

[i.4] ITU-T Recommendation G.107 (2008): "The E-model: a computational model for use in transmission planning".

[i.5] ITU-T Recommendation G.108 (1999): "Application of the E-model: A planning guide".

[i.6] ITU-T Recommendation G.109 (1999): "Definition of categories of speech transmission quality".

[i.7] ITU-T Recommendation G.113 (2007): "Transmission impairments due to speech processing".

[i.8] ITU-T Recommendation G.114 (2003): "One-way transmission time".

[i.9] ITU-T Recommendation G.1020 (2006): "Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks".

[i.10] ETSI ES 202 737: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

[i.11] ETSI ES 202 738: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for narrowband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user".

[i.12] ETSI ES 202 739: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".

[i.13] ETSI ES 202 740: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for wideband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user".

[i.14] ETSI ES 282 001 "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Functional Architecture".

[i.15] GSMA Document IR.34: "Inter-Service Provider IP Backbone Guidelines".

[i.16] ITU-T Recommendation G.8261 (2008): "Timing and synchronization aspects in packet networks".

[i.17] ITU-T Recommendation G.8262 (2007): "Timing characteristics of synchronous ethernet equipment slave clock (EEC)".

[i.18] ITU-T Recommendation G.8264 (2008): "Timing distribution through packet networks".

[i.19] IEEE 1588: "Standard for a Precision Clock Synchronization Protocol for Networked Measurement and Control System".

[i.20] ITU-T Recommendations of the P.862-series: "Perceptual evaluation of speech quality (PESQ): An objective method for end-to-end speech quality assessment of narrow-band telephone networks and speech codecs".

[i.21] ITU-T Recommendation P.834: "Methodology for the derivation of equipment impairment factors from instrumental models".

[i.22] ITU-T Recommendation G.711: "Pulse code modulation (PCM) of voice frequencies".

[i.23] ITU-T Recommendation G.726: "40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM)".

[i.24] ITU-T Recommendation G.727: "5-, 4-, 3- and 2-bit/sample embedded adaptive differential pulse code modulation (ADPCM)".

[i.25] ITU-T Recommendation G.728: "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

[i.26] ITU-T Recommendation G.729: "Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear prediction (CS-ACELP)".

[i.27] ITU-T Recommendation I.231.1: "Circuit-mode bearer service categories: Circuit-mode 64 kbit/s unrestricted, 8 kHz structured bearer service".

[i.28] ITU-T Recommendation G.826: "End-to-end error performance parameters and objectives for international, constant bit-rate digital paths and connections".

[i.29] ETSI ETS 300 726: "Digital cellular telecommunications system (Phase 2+) (GSM); Enhanced Full Rate (EFR) speech transcoding (GSM 06.60)".

3 Definitions and abbreviations

3.1 Definitions

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

access segment: network segment from the customer interface (UNI) to the interface on the customer side of the first Gateway Router

total transit segment: segment between Gateway routers, including the gateway routers themselves

NOTE: The network segment may include interior routers with various roles.

Segment-connection point: point between two segments

NOTE: The terms "interconnection" or "interconnection point" has been used in the NGN standards, e.g. in [i.14], the same terms are generally used for NNIs, not for the connection between access segment and transit segment, they might be misinterpreted. Therefore, throughout the present document, the terms "Segment-connection or "Segment-connection point" are used.

3.2 Abbreviations

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

AGW Access GateWay
ATM Abstract Test Method
DV Delay Variation
ESR Errored Second Ratio

GoB	Good or Better
GSMA	Global System for Mobile communications Association
GW	GateWay
IE	Equipment Impairment factor
IP	Internet Protocol
LAN	Local Area Network
MOS	Mean Opinion Score
MSAN	Multi Service Access Node
NGN	Next Generation Network
NNI	Network to Network Interface
NTP	Network Termination Point
PL	Packet Loss
PoW	Poor or Worse
PSTN	Public Switched Telephone Network
PTP	Point to Point
QoS	Quality of Service
SyncE	Synchronous Ethernet
UNI	User Network Interface
VoIP	Voice over IP

