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Technical Report

Speech and multimedia Transmission Quality (STQ); Quality of Service Implications of NGN Architectures

Reference

DTR/STQ-00133

Keywords

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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

Quality of Service (QoS) may be loosely defined as a measure of the end-user's perception of the ease of use and the accuracy of the facility being provided. QoS has always been an important measure of the performance of communications networks. In networks where the connectivity is fixed for the duration of particular communication it is relatively easy to define the maximum impairment produced in the individual elements making up the connection, such that their addition would still result in an acceptable service being provided.

In packet based networks, routing of individual data packets constantly varies. Due to the complexities and random nature of packet routing, any end-user transaction has the possibility of being completed via a very large number of links using varying technologies. This may result in the possibility of the end-user perception in terms of the distortion, delay, error rate, etc. changing from time to time during a particular transaction. End-to-end error-correcting techniques serve to reduce the perceived problems for data services but the quality of speech connections can be seriously impaired if network performance is not adequately controlled.

In de-regulated networks, the overall QoS can be further affected by inferior end-user equipment, something totally outside that network operators' control. It has been suggested that some users may be prepared to change their communication habits or tolerate some level of impairments in exchange for the benefits of mobility and ubiquity of communications. The purpose of their communication may also have an impact on the user's expectations; during a social call they may be more tolerant of impairments than during a business call.

It is not easy for network operators to determine the level of impairment users are prepared to accept and there are no longer specific requirements for this in international standards. In any case, the acceptable level is likely to be user and/or task-dependent. Also, delay and distortion measurements in a laboratory environment may be inappropriate in 'real-world' scenarios involving user-owned equipment.

The present document is addressed to network operators, service providers, users, manufacturers and regulators. It considers the end-to-end quality of service implications of planned and existing Next Generation Network (NGN) and hybrid technology networks. It is the intention that it will lead to improved standards on end-to-end QoS. This will facilitate the take-up of NGN services by allowing differentiation between NGN and traditional Internet service offerings. The present document may also have an impact on NGN architectural design and related standards.

1 Scope

The present document considers end-to-end quality of service implications of planned and existing Next Generation Network and hybrid technology network architectures including mobile architectures. These considerations were performed in two steps:

a) Initially a study of real-world architectures has been made based on material declared by network operators. A questionnaire (see Annex A) was prepared that aided the interviews with representatives of network operators. Additional material was made available through TISPAN and STQ members.

b) An analysis based on the material collected in step a) above was then performed of the network performance (including delay, jitter, packet loss) to be expected for specific access and core architectures, based on declared figures or existing standards.

Multiple reference connections, based on various access and core architectures and technologies, were designed and their end-to-end performance and QoS characteristics determined and included in a revision of TR 102 775 [i.15]. Analysis of this part of the work leads to recommendations on new standards needed and changes required to existing standards. The focus is on voice quality, but other applications have also been considered (e.g. "64 kit/s unrestricted" applications).

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.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
    - if it is accepted that it will be possible to use all future changes of the referenced document for the purposes of the referring document;
    - for informative references.

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

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

2.1 Normative references

The following referenced documents are 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] ETSI EN 383 001 (V1.1.1): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".

[i.2] ETSI EN 302 097 (2000): "Integrated Services Digital Network (ISDN); Signalling System No.7 (SS7); ISDN User Part (ISUP); Enhancement for support of Number Portability (NP) [ITU-T Recommendation Q.769.1 (2000), modified]".

[i.3] ETSI TR 183 014 (2005-12): "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN Emulation; Development and Verification of PSTN/ISDN Emulation".

[i.4] ETSI ES 282 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Functional Architecture".

[i.5] 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.6] 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.7] 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.8] 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.9] ETSI TS 102 737: "Speech and multimedia Transmission Quality (STQ); Transmission requirements for narrowband wireless terminals (handset and headset) from a QoS perspective as perceived by the user".

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[i.13] ETSI EN 300 175-8 (V2.2.1): "Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech and audio coding and transmission".

[i.14] ETSI EN 300 176-2 (V2.1.1): "Digital Enhanced Cordless Telecommunications (DECT); Test specification; Part 2: Audio and speech".

[i.15] ETSI TR 102 775: "Speech and multimedia Transmission Quality (STQ); Guidance on objectives for Quality related Parameters at VoIP Segment-Connection Points; A Support to NGN transmission planners".

[i.16] ETSI EG 201 769-1 (V1.1.1): "Speech Processing, Transmission and Quality Aspects (STQ); QoS parameter definitions and measurements; Part 1: Parameters for voice telephony service required under the ONP Voice Telephony Directive 98/10/EC".

[i.17] ETSI EG 202 057-1 (V1.3.1): "Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 1: General".

[i.18] ETSI EG 202 057-2 (V1.3.1): "Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 2: Voice telephony, Group 3 fax, modem data services and SMS".

[i.19] ETSI EG 202 057-3 (V1.1.1): "Speech Processing, Transmission and Quality Aspects (STQ); User related QoS parameter definitions and measurements; Part 3: QoS parameters specific to Public Land Mobile Networks (PLMN)".

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[i.81] ITU-T Recommendation G.114: "One-way transmission time".

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NOTE: This Recommendation is still a draft at the time of the publication of the present document.

3 Abbreviations

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

3GPP	3rd Generation Partnership Project
A-BGF	Access Border Gateway Function
ADM	Add and Drop Multiplexer
ADSL	Asymmetric Digital Subscriber Line
AGC	Automatic Gain Control
AGCF	Access Gateway Control Function
AGF	Access Gateway Function
AS	Application Server
ATM	Asynchronous Transfer Mode
BGF	Border Gateway Function
BRA	ISDN Basic Rate Access
C-BGF	Core Border Gateway Function
CL	Router Core Layer
CN	Corporate Network
CPE	Customer Premises Equipment
CPN	Customer Premises Network
cRTP	compressed Realtime Transport Protocol
DECT	Digital Enhanced Cordless Telecommunication
DL	Distribution Router Layer
DNS	Domain Name Server
DSL	Digital Subscriber Line
EC	Echo Canceller
FEC	Forward Error Correction
FMC	Fixed Mobile Convergence
FoIP	Fax over IP
FTP	File Transfer Protocol
HGW	Home Gateway
HRC	Hypothetical Reference Connection
HTTP	Hypertext Transfer Protocol
IAD	Integrated Access Device
I-BGF	Interconnection Border Gateway Function
IEEE	Institute of Electrical and Electronics Engineers
IGMP2	Internet Group Multicast Protocol, version 2
IMS	IP Multimedia Subsystem
INT	IMS Network Testing
IP	Internet Protocol
IPTV	Internet Protocol Television
ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
ITU-T	ITU Telecommunication Standardization Sector
IWF	InterWorking Function
LAN	Local Area Network
MAC	Medium Access Control
MGW	Media Gateway
MoIP	Modem over IP
MOS	Mean Opinion Score
MPEG	Moving Picture Experts Group
MPLS	Multi-Protocol Label Switching
NGCN	Next Generation Corporate Network
NGN	Next Generation Network
NR	Noise Reduction
OAB	One Address Book
OSPF	Open Shortest Path First
PABX	Private Automatic Branch Exchange
PAI	P-asserted Identity
P-CSCF	Proxy Call Session Control Function
PDD	Post Dial Delay

P-IWF	Non-IMS Interworking Function
PLC	Packet Loss Concealment
P-MGF	Non-IMS Media Gateway Function
P-MGW	Non-IMS Media Gateway
POTS	Plain Old Telephone System
PRA	ISDN Primary Rate Access
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RFC	Request For Comments
RIP	Routing Information Protocol
RTCP-XR	Real-Time Transport Control Protocol Extended Reports
RTP	Real-time Transport Protocol
RTSP	Real-Time Streaming Protocol
SBC	Session Border Control
SDH	Synchronous Digital Hierarchy
SDO	Standards Developing Organization
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SME	Small and medium enterprises
SMPTE VC-1	Society of Motion Picture and Television Engineers Video Codec, version 1
SPNE	Signal Processing Network Equipment
TDM	Time Division Multiplex
TISPAN	Telecommunications and Internet converged Services and Protocols for Advanced Networking
TM	Trade Mark
TME	Terminate Early
UDP	User Datagram Protocol
UE	User Equipment
UNI	User-Network Interface
VAD	Voice Activity Detection
VED	Voice Enhancement Devices
VoIP	Voice over Internet Protocol
WDM	Wavelength Division Multiplex
WiMAX	Worldwide Interoperability for Microwave Access
xDSL	any Digital Subscriber Line technology

4 Intention

This clause reflects the mandate given to create the present document.

