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Technical Report

Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points; A support to NGN transmission planners

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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.

This revision adds more details of delay introduced by network elements, jitter caused by access bandwidth limitations and on reference connection scenarios. This is intended as support to NGN transmission planners.

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 considers only the transport layer.

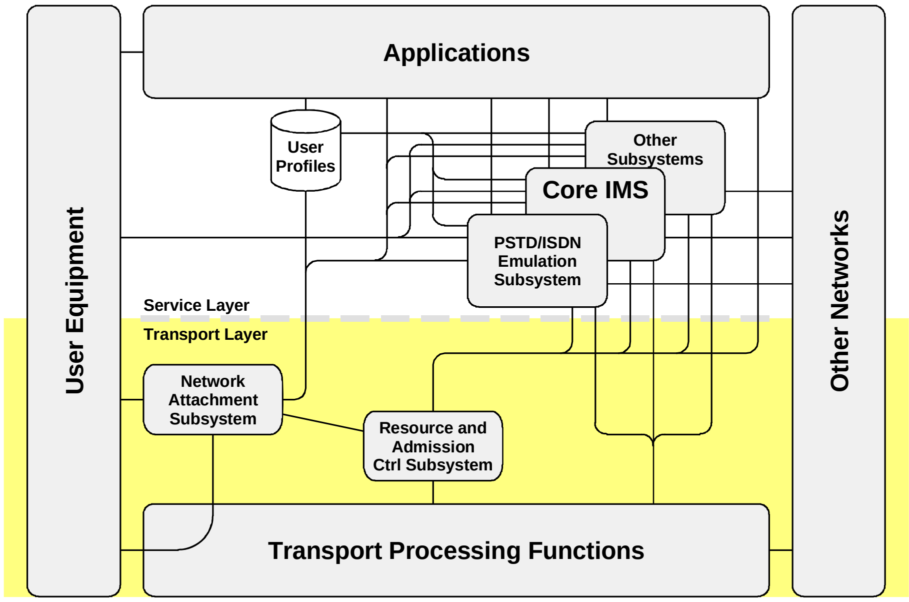

Figure 1: TISPAN NGN overall architecture (adapted from [i.14])

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
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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] Void.

[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 and multimedia Transmission Quality (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 and multimedia Transmission Quality (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 and multimedia Transmission Quality (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 and multimedia Transmission Quality (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.3445: "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] ITU-T Recommendation Q.115.1: "Logic for the control of echo control devices and functions".

[i.30] ETSI EN 300 726: "Digital cellular telecommunications system (Phase 2+) (GSM); Enhanced Full Rate (EFR) speech transcoding (GSM 06.60 version 8.0.1 Release 1999)".

[i.31] IETF RFC 1483: "Multiprotocol Encapsulation over ATM Adaptation Layer".

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

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.

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

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

3.2 Abbreviations

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

ACELP	Algebraic Code-Excited Linear Prediction
ADM	Add-Dropp-Multiplexer
ADPCM	Adaptive Differential Pulse Code Modulation
ADSL	Asymmetric Digital Subscriber Line
AGW	Access GateWay
ATM	Asynchronous Transfer Mode
BRAS	Broadband Remote Access Server
BSC	Base Station Controller
BTS	Base Transceiver Station
CL	Router Core Layer
CS-ACELP	Conjugate Structure Algebraic Code- Excited Linear Prediction
DL	Router Distribution Layer
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
DV	Delay Variation
EC	Echo Canceller
ESR	Errored Second Ratio
ETH	Ethernet
GoB	Good or Better
GSM	Global System for Mobile communications
GSMA	Global System for Mobile communications Association
GW	GateWay
IAD	Integrated Access Device
Ie	Equipment Impairment Factor
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPDV	IP packet Delay Variation
IPER	IP Packet Error Ratio
IPLR	IP Packet Loss Ratio
IPTD	IP Packet Transfer Delay
ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
ITU-T	ITU Telecommunication Standardization Sector
JB	De-jitter Buffer
LAN	Local Area Network
MGW	Media Gateway
MOS	Mean Opinion Score
MP-ACELP	Multipulse Algebraic code excited linear prediction
MP-MLQ	Multipulse Maximum Likelihood Quantization
MSAN	Multi Service Access Node
MTU	Maximum Transmission Unit
NGN	Next Generation Network
NI	Network Interface
NNI	Network to Network Interface
NTP	Network Termination Point
PDH	Plesiochronous Digital Hierarchy
PL	Packet Loss
PoW	Poor or Worse
PSTN	Public Switched Telephone Network
PTP	Point to Point
QoS	Quality of Service
SBC	Session Border Controller
SoIx	Service-oriented Interconnection
STM 1	Synchronous Transport Module 1
SyncE	Synchronous Ethernet
TRAU	Transcoder and Rate Adaption Unit
UMSC	UMTS Mobile Switching Center

UMTS	Universal Mobile Telecommunications System
UNI	User Network Interface
VoIP	Voice over Internet Protocol
WiMAX	Worldwide Interoperability for Microwave Access
xDSL	x Digital Subscriber Line

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-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.

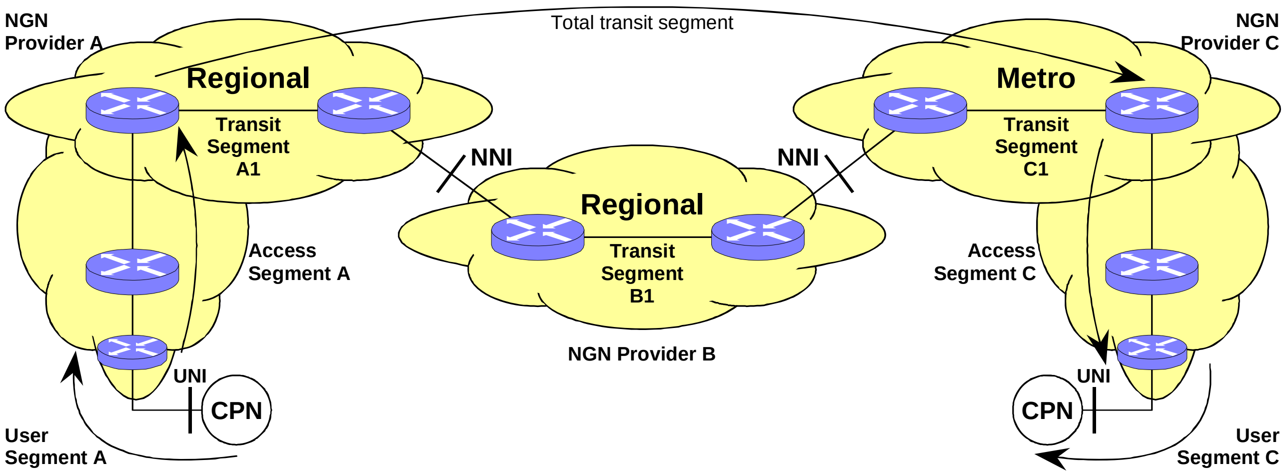

Figure 2: General Reference Configuration

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

- User segment A.
- UNIA (sending side).
- Access segment A.
- Segment-connection Point Ain.
- Total transit segment.
- Segment-connection Point Cout.
- Access segment C.
- UNIC (receiving side).
- User segment C.

The total transit segment can be further decomposed into:

- Transit segment A1.
- Segment-connection point Aout.
- Transit segment A2 (NNI).
- Segment-connection point Bin.
- Transit segment B1.
- Segment-connection point Bout.
- Transmit segment B2 (NNI).
- Segment-connection point Cin.
- 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.

As outlined in [i.15] the delay values for the total transit segment are in a fixed relation to the distances between different geographical regions (see table 2). Thus, for the near future dynamic allocation of delay budgets is not expected to be implemented between user segments, access segments and transit segments.

In figure 3, the upper part displays the division of the connection as seen from a QoS point of view whereas the lower part shows this division in terms of the NGN Functional Architecture [i.14].

NOTE: The reference points Ic, Iw, and Iz are defined in [i.14] in clause 7.2.2.