4 Reference Configuration

Compared to networks and systems that are circuit-based, those based on IP pose distinctly different challenges for planning and achieving the end-to-end performance levels necessary to adequately support the wide array of user applications (voice, data, fax, video, etc.). The fundamental quality objectives for these applications are well understood and have not changed as perceived by the user; what has changed is the technology (and associated impairments) in the layers below these applications. The very nature of IP-based routers and terminals, with their queuing methods and de-jitter buffers, respectively, makes realizing good end-to-end performance across multiple network operators a very major challenge for applications with stringent performance objectives. Fortunately ITU-T Recommendations Y.1540 [i.1] and Y.1541 [i.2] together provide the parameters needed to capture the performance of IP networks, and specify a set of "network QoS" classes with end-to-end objectives specified. It is widely accepted (i.e. beyond the ITU-T) that the network QoS classes of ITU-T Recommendation Y.1541 [i.2] should be supported by Next Generation Networks, and thus by networks evolving into NGNs. ITU-T Recommendation Y.1542 [i.3] considers various approaches toward achieving end-to-end (UNI-UNI) IP network performance objectives.

The general reference configuration for the present document follows the principles shown in figure 2; the number of concatenated transit providers may vary.

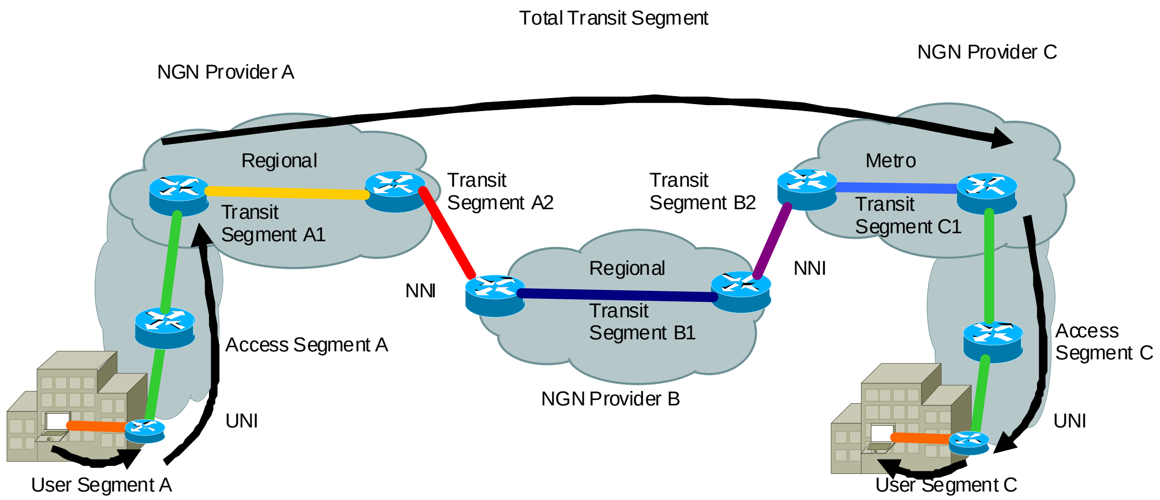

Figure 2: General Reference Configuration

Thus the end-to-end connection can be decomposed into the following elements:

- User segment A.
- UNI_A (sending side).
- Access segment A.
- Segment-connection Point A_{in}.
- Total transit segment.
- Segment-connection Point C_{out}.
- Access segment C.
- UNI_C (receiving side).
- User segment C.

The total transit segment can be further decomposed into:

- Transit segment A1.
- Segment-connection point A_{out}.
- Transit segment A2 (NNI).
- Segment-connection point B_{in}.
- Transit segment B1.
- Segment-connection point B_{out}.
- Transmit segment B2 (NNI)
- Segment-connection point C_{in}.
- Transit segment C1.

4.1 Generic Segment-connection Points

Due to real-world constraints the simplified **static divisor** approach according to ITU-T Recommendation Y.1542 [i.3] has been chosen for the impairment apportionment between access and transit networks.

This approach "divides" the UNI-to-UNI path into three segments and budgets the impairments such that the total objective is met in principle.