4.1 Overview

The present document provides information on the end-to-end quality of service implications of planned and existing Next Generation Network and hybrid technology network architectures including mobile architectures. A study of real-world architectures has been made based on material declared by network operators, material made available through TISPAN and STQ members and by surveying operators.

NOTE 1: This survey was performed by interviewing operators based on a questionnaire (see Annex A).

Based on the replies to the questionnaire, an analysis has been performed of the network performance (including delay, jitter, packet loss) to be expected for specific access and core architectures, based on declared figures or existing standards (see Annex B).

Following the results of this analysis, multiple reference connections were designed, based on various access and core architectures and technologies, and their end-to-end performance and QoS characteristics determined (see clause 6). Analysis of this part of the work lead to conclusions (see clause 7) and gaps identified in standardization (see clause 8).

NOTE 2: The focus is on voice quality, but other applications were also considered where appropriate information is available (e.g. "64 kbit/s unrestricted" applications).

4.2 Relations

4.2.1 Relation to other STQ workitems

The present document forms part of STQ's roadmap with respect to Quality aspects of NGN [i.47] and relates to TS 102 713 [i.73] (currently at drafting stage) and with the architecture work of ETSI TISPAN and of 3GPP. A need to assess the implications of real NGN and hybrid technology network architectures on network performance and QoS was asserted. Network evolution (NGN, FMC, 3G and WiMAX) is developing in a way which has implications for the quality observed by the end user but is not yet captured in standards. The present document discusses some aspects that may not, in fact, be suitable for regular standardisation as they may be transitional or proprietary.

The present document forms part of STQ's roadmap with respect to Quality aspects of NGN and relates directly to the subject of End to End QoS, one of the provisional list of strategic topics for 2009 contained in the ETSI Board report to GA#51.

4.2.2 Market impact, benefits to be gained

It is the intention that the present document will lead to improved standards on guaranteed end-to-end QoS which will facilitate the take-up of NGN services by allowing differentiation between NGN and traditional Internet service offerings. The present document may also have an impact on NGN architectural design and related standards.

4.2.3 Stakeholders

The present document is addressed to:

- Network operators and service providers;
- Users;
- Manufacturers; and
- Regulators.

4.2.4 Relation with other SDOs or ETSI TBs

- ETSI TISPAN and INT:
  The subject of QoS over NGNs is very closely related to the core activities of these two TBs.
- 3GPP:
  Ensuring good voice quality over NGNs is vital to the long-term success of Fixed Mobile Convergence (FMC) and to ensure users are satisfied with the quality of calls made from 3G to NGN environments.
- IEEE and ITU-T:
  The subject of QoS over NGNs is also related to some of the activities of these two (and many other) standards bodies. The present document may ultimately influence the direction of standards being developed by these bodies.

4.3 Further impacts

The content of the present document facilitates further standards work by:

- validating that the QoS and performance standards being produced are being applied in real-world NGN implementations;
- identifying important gaps in the current standards work programme which may be a barrier to NGN roll-out and the provision of good end-to-end QoS in multi-owner and multi-technology networks.

5 Overview of architectural elements in NGN

5.1 General

Figure 1 to 9 show the reference configuration for the discussions of end-to-end QoS issues in the present document. Figures 1 and 2 show the reference configuration for the transit segment. Figures 3 to 9 show reference configurations for various access arrangements.

Figure 1 shows a reference configuration where all transit segments are NGNs.

NOTE 1: Layer 2 transit paths may also be utilized within an NGN.

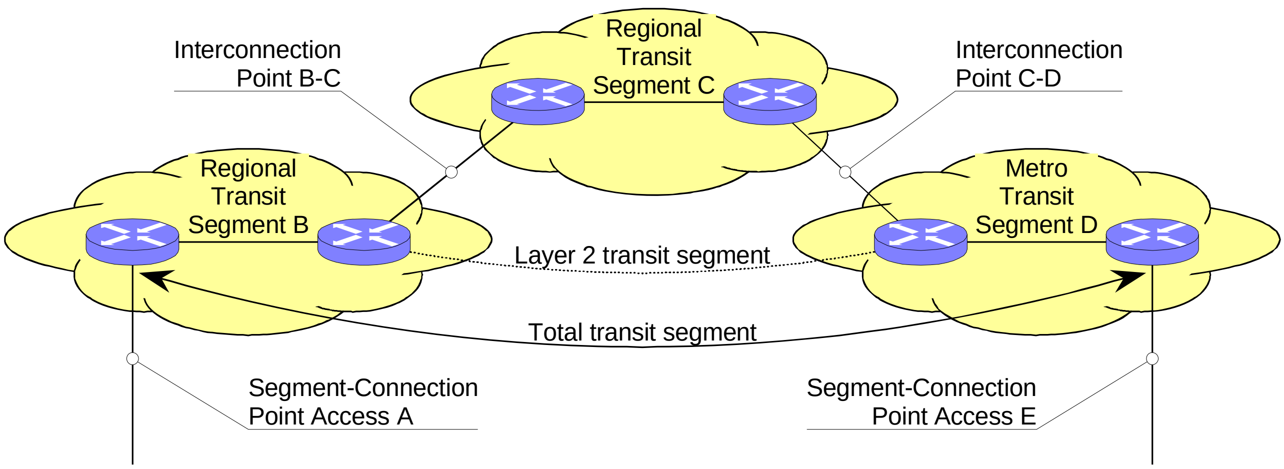


Figure 1: Reference configuration for QoS assessment of the transit segment (all NGN)

Figure 2 shows a reference configuration where the transition between NGN and TDM occurs within the transit segment.

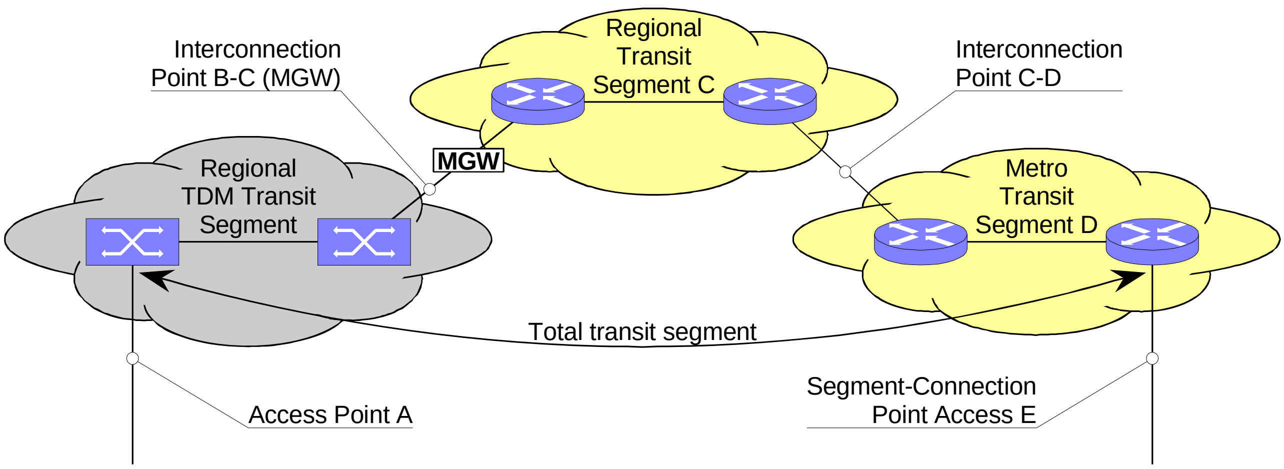


Figure 2: Reference configuration for QoS assessment (TDM transit segment)

Figure 3 shows a reference configuration where NGN technology is deployed up to the user equipment (UE). The UE, therefore, should implement:

1) In the outgoing direction:

   - coding of the voice signal (e.g. G.711 [i.24] to [i.26], G.723.1 [i.27] to [i.30], G.729 [i.31] to [i.42], etc.);
   - packetization of the outgoing packets; and
   - handle echo control.

2) In the incoming direction:

   - manage the de-jitter buffer;
   - perform the Packet Loss Concealment;
   - decoding of the voice signal; and
   - handle echo control.

NOTE 2: The UE is not necessarily a voice terminal, it might be a facsimile terminal, a modem that communicates with a modem in a PSTN/ISDN, or an equipment that utilises the ISDN "64 kbit/s unrestricted" bearer service.