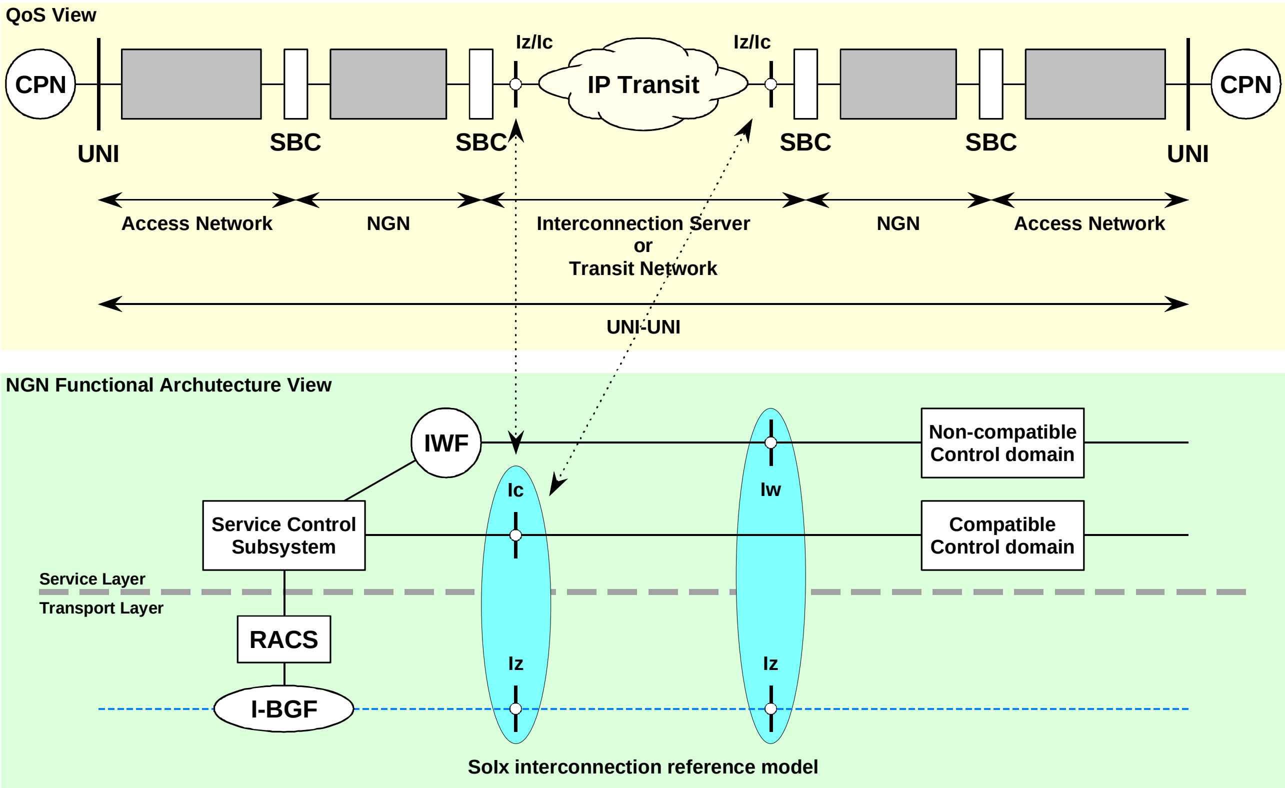

Figure 3: Division of the connection

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 5.

As illustrated in figure 3, SoIx interconnection is typically characterized by the presence of two types of information exchanged between the two interconnected domains:

- Service-related signalling information, that allows to identify the end-to-end service that has been requested. For example, in case of IMS-to-IMS SoIx interconnection, this is mapped to SIP signalling on the Ic reference point.
- Transport information, that carries the bearer traffic.

The presence of the service-related signalling in SoIx interconnection enables the end-to-end service awareness.

An NGN interconnection could be a SoIx even if the transport information is not exchanged between the interconnected domains, as long as service-related signalling is exchanged.

An NGN transport layer interconnection is considered being part of an NGN SoIx interconnection if the transport layer is controlled from the service layer in both of the interconnected domains.

- **SoIx Interconnection interface** includes at least Ic and Iz reference points between two interconnected domains that have same or compatible service control sub systems/domains.
- **SoIx Interconnection interface with Interworking** includes at least the Iw and Iz reference points between two interconnected domains that have non- compatible service control sub systems/domains.

4.2 Transport Reference Parameters and Configurations

At the Segment-connection Points (figure 3) different access networks can be connected. Following access networks can be considered:

- PSTN/ISDN classic access Configuration.
- NGN PSTN/ISDN access Configuration.
- Access DSL Configuration.
- WiMAX.
- GSM.
- UMTS.

The Access Points for WiMAX GSM and UMTS are for further study. In the following clauses are defined the end-to-end delay, and the Talker Echo Loudness Rating The detailed values of jitter and delay are for the access are described in clause 6.

4.2.1 Reference Configurations

The following clauses describes the Backbone and access reference configuration. In the calculation is at the Segment-connection point taken into account only one SBC.

4.2.1.1 Backbone Configuration

Figure 4 shows the backbone configuration. The number of elements used in the configuration and the delay values are is described in clause 6.

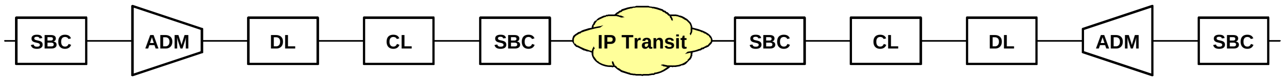

Figure 4: Backbone

4.2.1.2 PSTN/ISDN classic access Configuration

Figure 5 shows the PSTN/ISDN classic access configuration. The number of elements used in the configuration and the delay values are is described in clause 6.

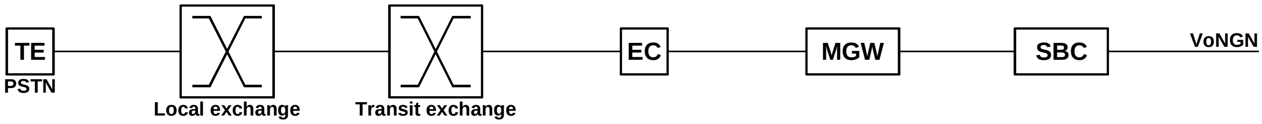

Figure 5: Reference configuration for PSTN/ISDN with classical access

4.2.1.3 NGN PSTN/ISDN access Configuration

Figure 6 shows the NGN PSTN/ISDN classic access configuration. The number of elements used in the configuration and the delay values are is described in clause 6.

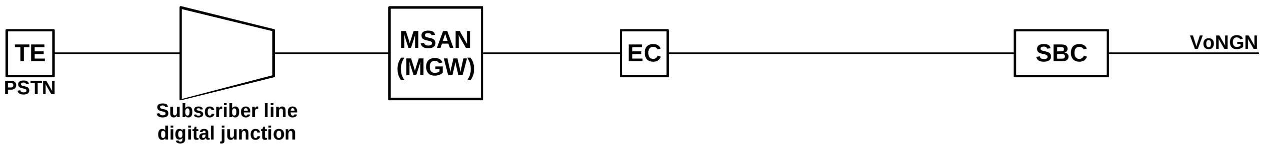

Figure 6: Reference configuration for NGN with PSTN/ISDN access

4.2.1.4 Access DSL Configuration

Figure 7 shows the xDSL access configuration. The number of elements used in the configuration and the delay values are is described in clause 6.

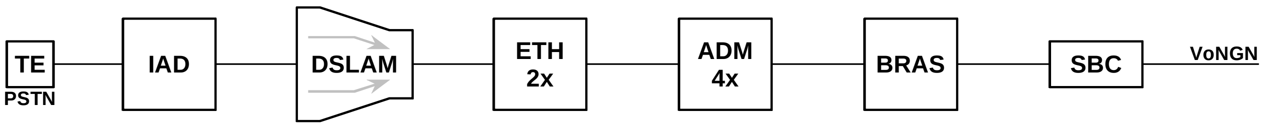

Figure 7: Reference configuration for DSL access

4.3 Delay Values

4.3.1 Backbone Delay

Table 1 shows the long distance delay values for typical reference distances.

Table 1: Long Distance Delay

Distance	Delay
1 400 km	11 ms
5 000 km (Intra Regional)	29 ms
10 000 km (Inter Regional)	54 ms
27 500 km (Inter Regional)	142 ms
NOTE: Delay values see also table 2.	

Table 2 shows delay values between originating and terminating Service Provider premises. The End-to-End delay values are based on values contained in the GSMA document IR3445.

Table 2: End-to-End delay values between originating and terminating Service Provider premises

EF& AF-4	Middle Europe	North Europe	East Europe	South Europe	East Asia	South East Asia	Oceania	N.America East Cost	N.America West Cost	Central America	South America	Africa
Middle Europe	28	23	40	36	170	180	190	60	100	113	165	121
North Europe	23	20	18	38	175	180	200	65	108	125	168	135
East Europe	40	18	20	51	180	185	220	83	108	141	175	131
South Europe	36	37	51	36	173	178	190	73	110	124	168	109
East Asia	170	175	180	173	75	83	138	170	143	177	230	192
South East Asia	180	180	185	178	83	63	128	180	155	245	240	126
Oceania	190	200	210	190	138	128	45	180	155	185	235	144
N.America E. Cost	60	65	83	73	170	180	180	20	45	46	140	163
N.America W. Cost	100	108	108	110	143	155	155	45	20	63	150	209
Central America	113	125	141	124	178	245	185	46	123	20	67	147
South America	165	168	175	168	230	240	235	240	150	67	60	140
Africa	121	135	131	109	192	125	144	163	209	147	90	90
NOTE: See [i.15].												