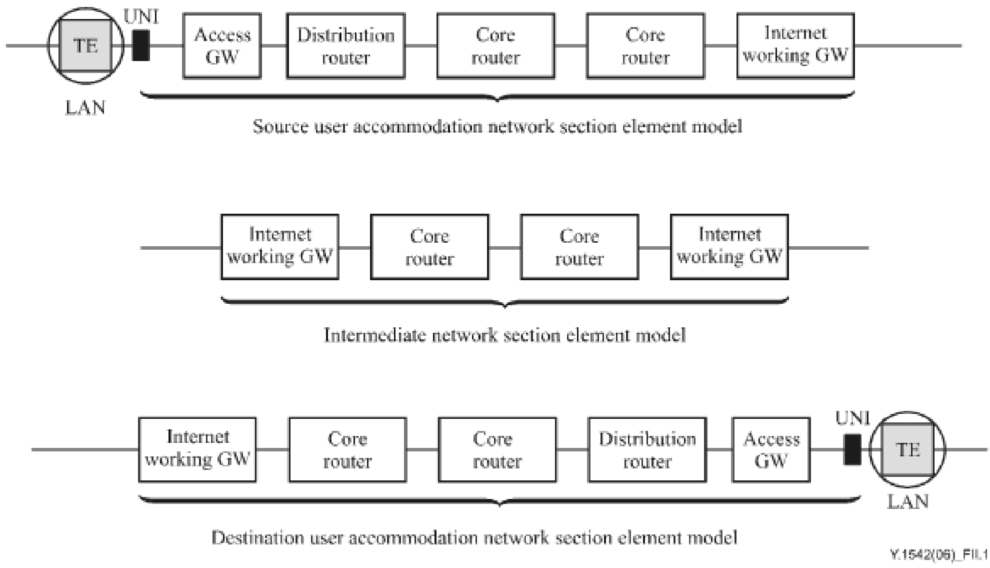

Figure 3: Division of the connection

Hence, there should be objectives for the following portions of the connection:

- UNI (send side) <<----->> Segment-connection Point A.
- Segment-connection Point A <<----->> Segment-connection Point C.
- Segment-connection Point C <<----->> UNI (receive side).

The guidance on respective objectives is given in clause 6.

4.2 Segment-connection Points within the Total Transit Segment

For Segment-connection points within the total transit segment, e.g. Segment-connection points A_{out},B_{in}, B_{out},C_{in} the objectives are for further study. However it is anticipated that service providers may implement any of the methods outlined in ITU-T Recommendation Y.1542 [i.3] in order to meet jointly the objectives in clause 6.2 of the present document.

4.3 Voice Terminals

In order to be able to achieve the goal of users being satisfied or even very satisfied with the overall voice communication quality it is assumed that the VoIP terminals used in this context comply with one or more of the following ETSI standards:

- ES 202 737 [i.10]: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".
- ES 202 738 [i.11]: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for narrowband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user".
- ES 202 739 [i.12]: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user".
- ES 202 740 [i.13]: "Speech Processing, Transmission and Quality Aspects (STQ); Transmission requirements for wideband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user".

5 Segment-connection Voice Quality Parameters

The following parameters should to be considered at any transit segment and Segment-connection point with respect to voice communication services.

5.1 Delay

The objectives for delay should be structured in a way which enables both:

- Transmission planning.
- Verification in the field.

Therefore, the following approach should be considered:

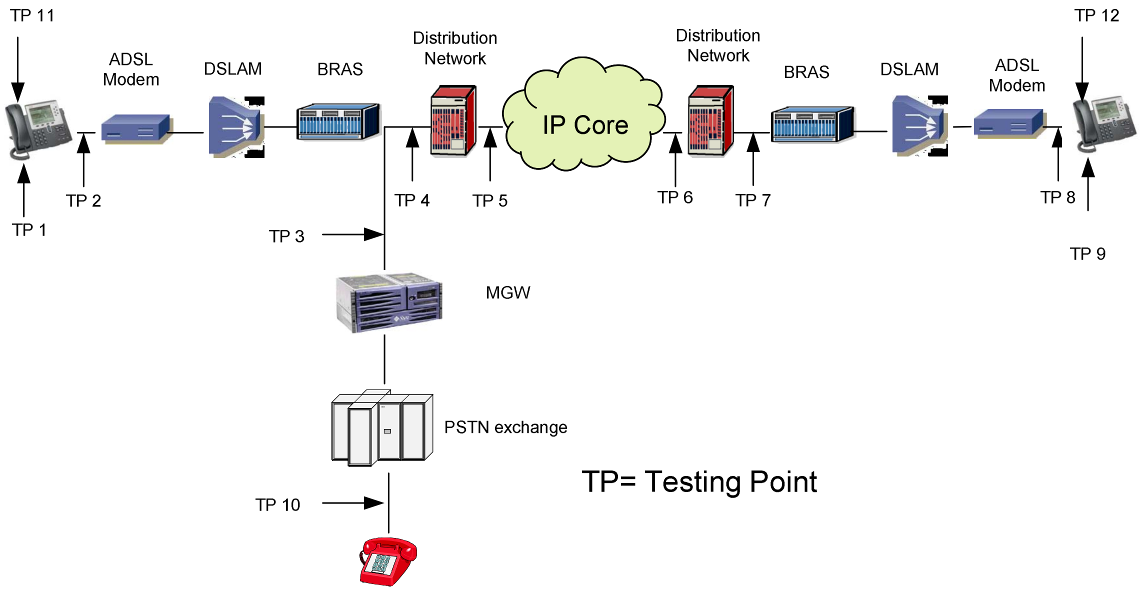

Figure 4: Example reference points for delay definition and verification in the field