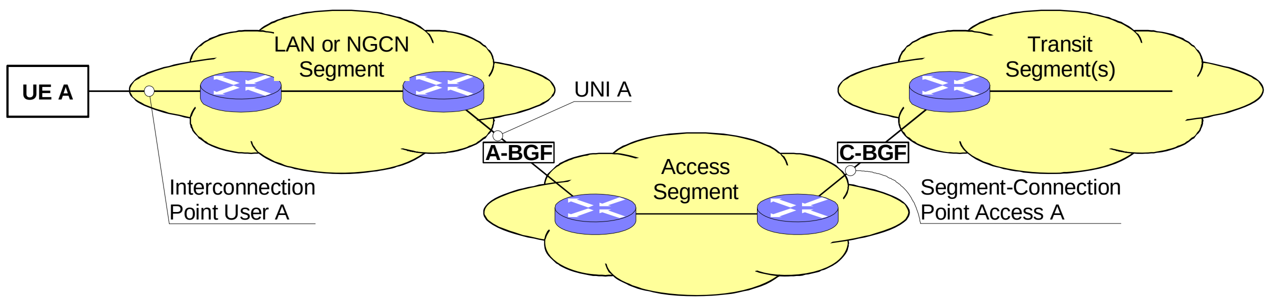


Figure 3: Reference configuration for QoS assessment (LAN or NGCN case)

Figure 4 shows a reference configuration where NGN technology is deployed in the corporate network (NGCN). Legacy UEs are attached to the NGCN via Access Gateway Functions (AGF); the "PABX functionality" is performed in the NGCN.

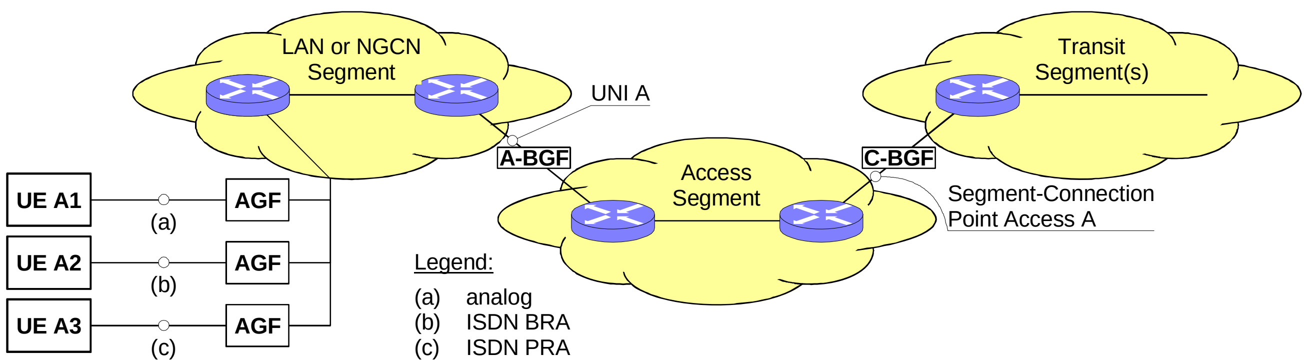


Figure 4: Reference configuration for QoS assessment (LAN or NGCN and legacy UEs)

Figure 5 shows a reference configuration where legacy PSTN/ISDN UE is handled by the last node in the transit segment.

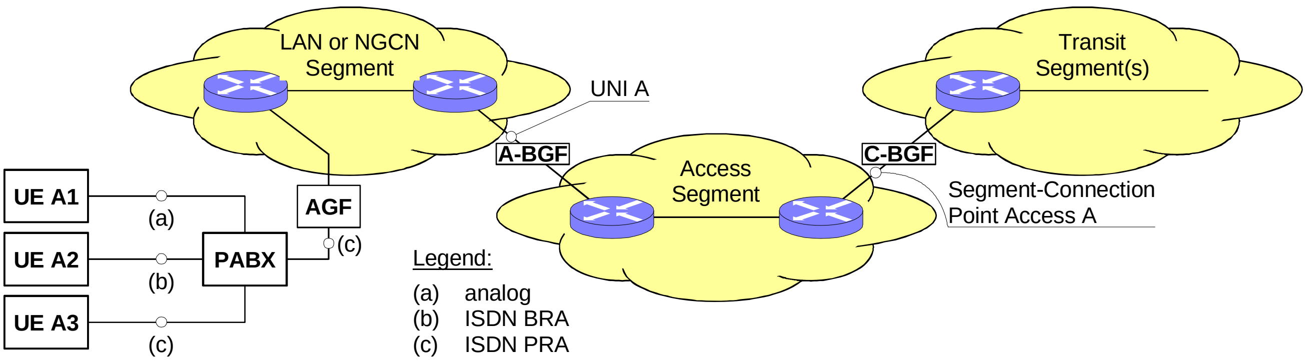


Figure 5: Reference configuration for QoS assessment (PSTN emulation/simulation with PABX)

Figure 6 shows a reference configuration where legacy PSTN/ISDN UE is handled by the last node in the transit segment.

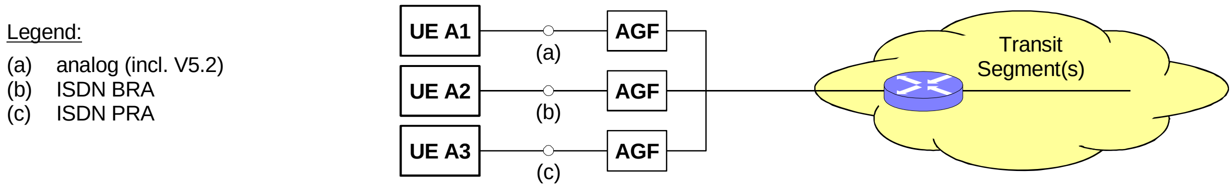


Figure 6: Reference configuration for QoS assessment (PSTN emulation/simulation in transit segment)

Figure 7 shows a reference configuration where legacy PSTN/ISDN UE is handled by the last node in the access segment.

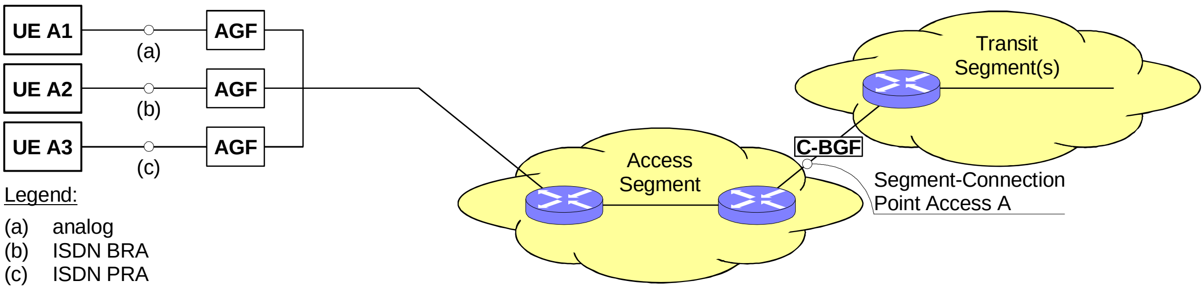


Figure 7: Reference configuration for QoS assessment (PSTN emulation/simulation in access segment)

From a point of view from QoS analysis, the reference configurations in Figures 2, 4, 6 and 7 will lead to the same analysis as the TDM part and/or the PSTN/ISDN part represent a lossless, jitter free segment with practically no additional delay.

5.2 Non-IMS Access to an NGN

This clause discusses the use of access solutions which are not standardized in the context of IMS. From an NGN point of view such access solutions are often also referred to as "proprietary" even though these access technologies are mostly standardized on their own right.

Figures 8 and 9 show typical scenarios for the deployment of non-IMS solutions. It shows the non-IMS part in the access between the PSTN/ISDN part and the NGN.

NOTE: The PSTN/ISDN segment may consist only of the cable that, for example, attaches a phone to a cable modem.

Example aggregation segments - where functionality beyond TDM is used - are:

- PSTN access via DSL;
- PSTN access via optical networks;
- PSTN access via broadband cable;
- PSTN access via WiMAX access segment.

Typically, in such access arrangements, voice is transported in packets similar to the transport used in NGN. The difference lies in the use of non-IMS transport of those packets - not via IP as in NGN, e.g. layer 2 frames might be used for the transport.

This packetization in the Aggregation Segment is not jitter free and might also suffer from packet loss. Therefore, for a QoS assessment the idiosyncrasies of the transport should be considered.