4.4 Network parameters: End-to-End Delay, Talker Echo Loudness Rating, R Value

In this clause, end to end delay values (mouth to ear) for different access lines and the respective R-values (depending on the calculated delay) are shown.

The following clause describes the Network parameters: End-to-End Delay, Talker Echo Loudness Rating for a national network. The detailed values of End-to-End Delay, and Jitter values and the time values for the jitter buffers are described in clause 6.

4.4.1 Delay with regional propagation delay (1 400 km / 11 ms)

For the calculation of the Voice Quality parameters used network parameters are contained in clause 6. For the calculation is used the Packet size of 20 ms, the access of DSL line 128 kbit/s uplink; 128 kbit/s downlink, DSL line 256 kbit/s uplink; 256 kbit/s downlink, DSL line 384 kbit/s uplink; 1 024 kbit/s downlink. The codecs G.729A, G.711 and G.726/40/20.

The delay values of the used components are state of the art. The R values are based on ISDN Terminals with the Talker Echo Loudness Rating TELR = 65. For DECT terminals the Talker Echo Loudness Rating TELR = 65 is used under the condition that the echo cancellation is deployed in the gateway according ITU-T Recommendation Q.115.1 [i.29].

For other national networks which have different propagation delay, the Access parameters from clause 6 can be used and the propagation delay from table 1 and 2 can be added. The explanation of the calculation of the delay values will be included in a further revision of the present document. In case of interleaving on the DSL access line, the additional delay has to be added.

To enable an easy comparison of the user satisfaction the tables are coloured in the same colours as table 11: Relation between R-value and user satisfaction.

Table 3 shows End-to-End delay in ms and R value between DSL line 128 kbit/s uplink; 128 kbit/s downlink and PSTN/ISDN G.729. The R values are based on ISDN Terminals with the Talker Echo Loudness Rating TELR = 65.

Table 4 shows End-to-End delay in ms and R value between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.711; G.726/40/20. The R values are based on ISDN Terminals with the Talker Echo Loudness Rating TELR = 65.

Table 5 shows End-to-End delay in ms and R value between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.729A. The R values are based on ISDN Terminals with the Talker Echo Loudness Rating TELR = 65.

Table 6 shows End-to-End delay ms Delay between DSL line 384 kbit/s uplink; 1 024 kbit/s downlink and PSTN/ISDN for G.711. The R values are based on ISDN Terminals with the Talker Echo Loudness Rating TELR = 65.

Table 7 shows End-to-End delay in ms and R value between DSL line 128 kbit/s uplink; 128 kbit/s downlink and PSTN/ISDN G.729. The R values are based on DECT Terminals.

Table 8 shows End-to-End delay in ms and R value between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.711; G.726/40/20. The R values are based on DECT Terminals. (Q.115.1 [i.29])

Table 9 shows End-to-End delay in ms and R value between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.729A. The R values are based on DECT Terminals.

Table 10 shows End-to-End delay ms Delay between DSL line 384 kbit/s uplink; 1 024 kbit/s downlink and PSTN/ISDN for G.711. The R values are based on DECT Terminals.

Table 3: End-to-End delay in ms and R value between DSL line 128 kbit/s uplink; 128 kbit/s downlink and PSTN/ISDN G.729 with ISDN Terminals

	PSTN/ISDN Delay (ms) / R	PSTN/ISDN- NGN Delay (ms) / R	DSL Delay (ms) / R
PSTN/ISDN	63 / 92	62 / 92	98 - 123 / 83
PSTN/ISDN- NGN	62 / 92	61 / 92	97 -122 / 83
DSL	120 - 145 / 83	119 - 144 / 83	155 - 180 / 81

Table 4: End-to-End delay in ms and R value between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.711 and G.726/40/20 with ISDN Terminals

	PSTN/ISDN Delay (ms) / R	PSTN/ISDN- NGN Delay (ms) / R	DSL Delay (ms) / R
PSTN/ISDN	63 / 92	62 / 92	73 - 98 / 91
PSTN/ISDN- NGN	62 / 92	61 / 92	72 - 97 / 91
DSL	81 - 106/ 91	80 - 105 / 91	122 / 90

Table 5: End-to-End delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink for G.729A with ISDN Terminals

	PSTN/ISDN Delay (ms) / R	PSTN/ISDN- NGN Delay (ms) / R	DSL Delay (ms) / R
PSTN/ISDN	63 / 92	62 / 92	69 - 99 / 84
PSTN/ISDN- NGN	62 / 92	61 / 92	68 - 98 / 84
DSL	91 - 116 / 83	90 - 115 / 83	122 / 83

Table 6: End-to-End delay ms Delay between DSL line 384 kbit/s uplink; 1 024 kbit/s downlink and PSTN/ISDN for G.711 with ISDN Terminals

	PSTN/ISDN Delay (ms) / R	PSTN/ISDN- NGN Delay (ms) / R	DSL Delay (ms) / R
PSTN/ISDN	63 / 92	62 / 92	67 - 92 / 91
PSTN/ISDN- NGN	62 / 92	61 / 92	66 - 91 / 91
DSL	79 / 91	78 / 91	83 - 108 / 91

Table 7: End-to-End delay in ms and R value between DSL line 128 kbit/s uplink; 128 kbit/s downlink and PSTN/ISDN G.729 with DECT Terminals TELR = 65 with the condition that the echo cancellation is deployed in the gateway according ITU-T Recommendation Q.115.1 [i.29]

	PSTN/ISDN Ie = 7 Delay (ms) / R	PSTN/ISDN- NGN Ie = 7 Delay (ms) / R	DSL Ie = 7 Delay (ms) / R
PSTN/ISDN	77 / 84	76 / 84	112 - 137 / 76
PSTN/ISDN- NGN	76 / 84	75 / 84	111 - 136 / 76
DSL	134 - 159 / 75	133 - 158 / 75	169 - 194 / 72

Table 8: End-to-End delay in ms and R value between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.711 and G.726/40/20 with DECT Terminals TELR = 65 with the condition that the echo cancellation is deployed in the gateway according ITU-T Recommendation Q.115.1 [i.29]

	PSTN/ISDN Ie = 7 Delay (ms) / R	PSTN/ISDN- NGN Ie = 7 Delay (ms) / R	DSL Ie = 7 Delay (ms) / R
PSTN/ISDN	77 / 84	76 / 84	87 - 112 / 84
PSTN/ISDN- NGN	76 / 84	75 / 84	86 - 111 / 84
DSL	95 - 110 / 84	94 - 109 / 84	136 / 84

Table 9: End-to-End delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink for G.729A with DECT Terminals TELR = 65 with the condition that the echo cancellation is deployed in the gateway according ITU-T Recommendation Q.115.1 [i.29]

	PSTN/ISDN Ie = 7 Delay (ms) / R	PSTN/ISDN- NGN Ie = 7 Delay (ms) / R	DSL Ie = 7 Delay (ms) / R
PSTN/ISDN	77 / 84	76 / 84	83 - 113 / 76
PSTN/ISDN- NGN	76 / 84	75 / 84	82 - 112 / 76
DSL	105 - 130 / 76	104 - 129 / 76	136 / 76

Table 10: End-to-End delay ms Delay between DSL line 384 kbit/s uplink; 1 024 kbit/s downlink and PSTN/ISDN for G.711 with DECT Terminals TELR = 65 with the condition that the echo cancellation is deployed in the gateway according ITU-T Recommendation Q.115.1 [i.29]

	PSTN/ISDN Ie = 7 Delay (ms) / R	PSTN/ISDN- NGN Ie = 7 Delay (ms) / R	DSL Ie = 7 Delay (ms) / R
PSTN/ISDN	77 / 84	76 / 84	81 - 106 / 84
PSTN/ISDN- NGN	76 / 84	75 / 84	80 - 105 / 84
DSL	93 - 105 / 84	92 - 104 / 84	97 - 122 / 83

4.4.2 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 11 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 11 shows Relation between *R*-value and user satisfaction.