- Delay from TP1 to TP2 plus delay from TP8 to TP12 = Speech processing related Delay in an IP environment (T_{SP}), see clause 5.1.1.
- Delay from TP2 to TP4 = Access Delay (T_A), see clause 5.1.2.
- Delay from TP7 to TP8 = Access Delay (T_A), see clause 5.1.2.
- Delay from TP4 to TP7 = Transit Delay (T_T), see clause 5.1.3.

5.1.1 Speech processing related Delay in an IP environment (T_{SP})

Delay related to speech processing in an IP environment can be calculated according to annex 1 of ITU-T Recommendation G.114 [i.8]. It is the sum of the amount of delay incurred by encoding and packetization at the send side, average de-jitter buffer size (i.e. half of the de-jitter buffer size) at the receive side and de-packetization and decoding at the receive side.

5.1.2 Access delay (T_A)

Access delay (T_A) includes Serialization and the Queueing time in the access defined in ITU-T Recommendation Y.1541 [i.2].

5.1.3 Transit Delay (T_T)

Transit delay (T_T) is part of IPTD according to ITU-T Recommendation Y.1541 [i.2]. In summary:

$$\text{IPTD (Y.1541)} = T_A \text{ (transmit side)} + T_T + T_A \text{ (receive side)}$$

5.2 Delay Variation (DV)

Delay Variation is used as defined in ITU-T Recommendations Y.1540 [i.1] and Y.1541 [i.2].

5.3 Packet Loss (PL)

Packet Loss is used as defined in ITU-T Recommendations Y.1540 [i.1] and Y.1541 [i.2].

5.4 Equipment Impairment Factor (Ie)

Equipment Impairment Factor (Ie) as tabulated in ITU-T Recommendation G.113 [i.7]; for access networks the codec used in the terminal has to be considered; since the Ie value applies for the entire process of encoding plus decoding, the Ie value from ITU-T Recommendation G.113 [i.7] has to be assigned always to the encoding side.

6 Guidance on Segment-connection Voice Quality Objectives

The objectives proposed in the present document are based on transmission planning aspects as outlined in ITU-T Recommendation G.107 [i.4] (The E-model) and its companion documents ITU-T Recommendations G.108 [i.5] and G.109 [i.6]. For the purposes of verification of these objectives, ITU-T Recommendations of the P.862- series [i.20] and eventually ITU-T Recommendation P.834 [i.21] should be consulted. For the calculation according to G.107 all input parameters excluding the delay and Ie related values are set to default values according to ITU-T Recommendation G.107 [i.4]. This means, that the R-Values reached with different delay and Ie values are under optimal conditions, any deviation from default values for the other parameters will most probably decrease the quality.

The overall aim of the Segment-connection voice quality objectives is to enable network operators, service providers and indirectly also equipment manufacturers to provide end-to-end voice quality with which users are satisfied or even very satisfied. In order to achieve this goal the simplified approach here is, to limit end-to-end delay to 150 ms, except for cases where this is not feasible due geographical constraints; Also the accumulated sum across the entire connection should not exceed Ie=12.

Annex A provides a summary of elsewhere published data that will proof useful in the context of the present document.

There may be other connections with higher or different impairment which still leave the users satisfied in accordance with the E-model or other relevant experience; however, in many cases this is currently not under the control of network providers when providing interconnection.

Therefore, the following guidance on objectives is given as a state-of-the-art reply to the present demand of the industry.

6.1 Guidance on Access Segment Objectives

The following objectives can be applied between the following points, it should be noted that these parameters may vary between both directions of transmission:

- UNI_A (sending side) ----->> Segment-connection point A (receiving side);
- Segment-connection point A (sending side) ----->> UNI_C (receiving side);

- UNI_A (sending side) ----->> Segment-connection point C (receiving side); and
- Segment-connection point C (sending side) ----->> UNI_C (receiving side).