Two possible scenarios exist for the transition from the non-IMS aggregation to the NGN:

1) In Figure 8 is shown the case where the non-IMS aggregation and the first transit segment of the NGN belong to the same management domain. The Access Gateway of the NGN - for which all interfaces are fully standardised - is augmented for functions to bring the non-IMS protocol to the standard NGN; nevertheless, this makes the modified Access Gateway to a Non-IMS Media Gateway.

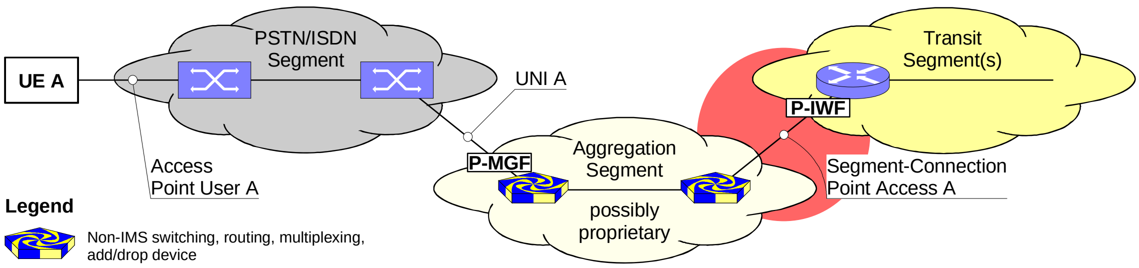


Figure 8: Reference configuration for QoS assessment (LAN or NGCN case)

2) In Figure 9 is shown the case where the non-IMS aggregation and the first transit segment of the NGN do not need to belong to the same management domain. The Non-IMS Media Gateway is located in the non-IMS aggregation and converts the media streams (and the associated signalling) to a standardised NGN access protocol. The media stream then interfaces to the standardised Core Border Gateway.

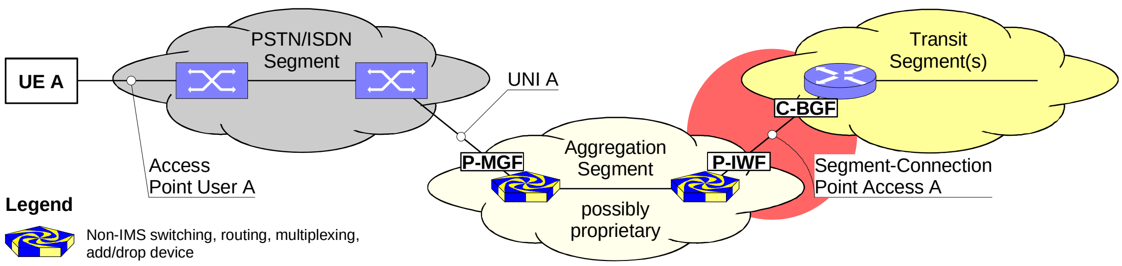


Figure 9: Reference configuration for QoS assessment (LAN or NGCN case)

5.2.1 Non-IMS Access Device

The Non-IMS Access Device (see Figures 8 and 9) converts ISDN/PSTN media streams and the associated signalling to the non-IMS methods and vice versa. The detailed functionality of the PAD might not necessarily be known by the general public. Depending on the configuration, several different PADs may exist in an aggregation, e.g. PSTN/ISDN, WiMAX codec, and cable modems.

NOTE: In Figures 8 and 9 only one sample PAD is shown.

The PAD should implement:

1) In the outgoing direction:
   - coding of the voice signal (e.g. G.711 [i.24] to [i.26], G.723.1 [i.27] to [i.30], G.729 [i.31] to [i.42], etc.);
   - packetization of the outgoing packets; and
   - handle echo control.
2) In the incoming direction:
   - manage the de-jitter buffer;
   - perform the Packet Loss Concealment;
   - decoding of the voice signal; and
   - handle echo control.

5.2.2 Non-IMS Media Gateway (P-MGW)

The Non-IMS Media Gateway (P-MGW) converts NGN media streams and the associated signalling to the non-IMS methods and vice versa. The detailed functionality of this conversion might not necessarily be known by the general public.

5.3 Transit networks

Due to real-world constraints the simplified division approach according to ITU-T Recommendation Y.1542 [i.23] 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 (see upper part of Figure 10).

NOTE 1: In Figure 10, the upper part displays the division of the connection as seen from a QoS point of view whereas the lower part shows this transit part in terms of the NGN Functional Architecture [i.4].The reference points Ic, Iw, and Iz are defined in [i.4] in clause 7.2.2.

NOTE 2: Interworking with non-compatible control domains is not addressed in the present document.

5.3.1 IP based Transit Network

If two NGNs are located in the same region or the two regions are close to each other this intra-regional transit between the NGNs often is performed by an IP transit (e.g. an Interconnection Server).

As the lower part of Figure 10 stipulates, the media stream and the signalling stream share the same IP transit infrastructure. The consequences of this mix are discussed in clause 5.4.

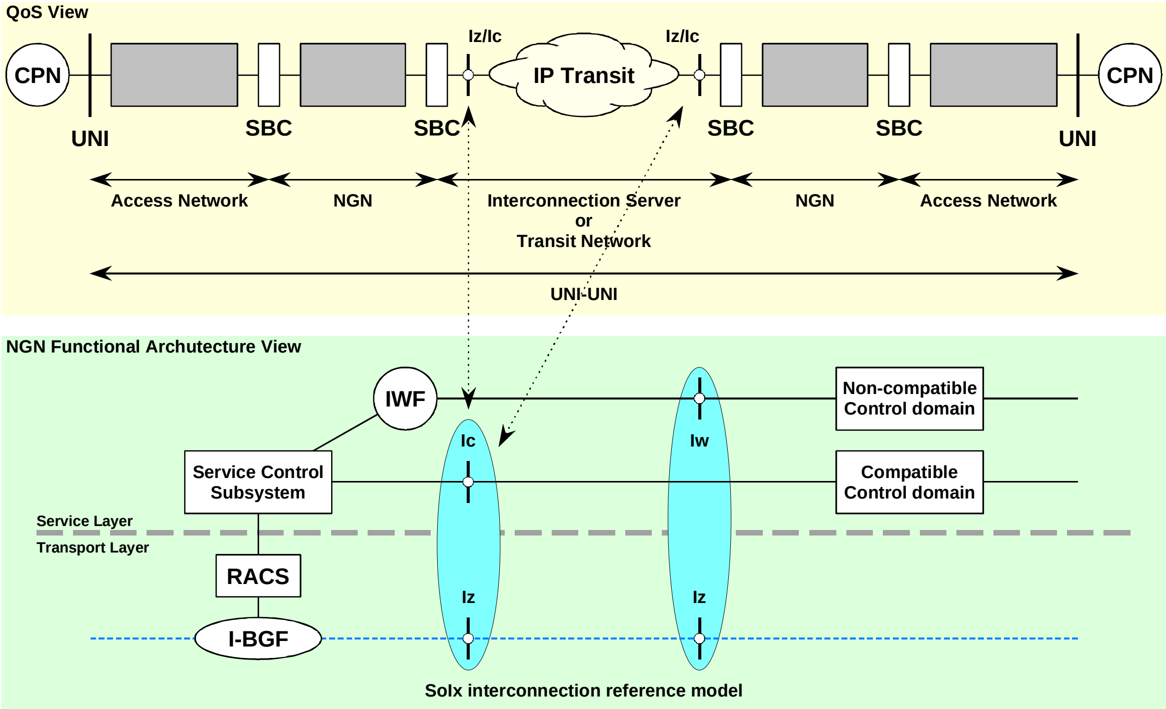


Figure 10: IP Transit

5.3.2 Layer 2 based Transit Network

For inter-regional transits often the deployment of a "Layer 2 Transit Network" is preferred (see Figure 11). Example of such Transit Networks are ATM, MPLS, Frame Relay, etc. The nature of this transport is the absence of any processing of NGN control information; even IP headers remain untouched. The consequences are that the mix of media and control streams remains unaltered also and only layer 2 delays (and possible loss of information) influences QoS matters; there exists no output and playout buffers as described in clause 5.4.