Table 11: Relation between *R*-value and user satisfaction

R Value	MOS Value	Categories of User Satisfaction
94	4,42	Very satisfied (Best)
93	4,40	
92	4,38	
91	4,36	
90	4,34	
87	4,195	Satisfied (High)
85	4,18	
82	4,09	
81	4,06	
80	4,03	Some users dissatisfied (Medium)
77	3,85	
73	3,74	
70	3,60	
68	3,50	Many users dissatisfied (Low)
60	3,10	
50	2,58	Nearly all users dissatisfied (Poor)
MOS = 1 + (0,035) * R + (000 007) * R (R - 60) (100 - R)		
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.		

5 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.

5.1 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 2 for details. The categories in following tables refers to ITU-T Recommendation G.109 [i.6] with the following notations:

Table 12: Guidance on objectives for either Access Segment for R > 90

Parameter	Value
Jitter [ms] sending	50 ms
Jitter [ms] receiving	20 ms
IPLR	3×10^{-4}
IPER	3×10^{-5}

5.2 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.

Table 13: Guidance on Objectives for Total Transit Segments

Parameter	Value
Intra-continent Jitter Value - **5 ms** per Provider (maximum of 2 involved in the service delivery chain) (see note)	10 ms
Inter-continent Jitter Value - **10 ms** per Provider (maximum of 2 involved in the service delivery chain) (see note)	20 ms
IPLR	$3{,}0 \times 10^{-4}$
IPER	3×10^{-5}
Ie	0
NOTE: The Jitter Values are based on values contained in the GSMA document IR3445.	

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].

5.2.1 Availability

Values for availability are following:

- Availability of the IP Backbone Service Provider Core: 99,995 %.
- Service Providers connection to IP Backbone Service Provider core with single connection: 99,7 %.
- Service Providers connection to IP Backbone Service Provider core with dual connection: 99,9 %.

5.3 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:

- ETSI 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".
- ETSI 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".
- ETSI 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".
- ETSI 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".

5.4 End-to-End Aspects

Figures 8 and 9 depict a summary of the proposed delay objectives and the end-to-end delay targets that can be achieved.

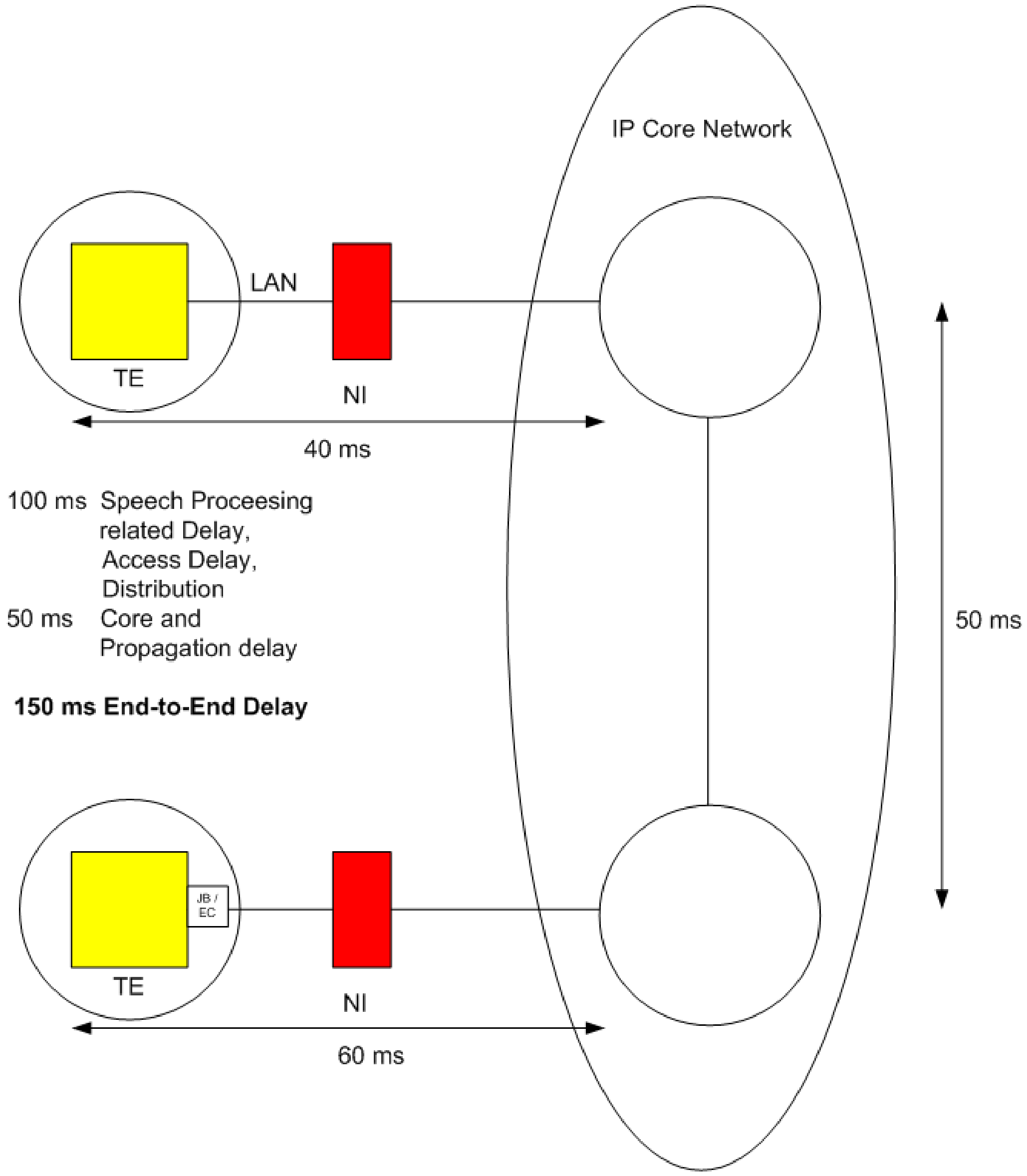

Figure 8: Delay Objectives for BEST (G.109) voice communication quality (R > 90)

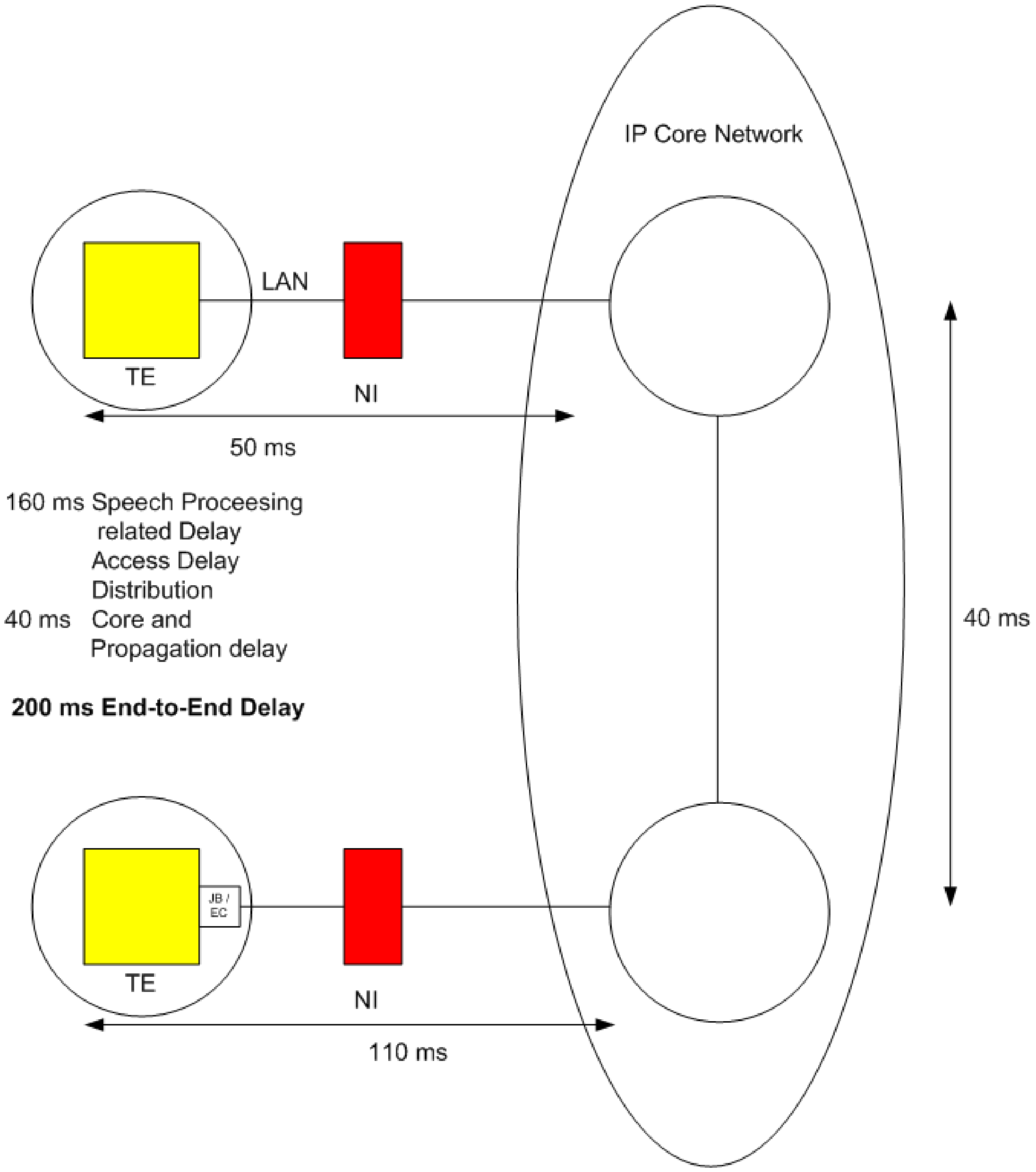

Figure 9: Delay Objectives for HIGH (G.109) voice communication quality (R > 80)