See figure 1 for details. The categories in table 1 refer to ITU-T Recommendation G.109 [i.6] with the following notations:

- BEST ===>> Users are very satisfied with voice communication quality (R > 90).
- HIGH ===>> Users are very satisfied with voice communication quality (R > 80).

Table 1: Guidance on objectives for either Access Segment

Category	BEST (G.109)	HIGH (G.109)
(T_{SP}) / ms	60	90
(T_A) / ms	20	30
DV / ms (see note 2)	20	20
IPLR	$3{,}0 \times 10^{-4}$	$3{,}0 \times 10^{-4}$
IPER	3×10^{-5}	3×10^{-5}
Ie (transmit)	2	12
Ie (receive)	0	0
NOTE 1: The values defined here are the upper bound of the mean values. NOTE 2: The de-jitter delay is included in T_{SP}.		

The proposed Ie objectives allow e.g. for ITU-T Recommendations G.711 [i.22] or G.726 [i.23] (40 kbit/s) in the BEST category and for G.726 [i.23] (32 kbit/s) or G.728 [i.25] (16 kbit/s) or G.729 [i.26] (8 kbit/s) in the HIGH category.

Even though in reality the encoding takes place before the UNI (and the decoding takes place behind the UNI) the coding process is taken into account as part of the access segment, because the bit stream present at the UNI is coded.

In cases where one entity has control of both access segments (local and remote), for example the BEST (G.109) category could be achieved by meeting the following objective: T_A (local) + T_{SP} + T_A (remote) < 100 ms.

6.2 Guidance on Total Transit Segment Objectives

The following objectives can be applied between:

- Segment-connection point A <<----->> Segment-connection point C.

See figure 1 for details. The objectives are based on the application of Class 0 of ITU-T Recommendation Y.1541 [i.2]. The determination of cases where Class 1 of ITU-T Recommendation Y.1541 [i.2] should be applied and the associated objectives are for further study.

The categories in table 2 refer to ITU-T Recommendation G.109 [i.6] with the following notations:

- BEST ===>> Users are very satisfied with voice communication quality (R > 90).
- HIGH ===>> Users are very satisfied with voice communication quality (R > 80).

Table 2: Guidance on objectives for Total Transit Segments

Category	BEST (G.109)	HIGH (G.109)
(T_T) / ms	50	50
DV / ms (see note 2)	10	10
IPLR	$3{,}0 \times 10^{-4}$	$3{,}0 \times 10^{-4}$
IPER	3×10^{-5}	3×10^{-5}
Ie	0	0
NOTE 1: The values defined here are the upper bound of the mean values. NOTE 2: The de-jitter delay is included in T_{SP}.		

The proposed transit delay value applies to total transit segments which are intra-continental, only. For total transit segments which are intercontinental 140 ms may be appropriate, see table I.2 of ITU-T Recommendation Y.1542 [i.3], the proposed objectives for the present document is for further study.

It is assumed that transcoding in the total transit segment can be avoided at all.

Transit delay includes the core and distribution delay as well as the propagation delay defined in ITU-T Recommendation Y.1541 [i.2].

6.3 End-to-End Aspects

Figures 5 and 6 depict a summary of the proposed delay objectives and the end-to-end delay targets that can be achieved.