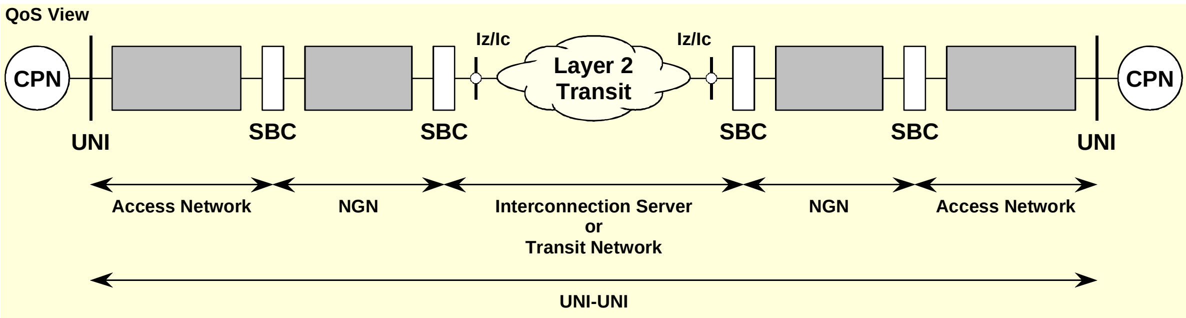


Figure 11: Layer 2 Transit

5.4 Playout Buffer

Figure 12 shows the typical structure of the output side towards the IP network in all devices connected to the NGN. This concerns servers, routers, NGN terminal equipment, etc. Packets to be transported are placed in an output buffer according to their priority. As an example, an NGN Telephone device might assign class 1 for signalling a high priority, class 2 for data a medium priority, and class 3 for statistics a low priority. The top priority, however, is assigned to realtime traffic.

NOTE 1: The output buffer is also known as "queuing buffer".

The size of the playout buffer needs at least to accommodate the largest packet of all priority classes. Usually, it is about 1½ times as large. As soon as the packet in the highest priority output buffer fits into the playout buffer, it is moved to the playout buffer.

NOTE 2: This algorithm does not include fair usage of the transmission resources.

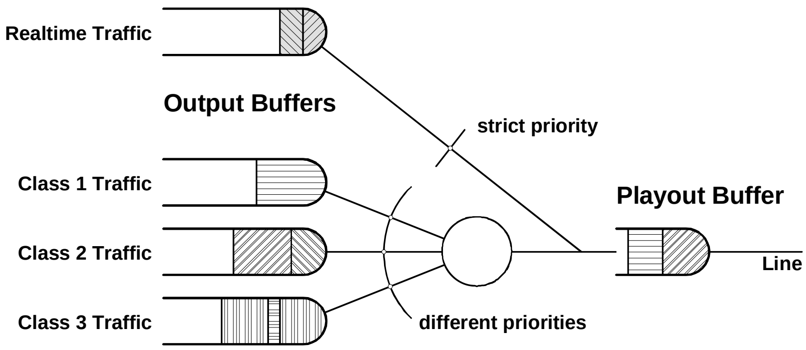


Figure 12: Playout Buffer

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. The de-jitter buffer at the receiving side should compensate the time which is needed when the packets leaves the output and playout buffers.

In addition, variable output queuing delay is added at every exit of routers and servers, e.g. by interleaving different streams or by the playout buffer mechanism described above.

5.5 De-jitter Buffer

From a de-jitter buffer's point of view, it has to adapt to the maximum end-to-end delay variation.

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

If the 50 ms nominal delay setting is used, the first voice sample received when the de-jitter buffer is empty is held for 50 ms before it is played out. This implies that a subsequent packet received from the network can be as much as 50 ms delayed (with respect to the first packet) without any loss of voice continuity. If it is delayed more than 50 ms, the de-jitter buffer empties and the next packet received is held for 50 ms before play out to reset the buffer. This results in a gap in the voice played out for about 50 ms.

NOTE 1: The 50 ms nominal delay setting is often deployed although it is not a standardized value.

The actual contribution of de-jitter buffer to the delay is the initial playout delay of the de-jitter buffer plus the actual amount the first packet was buffered in the network. The worst case is twice the de-jitter buffer initial delay (assumption is that the first packet through the network experienced only minimum buffering delay).

Figure 13 indicates that for other services, e.g. signalling, data, statistics, etc. no buffer at the input of the line is required from a functional point of view.

NOTE 2: The services may nevertheless need to buffer received information in case the information arrives before the previous information has been fully processed.

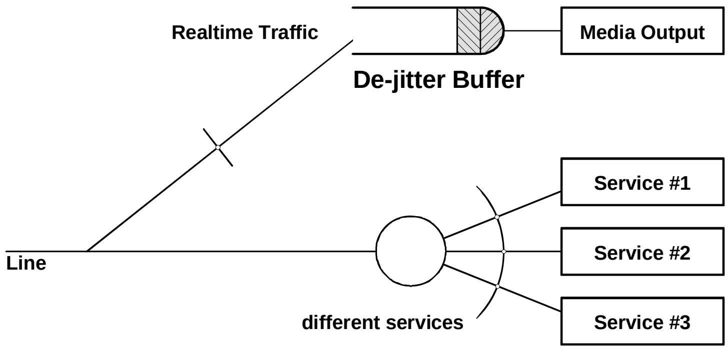


Figure 13: De-jitter Buffer

5.6 Terminals (user equipment)

There are no specific architectural considerations for terminals. However, from a QoS perspective VoIP and IMS voice terminals should comply with the following standards:

- ES 202 737 "Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user" [i.5];
- ES 202 738 "Transmission requirements for narrowband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user" [i.6];
- ES 202 739 "Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user" [i.7];
- ES 202 740 "Transmission requirements for wideband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user" [i.8];
- TS 102 737 "Transmission requirements for narrowband wireless terminals (handset and headset) from a QoS perspective as perceived by the user" [i.9];
- TS 102 738 "Transmission requirements for narrowband wireless terminals (handsfree) from a QoS perspective as perceived by the user" [i.10];
- TS 102 739 "Transmission requirements for wideband wireless terminals (handset and headset) from a QoS perspective as perceived by the user" [i.11];
- TS 102 740 "Transmission requirements for wideband wireless terminals (handsfree) from a QoS perspective as perceived by the user" [i.12];
- EN 300 175-8 V2.2.1 (2009-02): "Digital Enhanced Cordless Telecommunications (DECT);Common Interface (CI);Part 8: Speech and audio coding and transmission" [i.13]; and
- EN 300 176-2 V2.1.1 (2009-05): "Digital Enhanced Cordless Telecommunications (DECT);Test specification; Part 2: Audio and speech" [i.14].

5.7 Corporate Networks

There are no specific architectural considerations for Next Generation Corporate Networks; with regard to the QoS implications of their architecture they can essentially be treated similar as regular NGNs. Other aspects like security and privacy may differ but are out of scope of the present document.

5.8 Access Networks

There are no specific architectural considerations for the access. However, QoS impacts of the access are discussed in the course of the present document.

6 Reference Connections

Based on the reference architectures discussed in clause 5 reference connections for transmission planning have been developed and can be found in TR 102 775 [i.15].

7 Conclusions from STF 363

This clause discusses the general conclusions from STF 363 whereas the detailed findings on gaps in standardization are contained in the following clause 8.

- Many operators show a high interest in providing good end-to-end QoS to their customers.
- It seems to be with large difficulties for operators to collect or to communicate all QoS relevant information.
- In many cases no proper end-to-end transmission planning for NGN is in place.
- As far as replies to the questionnaire have been given by the operators it gives the impression that operators do not adhere to a common set of requirements and/or standards in order to obtain end-to-end QoS. This indicates that obtaining end-to-end QoS between users of different operators' NGNs might need further investigation / standardization.
- Fax-over-IP (FoIP) and Modem-over-IP (MoIP) constitute serious problems in the transition to NGN with the exception of areas where customers did not use such services in the past.
- Characterization of VoIP edge devices and gateways over a wide range of network conditions is rarely used as optimization approach.
- Synchronous Ethernet as per ITU-T Recommendation G.826x-series [i.91] to [i.94] is required but mostly not implemented.
- As the xDSL access (IAD) is usually not synchronized, "64 kbit/s unrestricted" (transparent mode) cannot be offered on DSL access lines.
- Those operators who have already deployed PSTN emulation do all not support ISDN "64 kbit/s unrestricted" service, which is required for example for the following applications:
  - Video conferencing systems (conventional).
  - Radio and TV Channels Transmission.
  - Secure data transfer between banks.
  - Data transfer between hospitals.
  - ISDN Fax group 4.
  - Remote maintenance of PBX.
  - LEONARDO (proprietary Apple application for the transport of nx64 kbit/s).

Conclusions on issues other than QoS can be found in Annex C.