6 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 14: 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	$1,0 \times 10^{-8}$

7 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. A distinction needs to be made between time and timing synchronisation. Legacy networks tend only to be interested in timing synch whereas in IP based NGN, both time and timing can be important. Synchronous Ethernet provides timing synch whereas PTP and NTP provide both if correctly implemented.

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 (samples) 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 Bytes in the Packet	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) = de-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	GSM 06.60 (EN 300 726 [i.30]), 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 $1\text{-}10^{-3}$ 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 Comparison of Codecs, Link Speed and Capacity examples - Comparison of Codecs, Link Speed and Capacity

Table A.5: Comparison of Codecs, Link Speed and Capacity examples - Comparison of Codecs, Link Speed and Capacity; (ADSL RFC 1483 [i.31] Bridging)

	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	10	20	30
Frames per Packet (Samples)	80	160	240	160	240	160	80	2	3
IP Payload (Bytes)	120	200	280	120	160	140	90	60	70
IP Bitrate needed (kbit/s)	96	80	74	48	42	56	72	24	18
ATM Cells Needed IP Packet Size	4	5	7	4	4	4	3	2	2
ATM Bytes Needed	212	265	371	212	212	212	159	106	106
Serialization time (ms) 128 kbit/s (ATM Bytes Needed x8) / 128	13	17	23	13	13	13	10	7	7
Serialization time 256 kbit/s	7	8	12	7	7	7	5	3	3
Serialization time 384 kbit/s	4	6	8	4	4	4	3	2	2
Serialization time 1 024 kbit/s	2	2	3	2	2	2	1	1	1
ATM Bitrate Needed (kb/s) /channel (ATM Bytes X 8 / duration)	170	106	98	84,4	56	84	127	42	28
Delay in IP environment (ms) (2N + 1)x frame size + Look ahead Where: N = number of frames per packet; frame size is in ms	30	60	90	60	90	60	30	55	75
Delay in IP environment (ms) 2 x frame size + Look ahead Where: frame size is in ms	20	40	60	40	60	40	20	-	-

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

A.5 Serialization Delay

Table A.6: Serialization Delay in Milliseconds for Different Frame Sizes

Frame Size (bytes)	Line Speed (Kbps)										
	19,2	56	64	128	256	384	512	768	1 024	1 544	2 048
38	15,83	5,43	4,75	2,38	1,19	0,79	0,59	0,40	0,30	0,20	0,15
48	20,00	6,86	6,00	3,00	1,50	1,00	0,75	0,50	0,38	0,25	0,19
64	26,67	9,14	8,00	4,00	2,00	1,33	1,00	0,67	0,50	0,33	0,25
128	53,33	18,29	16,00	8,00	4,00	2,67	2,00	1,33	1,00	0,66	0,50
256	106,67	36,57	32,00	16,00	8,00	5,33	4,00	2,67	2,00	1,33	1,00
512	213,33	73,14	64,00	32,00	16,00	10,67	8,00	5,33	4,00	2,65	2,00
1 024	426,67	149,29	128,00	64,00	32,00	21,33	16,00	10,67	8,00	5,31	4,00
1 500	625,00	214,29	187,50	93,75	46,88	31,25	23,44	15,63	11,72	7,77	5,86
2 048	853,33	292,57	256,00	128,00	64,00	42,67	32,00	21,33	16,00	10,61	8,00

Table A.7: Best and Worst Case Processing Delay

Coder	Rate	Required Sample Block	Best Case Coder Delay	Worst Case Coder Delay
ADPCM, G.726	32 kbit/s	10 ms	2,5 ms	10 ms
CS-ACELP, G.729A	8,0 kbit/s	10 ms	2,5 ms	10 ms
MP-MLQ, G.723.1	6,3 kbit/s	30 ms	5 ms	20 ms
MP-ACELP, G.723.1	5,3 kbit/s	30 ms	5 ms	20 ms

A.6 Transport Reference Parameters

A.6.1 Void

A.6.2 Network and Access Parameters

Table A.8 shows the voice sample size for the transport network at the various instances in ms and bytes respectively.

Table A.8: Transport Network parameters

Voice Packet Size						
Codec	Packet size in ms	In Byte	RTP	UDP	IP	IP Packet size IPV4
G.711	10	80	12	8	20	120
G.711	20	160	12	8	20	200
G.729	10		12	8	20	
G.729	20		12	8	20	

Table A.9 shows access network parameters for different technologies.

NOTE: This values are examples, they can differ due to different settings of the DSL connections.

Table A.9: Access Network parameters

Technology	Transport in Byte	Number of Packets	Voice with overhead	Data rate in kbit/s
Ethernet	218	N/A	265 Byte	90,4
ATM	N/A	5	265 Byte	106

Table A.10 shows ADSL serialization times for access and codecs.

Table A.10: ADSL serialization times

ADSL access line upstream (kbit/s)	ADSL access line downstream (kbit/s)	Serialization time for G.711 20 ms (ATM based)	Serialization time for G.729A (ATM based)
128	128	17 ms	7 ms
256	256	8 ms	3 ms
384		6 ms	2 ms
512		4 ms	
768		3 ms	
1 024	1 024	2 ms	1 ms

Table A.11 shows backbone parameters.

Table A.11: Backbone parameters

Parameter	Value
Intra-continent jitter value - 5 ms per provider (maximum of 2 involved in the service delivery chain)	10
Inter-continent jitter value - 10 ms per provider (maximum of 2 involved in the service delivery chain)	20
IPLR	$3{,}0 \times 10^{-4}$
IPER	3×10^{-5}
Ie	0

Table A.12 shows various coder parameters.

Table A.12: Coder parameters

Coder Processing Delay	Algorithmic Delay G.729	Decompression Delay
Default values: GW: 2 ms IAD: 10 ms	15 ms	G.711: 1 ms G.729: 2 ms

Table A.13 shows coder delay values.

Table A.13: Coder delay

Packetization Delay	20 ms
Worst Case Compression Time per Block	GW = 2 ms; IAD = 10 ms
Algorymic delay G.729	5 ms look ahead + 10 ms frame size
Serialization	Depends on the access

Table A.14 shows decoder delay values.

Table A.14: Decoder delay

Decompression time per block	1 ms G.711, 2 ms G.729
Serialization time	Depends on the access
De-jitter Buffer Size Min	Buffering delay caused due to download 1 x Serial. time data packet (1 500 Byte)
De-jitter Buffer Size Max	Buffering delay caused due to download 1 x Serial. time data packet (1 500 Byte)
Play out buffer Size Min	1 time data packet (1 500 Byte)
Play out buffer Size Max	1,5 time data packet (1 500 Byte) + voice serial time
De-jitter buffer delay	0,5 time de-jitter buffer size

A.6.3 Delay and Jitter Values

A.6.3.1 Delay and Serialization time

Table A.15 shows delay values or respective ranges for various network elements and signalization frames.

The serialization times of signalling packets produce delay variation for media packets, even if no other traffic (e.g. data traffic is present).