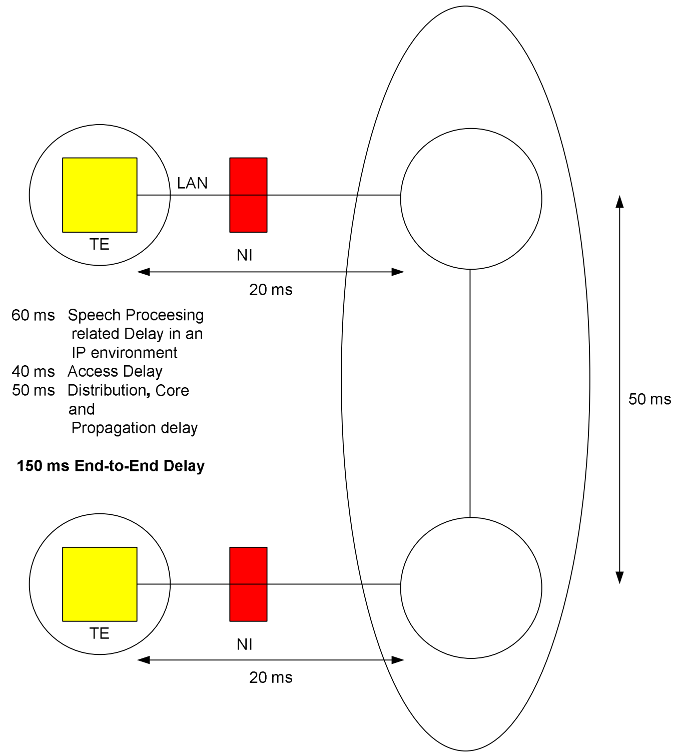

Figure 5: Delay Objectives for BEST (G.109) voice communication quality (R > 90)

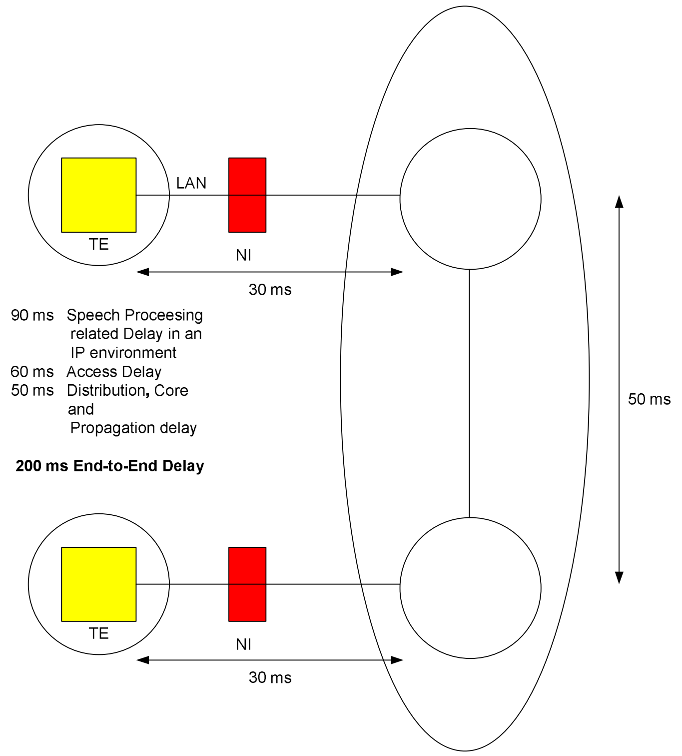

Figure 6: Delay Objectives for HIGH (G.109) voice communication quality (R > 80)

7 Possible Implications due to other services

Even though the present document is focussed on VoIP, it may be worthwhile to consider - at the time of deployment - implications that may arise due to other services which are likely to be carried over the same infrastructure. The following may serve as one example, with its provisional objectives being derived from past and current implementations in traditional networks.

EXAMPLE: The IP-based network should also be capable to carry the 64 kbit/s transparent data service described in ITU-T Recommendation I.231.1 [i.27], also known as "64 k clear-mode". The basis of the objective here is use of ITU-T Recommendation G.826 [i.28], a standard for synchronous digital networks. While the IP core is a packet network and not a synchronous network, it is being used to emulate a service currently transported over a synchronous network. Hence the performance of the emulation should be no worse than the performance of the synchronous network as specified by ITU-T Recommendation G.826 [i.28]. The standard requires an Errored Second Ratio (ESR) of < 0,16 for an STM-1 link which can carry about 1 200 "clear-mode" channels. From this, the end-to-end probability of loss per packet can be shown to be about $1,5 \times 10^{-6}$. ITU-T Recommendation G.826 [i.28] budgets 18,5 % of this to each national network, so the packet loss for a national connection should be no more than $2,75 \times 10^{-7}$. Allocation of this ratio to individual operators' networks within the national network is yet to be agreed, but it is fairly unlikely that there will be more than three operators' switched networks between any customer and the international gateway, so an initial allocation could be $9,0 \times 10^{-8}$ to each operator's network.

Table 3

Summary of provisional objectives	
Parameter	**Provisional Objective**
IP packet loss ratio for national connections	$2,75 \times 10^{-7}$
IP packet loss ratio for each operator's network	$9,0 \times 10^{-8}$
end-to-end probability IP packet loss ratio	$1,5 \times 10^{-6}$
IP packet error ratio for each operator's network	$1,0 \times 10^{-8}$

8 Synchronization of endpoints

To ensure the synchronization of the endpoints (e.g. MSAN, GW; AGW) the endpoints should be synchronized with Synchronous Ethernet (SyncE) based on the ITU-T Recommendations G.8261 [i.16], G.8262 [i.17] and G.8264 [i.18]. Additionally, PTP (IEEE 1588 v2 [i.19]) and NTPv4 may be used as mean for synchronization of endpoints.