8 Gaps identified in Standardization

This clause lists the gaps in standardization identified by STF 363 with respect to the QoS implications of NGN architectures. Each clause describes one of these gaps plus possible future actions.

8.1 Task depending prediction of Interaction Quality as perceived by the User

In many cases network operators have substantial problems to determine the total end-to-end delay values to which their customers are subjected. This is caused by different issues:

- complexity of delay composition in an IP environment
- lack of knowledge of the delay introduced by 3rd party vendors' equipment
- limited or no control over customer owned devices; e.g. phones, HGWs

Furthermore, network operators encounter problems in determining how much delay (under echofree conditions) their customers are prepared to happily accept. This again is caused by different issues:

- there are no longer hard requirement in ITU-T Recommendations or other standards
- the perception of delay impairments is clearly task-dependant
- delay measurement between acoustical interfaces may be feasible in a lab environment but they are clearly inappropriate for a large number of connections in the field, including interconnection scenarios and user-owned equipment.

It is proposed to start work in ETSI STQ on a methodology to objectively estimate (or predict) the quality of interaction of a real conversation as perceived by the user(s).

It is believed that such an objective methodology could proof extremely useful for network operators if it provides an indication in terms of a kind of traffic light instead of a numeric value.

One possible application of such an indicator is the ETSI Electronic Meeting conference bridge; green traffic lights could indicate that remote participants do not encounter any disadvantages.

8.2 Detection of Malfunctioning Speech Processing Devices, like Echo Cancellers

No standard is yet available that would network operators enable to check whether speech processing devices like echo cancellers, comfort noise generators, etc. work properly or malfunction in everyday's operation in the field. There are currently only proprietary solutions in use.

However, it is believed that it might be useful to define respective parameters at the functional interfaces between different network segments and related testing methodologies.

8.3 Identification of QoS Compliant VoIP Terminals and gateways by a common identifier

Network operators are faced with the challenge that in most cases they are lacking control of the terminals installed at their customers' premises and their impact of such terminal on end-to-end QoS.

It is proposed to standardize a mechanism by which manufacturers of VoIP terminals which comply with STQ's high quality standards, such as:

- ES 202 737 "Transmission requirements for narrowband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user" [i.5];

- ES 202 738 "Transmission requirements for narrowband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user" [i.6];
- ES 202 739 "Transmission requirements for wideband VoIP terminals (handset and headset) from a QoS perspective as perceived by the user" [i.7];
- ES 202 740 "Transmission requirements for wideband VoIP loudspeaking and handsfree terminals from a QoS perspective as perceived by the user" [i.8];
- TS 102 737 "Transmission requirements for narrowband wireless terminals (handset and headset) from a QoS perspective as perceived by the user" [i.9];
- TS 102 738 "Transmission requirements for narrowband wireless terminals (handsfree) from a QoS perspective as perceived by the user" [i.10];
- TS 102 739 "Transmission requirements for wideband wireless terminals (handset and headset) from a QoS perspective as perceived by the user" [i.11]; and
- TS 102 740 "Transmission requirements for wideband wireless terminals (handsfree) from a QoS perspective as perceived by the user" [i.12];
- EN 300 175-8 (V2.2.1): "Digital Enhanced Cordless Telecommunications (DECT);Common Interface (CI);Part 8: Speech and audio coding and transmission" [i.13]; and
- EN 300 176-2 (V2.1.1): "Digital Enhanced Cordless Telecommunications (DECT);Test specification; Part 2: Audio and speech" [i.14].

could enter a common identifier, e.g. the MAC address ranges of mentioned terminals into a specific data base which is accessible to network operators. The network operators in turn could then use the information whether the terminal complies with the high quality standards to determine the level of support for each customer enquiring at customer support contact centres. At the end, this solution could be used to apply resources specifically towards increased end-to-end quality as perceived by the user.

8.4 Dynamic vs. Fixed Delay Allocation

As outlined in GSMA Document IR.34 "Inter-Service Provider IP Backbone Guidelines" [i.44] the delay values for the total transit segment are in a fixed relation to the distances between different geographical regions. Thus, for the near future dynamic allocation of delay budgets as described in the framework document of ITU-T SG12, Recommendation Y.1542 [i.23] is not expected to be implemented between user segments, access segments and transit segments. Consequently, more detailed ITU-T Recommendations on fixed delay allocation are desirable.

8.5 Distribution of relevant core network QoS parameters

There is a need to better standardize the distribution of relevant core network QoS parameters between interconnected providers of the core network (total transit segment). Better QoS-awareness results in better QoS provisioning. The operators are using different QoS strategies, which are requiring different QoS information (parameters). The better QoS-awareness can help the better coexistence of different QoS approaches (deployed at different providers) and finally can improve end-to-end QoS in the interconnected telecommunications networks. For this purpose, the relevant QoS parameters have to be updated. For instance, parameters like equipment impairment factor (Ie), related information about speech processing devices, de-jitter buffer information, etc have to be added. Since ITU-T Study Group 16 (Q.18/16) [i.78] has started work on a related Recommendation G.MDCSPNE "Mechanism for dynamic coordination of SPNE to achieve optimal end-to-end voice quality" it is suggested to monitor and contribute to this ongoing work as appropriate.

8.6 Impact of xDSL (ADSL) bandwidth limitations

The impact of xDSL bandwidth on delay variation is well known by experts but has not yet been described in standards. Since this is a pre-requisite for proper transmission planning by network operators STF 363 made an initiative towards ETSI STQ who started immediate work on this topic which is to be found in draft TR 102 720 [i.74] "Delay variation on access lines" with the following scope: "Show the effects of bitrate of unshared access lines on delay variation of high priority real time services. This work will include study of the effects of packet prioritization mechanisms".

8.7 Signalling Packet Size Impact

Upon the initiative of STF 363 the impact of the signalling or coexistence service (web browsing, etc.) packets on voice packets in xDSL access will be described in draft TR 102 720 [i.74]. It can be seen that the packets mentioned above have a big impact on voice packets. This problem also exists in case of other type of access lines and was described in the scientific literature [i.48], [i.49]. Some of the results have been also presented in ETSI STQ workshop on Effects of transmission Performance on Multimedia QoS in Prague. In this case the optimal size of signalling and coexistence service packets has to be defined for optimal performance purposes, especially in case of low bandwidth xDSL and wireless access lines. Otherwise VoIP can be restricted only to high bandwidth access lines or some kind of wireless access technologies only, as also partly concluded in draft TR 102 720 [i.74].

8.8 Hypothetical Reference Connections for Delay

Currently, ITU-T Recommendation G.103 (1998) [i.43] is the only standard describing hypothetical reference connections (HRCs) which have been proven to be useful for network operators or administration when determining network topologies or national transmission plans. Figure 14 shows as an example "The longest ISDN international connection likely to occur in practice".

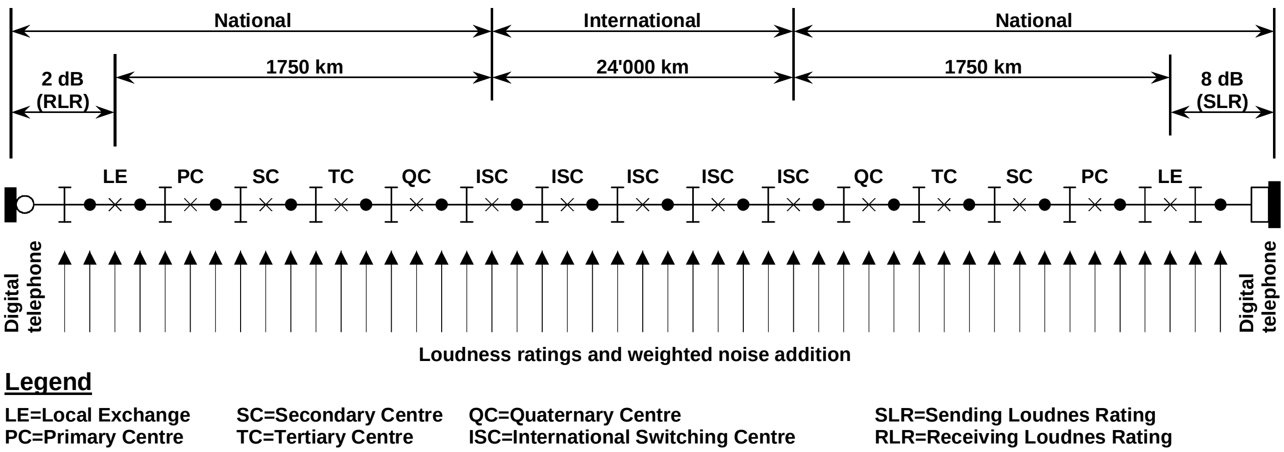


Figure 14: Example HRC from [i.43]

In contrast, Figure 15 shows a typical reference connection from TR 102 775 [i.15] reflecting the state-of-the-art considerations for transmission planning in NGN.