Table A.15: Network element delay and Serialization time for signalization frame

Network element	Delay	Serialization time for signalization frame	Comments
Legacy network switch	0,45 s	0	
MGW Sending	22 ms	0	G.711; 20 ms packetization; STM 1
MGW Receiving	26 ms	0	½ De-jitter Buffer 50 ms + Depacketization + PLC; STM 1
ADM	0,1 ms	0	
Transmission fiber optic	5 µs/km	0	
IAD Sending ATM line 128 kbit/s	47 ms	50 - 94 ms Caused by Signalization	G.711; 20 ms packetization, 128 kbit/s ATM Line (reInvite 1 500 Byte x 8 / 128 10^3 = 94 ms; Registration 800 Byte 50 ms)
IAD Sending ATM line 256 kbit/s	38 ms	25 - 46 ms Caused by Signalization	G.711; 20 ms packetization, 256 kbit/s ATM Line (reInvite 1 500 Byte x 8 / 256 10^3 = 46 ms; Registration 800 Byte 25 ms)
IAD Sending ATM line 384 kbit/s	36 ms	17 - 31 ms Caused by Signalization	G.711; 20 ms packetization, 384 kbit/s ATM Line (reInvite, 1 500 Byte x 8 / 384 10^3 = 31 ms; Registration 800 Byte)
IAD Sending ATM line 512 kbit/s	34 ms	13 - 23 ms Caused by Signalization	G.711; 20 ms packetization, 512 Kbit/s ATM Line (reInvite 1 500 Byte x 8 / 512 10^3 = 23 ms; Registration 800 Byte = 13 ms)
IAD Sending ATM line 768 kbit/s	33 ms	8 - 15 ms Caused by Signalization	G.711; 20 ms packetization, 768 Kbit/s ATM Line (reInvite 1 500 Byte x 8 / 768 10^3 = 15 ms; Registration 800 Byte = 8 ms)
IAD receiving ATM line 128 Kbit/s Jitter Buffer = 150 ms	93 ms	37 ms Caused by Signalization	G.711; 20 ms packetization, ½ De-jitter Buffer 150 ms + depacketization + PLC 128 kbit/s ATM Line (200 OK, 401 600 Byte x 8 / 128 10^3)

Network element	Delay	Serialization time for signalization frame	Comments
IAD receiving ATM line 128 Kbit/s Jitter Buffer = 200 ms	118 ms	37 ms Caused by Signalization	G.711; 20 ms packetization, ½ De-jitter Buffer 200 ms + depacketization + PLC 128 kbit/s ATM Line (200 OK, 401 600 Byte x 8 / 128 10^3)
IAD receiving ATM line 256 kbit/s Jitter Buffer = 100 ms	59 ms	19 ms Caused by Signalization	G.711; 20 ms packetization ½ De-jitter Buffer 100 ms + depacketization + PLC 256 kbit/s ATM Line (200 OK, 401 600 Byte x 8 / 256 10^3)
IAD receiving ATM line 256 kbit/s Jitter Buffer = 150 ms	84 ms	19 ms Caused by Signalization	½ De-jitter Buffer 150 ms + depacketization + PLC 256 kbit/s ATM Line (200 OK, 401 600 Byte x 8 / 256 10^3)
IAD receiving ATM 1 024 kbit/s Jitter Buffer = 50 ms	28 s	5 ms Caused by Signalization	½ De-jitter Buffer 50 ms + depacketization + PLC 1 024 kbit/s ATM Line (200 OK, 401 600 Byte x 8 / 1 024 10^3)
IAD receiving ATM 1 024 kbit/s Jitter Buffer = 100 ms	53 s	5 ms Caused by Signalization	½ De-jitter Buffer 100 ms + depacketization + PLC 1 024 kbit/s ATM Line(200 OK, 401 600 Byte x 8 / 1 024 10^3)
DSLAM sending	0,1 - 0,3 ms		(Packet size 500 - 1 500 Byte)
DSLAM Receiving	0,1 - 0,3 ms		(Packet size 500 - 1 500 Byte)
Ethernet switch	0,1 - 0,3 ms		(Packet size 500 - 1 500 Byte)
ATM switch STM 1	0,1 - 0,3 ms		(Packet size 500 - 1 500 Byte)
ATM Access switch STM 1 (APEX)	0,3 - 1 ms		(Packet size 500 - 1 500 Byte)
BRAS	1 - 3 ms		
WiMAX	25 ms		
IAD Sending Ethernet	29 ms		
IAD Receiving Ethernet	24 ms		
Mobile Station GSM Uplink	72,1 ms		
Mobile Station GSM Downlink	14,3 ms		
BTS Uplink	15,8 ms		
BTS Downlink	40,8 ms		
PtP Microwave link	15,8 ms		
PDH Micowave link	1,4 ms		
BSC	1 ms		
TRAU Uplink	1,5 ms		
TRAU Downlink	39 ms		
MSC Uplink	0,5 ms		
MSC Downlink	1,5 ms		
UE, R.99	37 ms		
Node B, Uplink R.99	22 ms		
Node B, Downlink R.99	9 ms		
RNC Uplink R.99	12 ms		
RNC Downlink	12 ms		
TRAU R.99	11 ms		
UMSC	5 ms		
SBC	0,1 ms		
Router Distribution Layer	0,5 ms		
Router Core Layer	0,1 ms		
Seriazation time 128 kbit/s	16,56 ms		128 Kbit/s ATM Line
Seriazation time 256 kbit/s	8,28 ms		256 Kbit/s ATM Line
Seriazation time 384 kbit/s	5,52 ms		384 Kbit/s ATM Line
Seriazation time 512 kbit/s	4,14 ms		512 Kbit/s ATM Line
Seriazation time 1 024 kbit/s	2,07 ms		1 024 Kbit/s ATM Line
Seriazation time 2 048 kbit/s	1,03 ms		2 048 Kbit/s ATM Line
Seriazation time downlink 2 048 kbit/s	1 ms		

Network element	Delay	Serialization time for signalization frame	Comments
Digital transit exchange			
digital-digital	0,45 ms		
Digital local exchange			
analogue-analogue	1,5 ms		
Digital local exchange, analogue subscriber line-digital junction	0,975 ms		
Digital local exchange, digital subscriber line-digital junction	0,825 ms		
Echo cancellers	0,5 ms		

A.6.3.2 Queuing and Buffering Delay

After the compressed voice payload is built, a header is added and the frame is queued for transmission on the network connection. Voice needs to have strict priority in the router/gateway. Therefore, a voice frame should only wait for either one or several data frames that already plays out (depending on the implementation of the prioritization algorithm, or for other voice frames ahead of it. Essentially the voice frame waits for the serialization delay of any preceding frames in the output queue. Queuing delay ($ß_n$) is a variable delay and is dependent on the trunk speed and the state of the queue. There are random elements associated with the queuing delay.

$t_{D\text{-}max}$ = (Maximum # Data MTU bytes + 48 overhead) / (link speed kbps / 8).

Total core network maximum data MTU queuing time is: = $t_{Q\text{-}wo}$ * (number of hops -1).

Table A.16 shows queuing and buffering delay values caused by different configurations based on the "worst case" assumption that either several voice terminals are connected or that voice and video services are operated at the same time.

Table A.16: Queuing and Buffering Delay

Network element	Max Queuing/ Buffering Delay $t_{D\ max}$	Queuing / Buffering delay caused due to interaction with data traffic (see note 1)
IAD sending ATM G.711; 128 Kbit/s ATM Line	94 ms	158 ms
IAD sending ATM G.729; 128 Kbit/s ATM Line	94 ms	148 ms
IAD sending ATM G.711; 256 Kbit/s ATM Line	47 ms	79 ms
IAD sending ATM G.729A; 256 Kbit/s ATM Line	48 ms	79 ms
IAD sending ATM G.711; 384 Kbit/s ATM Line	31 ms	53 ms
IAD receiving ATM G.711; 128 Kbit/s ATM Line	37 ms	
IAD receiving ATM G.711; 256 Kbit/s ATM Line	19 ms	
IAD receiving ATM G.711; 1 024 Kbit/s ATM Line	5 ms	
NOTE 1: 1,5 x Serial. time data packet (1 500 Byte) + voice serial time. NOTE 2: Serialization time for data frame caused by Signalization in regularly time distance (e.g. every 60 s).		