Annex A: Summary of Relevant Transmission Planning Data

This annex provides condensed information on transmission planning data that may be considered useful in the context of the present document.

A.1 Delay in VoIP Terminals

The following information is an excerpt from ES 202 737 [i.10], ES 202 738 [i.11], ES 202 739 [i.12] and ES 202 740 [i.13].

A.1.1 Send delay

For a VoIP terminal, send delay is defined as the one-way delay from the acoustical input (mouthpiece) of this VoIP terminal to its interface to the packet based network. The total send delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figures 2 and A.1 in ITU-T Recommendation G.1020 [i.9], respectively.

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

$$T(s) = T(ps) + T(la) + T(aif) + T(asp)$$

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 will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE: 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. Table A.1 provides examples for delay values calculated accordingly.

Table A.1

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.711 [i.22]	80	0,125	10	0	0	10	< 20
G.711 [i.22]	160	0,125	20	0	0	10	< 30

A.1.2 Receive delay

For a VoIP terminal, receive delay is defined as the one-way delay from the interface to the packet based network of this VoIP terminal to its acoustical output (earpiece). The total receive delay is the upper bound on the mean delay and takes into account the delay contributions of all of the elements shown in figures 3 and A.2 of ITU-T Recommendation G.1020 [i.9], respectively.

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

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

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 disassembly 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 will usually be quite small. For the purposes of the present document it is assumed that in the test setup this delay can be neglected.

NOTE: 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. Table A.2 provides examples calculated accordingly.

Table A.2

Codec	N	T(fs) in ms	T(aif) in ms	T(jb) in ms	T(plc) in ms	T(asp) in ms	T(r) Requirement in ms
G.711 [i.22]	80	0,125	0	10	10	10	< 30,125
G.711 [i.22]	80	0,125	0	10	0	10	< 20,125
G.711 [i.22]	160	0,125	0	10	10	10	< 30,125
NOTE 1: T(ps) = packet size = N * T(fs). NOTE 2: N = number of frames per packet.							

A.2 Impairment Factors of Codecs

The following data is an excerpt from annex I to ITU-T Recommendation G.113 [i.7].

Table A.3 provides provisional planning values for the equipment impairment factor *Ie* of some codecs which are relevant in the context of the present document. These Ie values refer to non-error conditions without propagation errors, frame-erasures or packet loss.

Table A.3: Provisional planning values for the equipment impairment factor *Ie*

Codec type	Reference	Operating rate kbit/s	*Ie* value
PCM	G.711 [i.22]	64	0
ADPCM	G.726 [i.23], G.727 [i.24]	40	2
	G.721 (1988), G.726 [i.23], G.727 [i.24]	32	7
LD-CELP	G.728 [i.25]	16	7
CS-ACELP	G.729 [i.26]	8	10
	G.729-A + VAD	8	11
ACELP	ETS 300 726 (GSM 06.60) [i.29], Enhanced Full Rate	12,2	5

A.3 Network QoS Classes for Voice Applications

The following information is an excerpt from ITU-T Recommendation Y.1541 [i.2].

Table A.4: Provisional IP network QoS class definitions and network performance objectives

		QoS Classes					
Network Performance Parameter	**Nature of Network Performance Objective**	**Class 0**	**Class 1**	**Class 2**	**Class 3**	**Class 4**	**Class 5 Un-specified**
IPTD	Upper bound on the mean IPTD	100 ms	400 ms	**Not relevant for voice communication!**			
IPDV	Upper bound on the $1-10^{-3}$ quantile of IPTD minus the minimum IPTD	50 ms	50 ms				
IPLR	Upper bound on the packet loss probability	1×10^{-3}	1×10^{-3}				
IPER	Upper bound	1×10^{-4}					

A.4 Categories of User Satisfaction

The following information is an excerpt from ITU-T Recommendation G.109 [i.6].

While the single parameters describe the individual factors affecting speech transmission quality, it is the combined effect of all parameters together which leads to the overall level of speech transmission quality as perceived by the user. For transmission planning purposes, the E-model (G.107) is a useful tool for assessing the combined effect of all parameters and hence differentiating between categories of speech transmission quality.