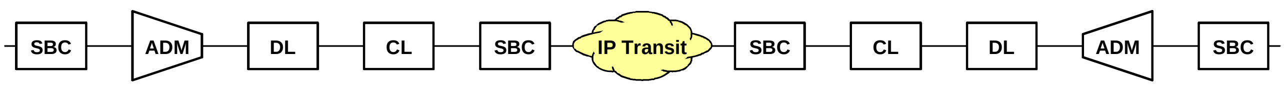


Figure 15: Reference connection for transmission planning

As can be seen, the focus in ITU-T Recommendation G.103 [i.43] is limited to noise and loudness ratings, whereas delay, delay variation, packet loss and transcoding are major concerns in NGN transmission planning. Therefore, it is suggested to support the ITU-T in updating ITU-T Recommendation G.103 [i.43] in order to provide proper HRCs for delay planning in the NGN taking into account the other factors mentioned.

8.9 Motivation of Customers to Use the Supported High Quality Terminals

Various network operators offer preferred terminals in their shops to the end customers. This could support the goal of achieving high end-to-end quality as perceived by the user if mentioned terminals would be compliant with the standards quoted in clause 8.3.

However, it is not yet clear how customers could be motivated to follow this approach and use said preferred terminals. It is suggested that ETSI HF have a closer look into this problem space.

8.10 NGN Update of E-model Approach in EG 202 057-2

In clause 5.3.3 as well as in Annex I of EG 202 057-2 [i.18] an implementation of the E model has been described which is referred to as "design figure of merit". However, this implementation applies to PSTN/ISDN networks, only. It is suggested to start work to update this modelling approach to make it compatible with NGN specific parameters.

8.11 Concept for NGN migration

Currently there is no guidance provided by standards documents how a proper migration strategy from PSTN/ISDN toward the NGN could look alike. Such a guide could proof extremely useful for network operators and service providers. It is suggested to start work along the following lines; a cooperation with the ITU-T might be appropriate:

1) Planning:

    - a fair practice: evaluate your products and services; which should be migrated/continue in legacy network/discontinued?
    - consider the meaning of distributed architecture:
      security (across multiple products, platform and vendors - consistent approach to secure design?);
    - performance (greatly varying demands between different services); and
    - resiliency;
    - organisational readiness: not only the tech-guys!
    - involve external stakeholders (enterprise customers, SME, wholesale customers, regulator).

2) Implementation:

    - Be serious about interoperability testing - allow time for element-to-element function verification.
    - Repeat this step for every major future revisions.
    - Allocate resources necessary for development of updated revision control and documentation programs.
    - All to be handed over to operations before "going live".

3) Migration and transition:

    - Make public all planned changes/updates of existing technical interfaces. Important for both end-users and wholesale customers.
    - Conduct "feature parity" considerations.
    - Allocate resources to insure the data quality is reliable across all sources for the migration program (provisioning systems, billing systems, circuit inventory databases, switch databases, etc.).
    - Be prepared for, and do not underestimate, the logistical complexity of migrating potentially millions of customers.
    - Develop (and test) roll-back plans, should problems arise.

4) Operations:

- What is the operational impact of the transition period itself?
- Make sure updated skills are at hand (to support multiple services delivered over multiple access networks).
- All-IP: from a network-based to a service-based management model?

8.12 Use of Technical Data in User Profiles

Users' expectations could be better fulfilled by pre-defined profiles. More about that can be found out in STF 360 reports and related documents. It is suggested to investigate whether in a further step - after the work of STF 360 has been completed - technical data, such as delay or quality classes, could be considered for integration into mentioned user profiles.

8.13 Psychological aspects of Telecommunications

It was postulated in related scientific literature that people are changing their communication habits. NGN is also one of the main reasons for doing that. Today communications technologies are important part of our lives and are bringing us mobility and freedom to daily life. Currently, we can communicate with our relatives or business partners from everywhere and any time. To overcome some problems related to roaming across different access networks, the researchers are trying to design the vertical handover approach [i.53] and [i.54]. The afore-mentioned facts have an impact on user behaviour. In some cases, the users are able to adjust to new transmission condition and can tolerate some level of impairments. This behaviour was not obtained before and main reason for such a change is a ubiquity of communication technologies. However, current standards do not provide sufficient guidance for network operators and service providers on this important issue; every operator or provider has to rely on his own best guess in this regard. Therefore, it is proposed to start work in ETSI in this direction.

8.14 Conversational Scenario related Quality

It is very important to distinguish the type or the purpose of communication because that has finally big impact on user's expectation and impressions. For instance if users are realizing a social call (free conversation between two friends or relatives), they are much more tolerant to some influences as well as the call pattern is quite different in comparison with the business or task-oriented call (getting travel or weather information, etc.). In second case (business and task-oriented calls), the tolerance is much smaller because users would like to get important information, like a project schedule information, invoiced amount of money, etc. That was also main aim of that call. On the other hand, not so much important information are mainly exchanged during social calls and users are not under pressure like in case of business calls, they are more relaxed. The aim of such a call is mainly held some kind of social connection. That means this behaviour has to be taken into account in case of transmission planning as well as in case of real measurements. Therefore, it is proposed to start work in ETSI in this direction. Some preliminary studies have been published in scientific literature [i.50], [i.51], [i.52], especially from call pattern (Active-Speech-Ratio) perspective.

8.15 Wideband telephony over xDSL (low rate access lines)

One of the widely promoted advantages of the move towards NGN are the introduction of new services, including wideband, super-wideband and full-band telephony (3-dimensional telephony is also a research topic). However, with low speed xDSL access these promotions are being undermined. But the issue could be resolved by investigation of other access technologies, e.g. fiber to the home, wireless access or new inventions.

It is suggested to start standardization work on reference transmission planning scenarios for mentioned cases.

8.16 De-Jitter Buffer Strategies

For network operators who are willing to prepare sound and solid transmission plans for their migration to NGN it is always extremely difficult to know how to consider de-jitter buffers in mentioned transmission planning.

To resolve that issue it is suggested for ETSI to start work on the standardization of de-jitter buffer strategies. ETSI should do the first step and provide a Deliverable that probably could outline the various options for this topic and the respective considerations for the NGN transmission planner. Such a document could successfully establish a useful basis for information exchange between manufacturers and network operators.

8.17 Standardized Interface for VoIP / IMS Terminals

The traditional TDM UNIs (User-Network Interfaces), i.e. analogue and ISDN S-Bus, are well standardised since many years. It is no problem to buy a phone off the shelf, connect it to a public telephony access, and it works (there may be some quality issues, but it is possible to make basic calls and even use supplementary services in most cases).

In contrast the situation for VoIP and IMS endpoints is a gallimaufry:

- There are still several protocols in parallel use (SIP, H.323, etc.)
- There is a difference between "simple" SIP and SIP for IMS: There are more strict requirements for SIP parameters for IMS than for other VoIP systems (and also special SIP private headers for IMS, e.g. PAI)
- Some SIP parameters are not well defined, e.g. for q-value (used for order of ringing for different SIP endpoints with the same phone number) different interpretations are possible
- Some concepts are insufficiently standardised, e.g. for P-CSCF discovery a RFC exists, but it does not work across domain borders (P-CSCF discovery is used to know where to send the SIP-messages)
- It is not only about SIP: it can also be about DNS (for SIP) or HTTP or FTP or….(in the language of IMS, it is not only about the Gm reference point, it is also about the Ut reference point)
- CPE manufacturers seem not yet to be ready for introducing IMS capable VoIP-CPEs

There is an urgent need for a standardised UNI, including:

- Minimum SIP/SDP requirements for basic services
- Minimum SIP/SDP requirements for supplementary services
- Address discovery in a redundant environment
- Minimum supported codec list
- OAB interface (One Address Book)
- Remote configuration (ETR 069 [i.75])
- Minimum speech quality requirements (for IP Phones and Home Gateways)

A standardised API (for an interworking module per network operator) could be an alternative to a standardised UNI.