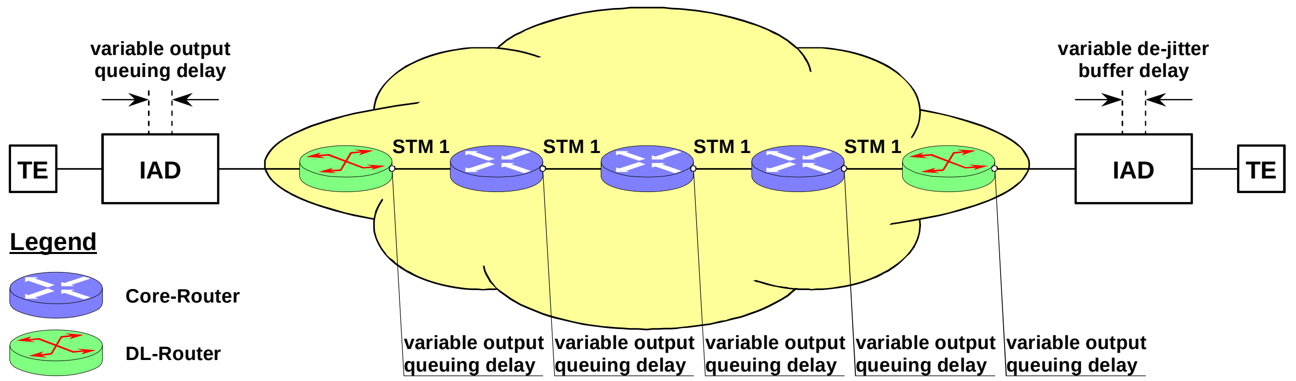

Figure A.1: Variable Delay and the De-Jitter Buffer Reference Diagram

From a de-jitter buffer point of view, it has to adapt to the maximum end-to-end delay variation (green curve in figure A.2 shows possible end-to-end audio delay variation, the steps are due to de-jitter buffer adaption to delay variation).

a) Playout buffer:

In the playout buffer are usually 1, 2 or more packets. In the playout buffer, no prioritisation exists. A packet in the playout buffer has to be sent first, even if it is a low priority packet and a strict priority packet is waiting.

NOTE: The de-jitter buffer should compensate the time which is needed that the packets lives the playout buffer.

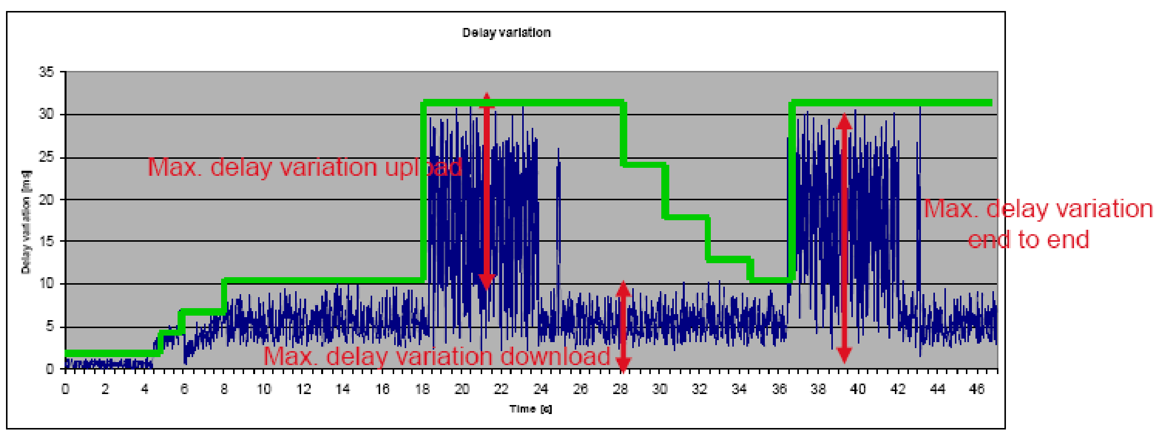

Figure A.2: Maximum Delay Variation

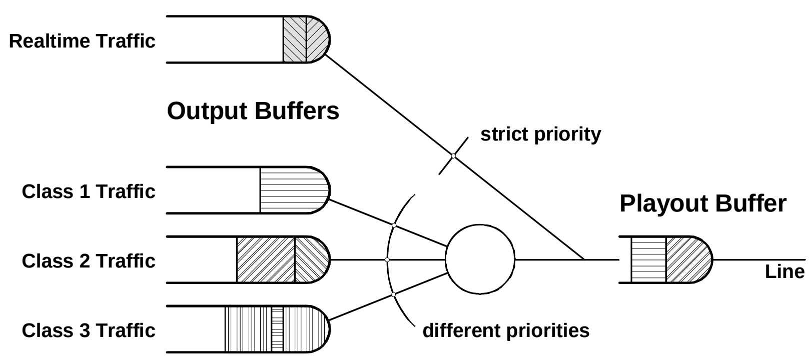

Figure A.3: Playout Buffer

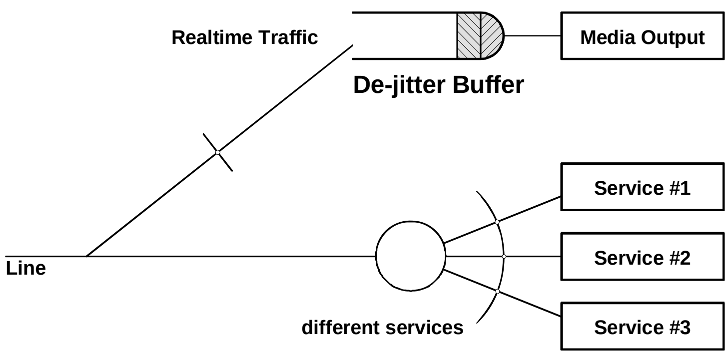

Figure A.4: De-jitter Buffer

b) De-jitter buffer:

The initial playout delay of the de-jitter buffer is configurable in most implementations. The maximum depth of the buffer before it overflows is normally set to 1,5 or 2,0 times this value.

A.6.3.3 Delay and Jitter Values for PSTN/ISDN classic access

Table A.17 shows end-to-end delay values between service provider premises.

Table A.17: One way delay values between originating and terminating Service Provider premises

	Digital local exchange, analogue subscriber line-digital junction	Transit Exchange	MGW	Echo cancellers	Summ
Sending	0,975 ms	0,45 ms	22 ms	0. 5 ms	24 ms (39 ms G.729)
Receiving 50 ms De-jitter Buffer	0,975 ms	0,45 ms	26 ms	0,5 ms	28 ms
Receiving 80 ms De-Jitter Buffer	0,975 ms	0,45 ms	36 ms	0,5 ms	38 ms
Receiving 100 ms De-Jitter Buffer	0,975 ms	0,45	51 ms	0,5 ms	53 ms
Receiving 150 ms De-Jitter Buffer	0,975 ms	0,45	76 ms	0,5 ms	78 ms

A.6.3.4 Delay and Jitter Values for NGN PSTN/ISDN access

Table A.18 shows end-to-end delay values between service provider premises with NGN PSTN/ISDN access.

Table A.19 shows the End-to-End delay values between PSTN/ISDN users for different De-jitter Buffer values.

Table A.18: One way delay values between originating and terminating Service Provider premises with NGN PSTN/ISDN access

	Digital local exchange, analogue subscriber line-digital junction	MGW	Echo cancellers	Summ
Sending	0,975 ms	22 ms	0. 5 ms	23 ms (38 ms G.729)
Receiving 50 ms De- Jitter Buffer	0,975 ms	26 ms	0,5 ms	27 ms
Receiving 100 ms De-Jitter Buffer	0,975 ms	51 ms	0,5 ms	52 ms
Receiving 150 ms De-Jitter Buffer	0,975 ms	76 ms	0,5 ms	77 ms

Table A.19: End-to-End delay values between PSTN/ISDN users for different De-jitter Buffer values

		PSTN/ISDN	PSTN/ISDN- NGN
De-jitter Buffer 50 ms	PSTN/ISDN	52	51
	PSTN/ISDN- NGN	51	50
		PSTN/ISDN	PSTN/ISDN- NGN
De-jitter Buffer 100 ms	PSTN/ISDN	77	76
	PSTN/ISDN- NGN	76	75

A.6.3.5 Delay and Jitter Values for Symmetric Access DSL (128 kbit/s)

Table A.20 shows the one way delay between originating and terminating Service Provider premises for ADSL line 128 kbit/s uplink; 128 kbit/s downlink G.729A.

Table A.21 shows the one way delay between originating and terminating Service Provider premises for ADSL line 128 kbit/s uplink; 128 kbit/s downlink G.711.

Table A.22 shows the De-jitter buffer values for DSL line 128 kbit/s uplink; 128 kbit/s downlink, for more registered terminals without additional data traffic.

Table A.23 shows the End-to-End Delay between DSL line 128 kbit/s uplink; 128 kbit/s downlink and POTS/ISDN for G.729A without regional propagation delay.

Table A.20: Delay for DSL line 128 kbit/s uplink; 128 kbit/s downlink G.711 for more registrated terminals

	IAD	DSLAM	ETH	ADM	BRAS	Sum
Sending	47 ms	0,3 ms (see note)	0,6 ms	0,4 ms	3 ms	51 ms
Receiving	68 ms (100 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	72 ms
Receiving	93 ms (150 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	97 ms
Receiving	118 ms (200 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	122 ms
NOTE: In case of interleaving, the additional delay should be added.						