The primary output of the E-model is the Transmission Rating Factor, R. Table A.5 gives the definitions of the categories of speech transmission quality in terms of ranges of Transmission Rating Factor R provided by ITU-T Recommendation G.107 [i.4]. Also provided are descriptions of "User satisfaction" for each category.

Table A.5: Definition of categories of speech transmission quality

R-value range	Speech transmission quality category	User satisfaction
$90 \leq R < 100$	Best	Very satisfied
$80 \leq R < 90$	High	Satisfied
$70 \leq R < 80$	Medium	Some users dissatisfied
$60 \leq R < 70$	Low	Many users dissatisfied
$50 \leq R < 60$	Poor	Nearly all users dissatisfied
NOTE 1: Connections with R-values below 50 are not recommended. NOTE 2: Although the trend in transmission planning is to use R-values, equations to convert R-values into other metrics e.g. MOS, % GoB, % PoW, can be found in ITU-T Recommendation G.107 [i.4], annex B.		

A.5 Comparison of Codecs, Link Speed and Capacity examples

	G.711			G.726				G.729A	
Codec Bit Rate (kb/s)	64	64	64	32	32	40	40	8	8
Ie	0			7		2		11	
Packet Frame Duration (ms)	10	20	30	20	30	20	30	20	30
Payload Size (bytes)	80	160	240	80 (320 bit/8 × 2)	120 (320 bit/8 × 3)	100 (400 bit/8 × 2)	150 (400 bit/8 × 3)	20	30
IP Packet Size (40 b header)	120	200	280	120	160	140	190	60	70
IP Bitrate needed (kbit/s)	96	80	74	48	42,6	56	51	24	19
ATM Cells Needed IP Packet Size / 47	3	5	6	3	4	3	5	2	2
ATM Bytes Needed (ATM cells × 53) + 8 Byte Trailer	167	273	326	167	220	167	273	114	114
ATM Bitrate Needed (kb/s) / channel (ATM Bytes × 8 / duration)	133	109	87	67	59	67	73	46	30
Delay in IP environment (ms) (2N+1) × frame size + Look ahead Where: N = number of frames per packet; frame size is in ms	30	60	90	60	90	60	90	100	150
MOS without network delay	4,4	4,4	4,4	4,2	4,2	4,3	4,3	4.0	4.0
End to end delay with 40 ms network delay e.g. IAD -> IAD	70	100	130	100	130	100	130	140	190
MOS	4,4	4,4	4,3	4,2	4,1	4,3	4,3	4,0	3,9
QoS two wire (PSTN)	4,2	4,2	4,1	4,0	3,9	4,1	4,1	3,8	3,7
End to end delay with 20 ms network delay e.g. TDM -> IAD	50	80	110	80	110	80	110	120	170
MOS	4,4	4,4	4,4	4,2	4,2	4,3	4,3	4,0	3,9
QoS two wire (PSTN)	4,2	4,2	4,2	4,0	4,0	4,1	4,1	3,8	3,7
NOTE: The values for MOS in this table are rounded to one decimal.									

MOS Calculation see: http://www.itu.int/ITU-T/studygroups/com12/emodelv1/calcul.php

A.6 Roundtrip Delay (2 T_T) for conversational and streaming traffic classes recommended by GSMA (Document IR 34)

EF & AF4	Middle-Europe	North-Europe	East Europe	South Europe	East Asia	South-East Asia	Oceania	N America (East Coast)	N America (West Coast)	Central America	S America	Africa
Middle-Europe	55	45	80	72	340	360	380	120	200	225	330	242
North-Europe	45	40	35	75	350	360	400	130	215	249	335	269
East Europe	80	35	40	102	360	370	420	165	215	281	350	262
South Europe	72	75	102	72	345	355	380	145	220	247	335	218
East Asia	340	350	360	345	150	165	275	340	285	353	460	383
South-East Asia	360	360	370	355	165	145	255	360	310	489	480	251
Oceania	380	400	420	380	275	255	90	360	310	369	470	287
N America (East Coast)	120	130	165	145	340	360	360	40	90	92	280	326
N America (West Coast)	200	215	215	220	285	310	310	90	40	126	300	418
Central America	225	249	281	247	353	489	369	92	126	40	137	294
S America	330	335	350	335	460	480	470	280	300	137	120	180
Africa	242	269	262	218	383	251	287	326	418	294	180	180

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