Table A.21: Delay for DSL line 128 kbit/s uplink; 128 kbit/s downlink G.729A for more registrated terminals

	IAD	DSLAM	ETH	ADM	BRAS	Sum
Sending	52 ms	0,3 ms (see note)	0,6 ms	0,4 ms	3 ms	56 ms
Receiving	59 ms (100 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	63 ms
Receiving	84 ms (150 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	88 ms
Receiving	109 ms (200 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	113 ms
NOTE: In case of interleaving, the additional delay should be added.						

Table A.22: De-jitter buffer values for DSL line 128 kbit/s uplink; 128 kbit/s downlink, for more registrated terminals without additional data traffic

	Access
Sending	94 ms (149 ms - G.729)
Receiving	37 ms
De-jitter buffer POTS- DSL: 94 ms DSL-> DSL: 131 ms De-jitter buffer Max: POTS- DSL: 149 ms DSL-> DSL: 186 ms	

Table A.23: End-to-End delay for DSL line 128 kbit/s uplink; 128 kbit/s downlink G.729 - worst case scenario (JB POTS/ DSL 150 ms, DSL-DSL 200 ms) and best case scenario (JB POTS/ DSL 100, DSL-DSL 150 ms)

	PSTN/ISDN	PSTN/ISDN- NGN	DSL
PSTN/ISDN			87 - 112
PSTN/ISDN- NGN			86 - 111
DSL	109 - 134	108 - 133	144 - 169

A.6.3.6 Delay and Jitter Values for Symmetric Access DSL (256 kbit/s)

Table A.24 shows the one way delay between originating and terminating Service Provider premises for ADSL line 256 kbit/s uplink; 256 kbit/s downlink G.711.

Table A.25 shows the one way delay between originating and terminating Service Provider premises for ADSL line 256 kbit/s uplink; 256 kbit/s downlink G.726/40/20.

Table A.26 shows the one way delay between originating and terminating Service Provider premises for ADSL line 256 kbit/s uplink; 256 kbit/s downlink G.729A.

Table A.27 shows the De-jitter buffer values for DSL line 256 kbit/s uplink; 256 kbit/s downlink.

Table A.28 shows the End-to-End delay ms Delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.711 without regional propagation delay.

Table A.29 shows the End-to-End delay ms Delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.726/40/20 without regional propagation delay.

Table A.30 shows the End-to-End delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.729A (Adaptive JB POTS/ DSL 100 ms) without regional propagation delay.

Table A.31 shows the End-to-End delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.729A (Adaptive JB POTS 80 ms / DSL 100 ms) without regional propagation delay.

Table A.24: One way delay for DSL line 256 kbit/s uplink; 256 kbit/s downlink G.711

	IAD	**DSLAM**	**ETH**	**ADM**	**BRAS**	**Sum**
Sending	38 ms	0,3 ms (see note 2)	0,6 ms	0,4 ms	3 ms	42 ms
Receiving	34 ms (50 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	38 ms
Receiving	59 ms (100 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	63 ms
NOTE 1: In the playout buffer 1, 2 or more packets. In the playout buffer, no prioritisation exists. A packet in the playout buffer has to be sent first, even if it is a low priority packet and a strict priority packet is waiting. NOTE 2: In case of interleaving, the additional delay should be added.						

Table A.25: One way delay for DSL line 256 kbit/s uplink; 256 downlink; G.726 /40/20

	IAD	DSLAM	ETH	ADM	BRAS	Sum
Sending	37 ms	0,3 ms (see note 2)	0,6 ms	0,4 ms	3 ms	41 ms
Receiving	33 ms (50 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	37 ms
Receiving	59 ms (100 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	62 ms
NOTE 1: In the playout buffer 1, 2 or more packets. In the playout buffer, no prioritisation exists. A packet in the playout buffer has to be sent first, even if it is a low priority packet and a strict priority packet is waiting. NOTE 2: In case of interleaving, the additional delay should be added.						

Table A.26: One way delay for DSL line 256 kbit/s uplink; 256 downlink; G.729A

	IAD	DSLAM	ETH	ADM	BRAS	Sum
Sending	48 ms	0,3 ms (see note 2)	0,6 ms	0,4 ms	3 ms	52 ms
Receiving	30 ms (50 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	34 ms
Receiving DSL - DSL	55 ms (100 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	59 ms
Receiving DSL - DSL	85 ms (150 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	89 ms
NOTE 1: In the playout buffer 1, 2 or more packets. In the playout buffer, no prioritisation exists. A packet in the playout buffer has to be sent first, even if it is a low priority packet and a strict priority packet is waiting. NOTE 2: In case of interleaving, the additional delay should be added.						

Table A.27: De-jitter buffer values for DSL line 256 kbit/s uplink; 256 kbit/s downlink

	Access
Sending	47 ms (79 ms)
Receiving	19 ms
De-jitter buffer DSL->POTS: 47 ms DSL->DSL: 62 ms De-jitter buffer Max: DSL->POTS: 79 ms DSL->DSL: 98 ms	

Table A.28: End -to - End delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.711 (JB POTS/DSL 100 ms) worst case and best case scenario (JB POTS/ DSL 50, DSL-DSL 100 ms)

	PSTN/ISDN	PSTN/ISDN- NGN	DSL
PSTN/ISDN			62 - 87
PSTN/ISDN- NGN			61 - 86
DSL	70 - 95	69 - 94	111

Table A.29: End-to-End delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.726 /40/20 (JB POTS/ DSL100 ms) worst case and best case scenario (JB POTS/ DSL 50, DSL-DSL 100 ms)

	PSTN/ISDN	**PSTN/ISDN- NGN**	**DSL**
PSTN/ISDN			61 - 86
PSTN/ISDN- NGN			60 - 85
DSL	69 - 94	68 - 93	105

Table A.30: End-to-End delay between DSL line 256 kbit/s uplink; 256 kbit/s downlink and PSTN/ISDN for G.729A (Adaptive JB POTS/ DSL 100 ms) worst case and best case scenario (JB POTS/ DSL 50, DSL-DSL 100 ms)

	PSTN/ISDN	**PSTN/ISDN- NGN**	**DSL**
PSTN/ISDN			58 - 88
PSTN/ISDN- NGN			57- 87
DSL	80 - 105	79 - 104	111

A.6.3.7 Asymmetric Access DSL (384 kbit/s uplink; 1 024 kbit/s downlink)

Table A.31 shows the one way delay between originating and terminating Service Provider premises for ADSL line 384 kbit/s uplink; 1 024 kbit/s downlink G.711.

Table A.32 shows the De-jitter buffer values for DSL line 384 kbit/s uplink; 1 024 kbit/s downlink.

Table A.33 shows End-to-End delay between DSL line 384 kbit/s uplink; 1 024 kbit/s downlink and PSTN/ISDN (JB POTS 50 ms; DSL 100 ms).

Table A.31: One way delay for DSL line 384 kbit/s uplink, 1 024 kbit/s downlink

	IAD (Receiving time ½ de-jitter Buffer + Depacketization + Ser. Time)	**DSLAM**	**ETH**	**ADM**	**BRAS**	**Sum**
Sending	36 ms	0,3 ms (see note)	0,6 ms	0,4 ms	3 ms	40 ms
Receiving DSL - POTS	28 ms (50 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	32 ms
Receiving DSL - DSL	53 ms (100 ms JB)	0,3 ms	0,6 ms	0,4 ms	3 ms	57 ms
NOTE: In case of interleaving, the additional delay should be added.						

Table A.32: De-jitter buffer value for DSL line 384 kbit/s uplink, 1 024 kbit/s downlink

<table>
<tr><th></th><th>Access</th></tr>
<tr><td>Sending</td><td>31 ms (53 ms)</td></tr>
<tr><td>Receiving</td><td>5 ms</td></tr>
<tr><td colspan="2">De-jitter:
DSL - > POTS: 31 ms
DSL-> DSL: 36 ms

De-jitter Max.:
DSL - > POTS: 53 ms
DSL-> DSL: 58 ms</td></tr>
</table>

Table A.33: End-to-End delay between DSL line 384 kbit/s uplink; 1 024 kbit/s downlink and PSTN/ISDN (JB POTS 50 ms; DSL 100 ms) worst case and best case scenario (JB POTS/ DSL 50, DSL-DSL 50 ms)

	PSTN/ISDN	PSTN/ISDN- NGN	DSL
PSTN/ISDN			56 - 81 ms
PSTN/ISDN-NGN			55 - 80 ms
DSL	68 ms	67	72 - 97 ms

Annex B: Bibliography

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

History

Document history		
V1.1.1	February 2009	Publication
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