8.18 Bundling of n x 64 kbit/s channels

Figure 16 illustrates the transport of n 64 kbit/s TDM channels via the NGN packet network. As can be seen each channel is treated by a separate de-jitter buffer at the receiving side. However these de-jitter buffers work independent of each other which results in the loss of frame synchronization due different queuing delay and in the loss of frame synchronization due packet loss. Standards on Synchronisation of de-jitter buffers for the transmission of bundled 64 kbit/s transparent data channels are urgently needed.

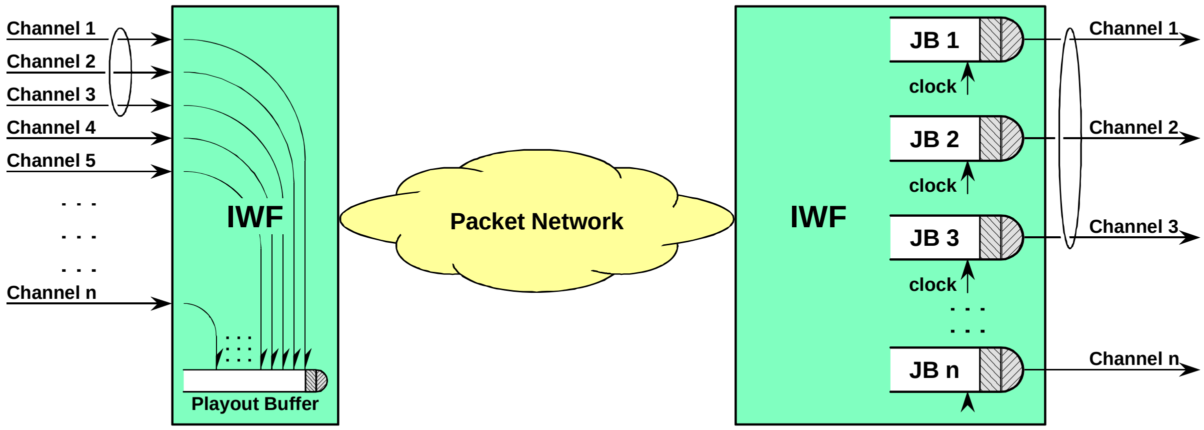


Figure 16: Transmission of bundled TDM channel via the NGN

8.19 Perceptual Impact of End-to-End Delay and Delay Variation on Fax-over-IP and Modem-over-IP

When migrating from traditional TDM based networks towards the NGN, knowledge on the perceptual impact which end-to-end delay and end-to-end delay variation have on Fax-over-IP (FoIP) and Modem-over-IP (MoIP) needs to be available. In addition existing quality metrics for Fax and Modem have to be reviewed and improved; This is required for the achievement of the strategic goals of TISPAN NGN release 3. In addition, the Fax service is in some European countries the only legally accepted way of "electronic" document transfer and in some European countries part of the Universal Service. STF 363 made an initiative towards ETSI STQ who started immediate work on this topic which is to be found in draft TR 102 719 [i.60] "Perceptual Impact of End-to-End Delay and End-to-End Delay Variation on Fax-over-IP (FoIP) and Modem-over-IP (MoIP)" with a scope based on the afore-mentioned reasons.

8.20 Transmission Requirements for IP-based Home Gateways and Other Media Gateways from a QoS Perspective as perceived by the User

The need for establishing IP gateway transmission requirements from a QoS perspective as perceived by the user is well known for some time; a respective study question in ITU-T SG12 [i.77] was lacking contributions from the industry since it had been established. ETSI STQ, however, have established some test methodologies for IP gateways in the context of the ETSI Speech Quality Test Events and Plugtests [i.55] to [i.57]. Also transmission requirements from a QoS perspective as perceived by the user are available [i.5] to [i.8].

There is an urgent need to establish such a standard, since at least one organization has already begun to provide a certification service for "Good Voice Quality" for home gateways [i.58].

STF 363 made an initiative towards ETSI STQ who started immediate work on this topic which is to be found in draft new ES 202 718 [i.61] "Transmission Requirements for IP-based Home Gateways and Other Media Gateways from a QoS Perspective as perceived by the User" with the following scope: "Define test conditions, test procedures, requirements, access network impairments and relevant properties of connected equipment at the endpoints required to guarantee a good speech quality as perceived by the users."

8.21 Revision of TR 102 775 (V1.2.1)

It was detected by STF 363 that version 1.2.1 of TR 102 775 [i.15] needed urgently to be updated in order to add 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. Consequently, a new largely extended draft has been elaborated by STF 363 in close cooperation with STQ.

8.22 Non-Tandeming of Speech Processing Devices

Tandeming of speech processing devices like echo cancellers or comfort noise generators can severely impact the end-to-end speech quality perception of users.

At various occasions it had been pointed out that it would be highly desirable to establish a signalling approach which can help to avoid such tandeming [i.59]. However, even though it can be seen as an issue requiring urgent action, this is still an open point. Since ITU-T Study Group 16 (Q.18/16) has started work on a related Recommendation G.MDCSPNE "Mechanism for dynamic coordination of SPNE to achieve optimal end-to-end voice quality" it is suggested to monitor and contribute to this ongoing work as appropriate.

9 Regulatory aspects

This clause describes observations of regulatory aspects which were made during the course of STF 363 and are here provided as additional information.

9.1 Universal Service Obligations for xDSL Access

The Finnish Regulator has chosen to require a minimum speed of 1 Mbit/s for xDSL access lines to each customer, as a universal service approach. It is without prejudgement of political and economical dependencies that one can conclude that such a solution has the potential to resolve the QoS (delay variation) problems which are encountered in low speed xDSL access lines. Despite the volume of the resources required to comply with mentioned obligation it supports various of the policy elements of the European Commission which depend on broadband access with satisfactory quality for All.

9.2 Users' perception of End-to-End QoS and the Market

It has been postulated for a long time that the forces of the Market will bring the Users into a position where they can obtain the end-to-end QoS they wish to perceive. This assumption, however, has been proven to be more than questionable. In general, contracts between users and network operators have a longterm binding character and do not contain any QoS provisions or even guarantees.

The work of STF 360 is taking "Contracted QoS" as an input parameter for their work, but we believe that such a parameter is available in exceptional cases only.

As indicated in the previous clause, network operators are lacking a knowledge of the end-to-end QoS they actually provide. Given that their customers are lacking the possibility to switch their contract to another operator on short notice, there is no alternate way for customers to employ the forces of the market to support their demand for high end-to-end QoS.

Consequently, it is suggested to investigate the possibilities of appropriate regulatory interventions in cases where customers of NGN operators are extremely unsatisfied with the end-to-end QoS provided to them; this might include issues of QoS contracting, QoS reporting or Terminate Early (TME) options for the customer contract. In this context new standards may be necessary for users to proof the level of QoS perceived.

9.3 Minimum R-value (E-model) for IP telephony

Some Regulators have set out an obligation for network operators to meet a minimum R-value besides delay and delay variation requirements when the current ISDN / PSTN is being migrated towards NGN [i.45], [i.46].

Whereas this may be seen as a proper attempt to protect the customer, it is not without problems since adequate transmission planning for NGN is currently not in place and the "design figure of merit" modelling approach have not yet been adapted to NGN (see clause 8.10).

9.4 Review of existing ETSI Guides used for regulatory purposes

It is desirable to investigate how, EG 201 769-1 [i.16], EG 202 057 [i.17] to [i.20] and EG 202 009-2 [i.21] can best be further developed so that they are also applicable to NGNs.

The development of EG 202 057 should have the highest priority; as it was indicated during the STQ workshop on QoS Implications of NGN Architectures in July 2009 this document is commonly used as a reference.

Annex A: Questionnaire: NGN Architectures deployed or planned

Definition of Quality of Service for the purpose of this Questionnaire:

For the purpose of the work STF 363 and for this questionnaire, Quality of Service is defined as the QoS as perceived by the user, as it is outlined in ITU-T Recommendation G.1000 clause 5.5.4.

ETSI has started an activity (known as STF 363) on end-to-end quality of service implications of planned and existing Next Generation Network and hybrid technology network architectures including mobile architectures. The target is to provide recommendations on new standards needed and changes required to existing standards. The focus will be on voice quality, but other applications may also be considered.

In this context it is important to acknowledge that terminals and access networks are clearly parts of the NGN architecture as well as PSTN emulation and simulation.

In order to achieve the goal outlined above, STF 363 has prepared the following questions for the evaluation of real-world architectures. Please, answer as many questions as possible and give details where available. Your contribution is highly appreciated.

NOTE: If you cannot answer a question or do not wish to share the respective information, please leave the field empty and tick neither box.

Company:

Name:

Job Function :

Is the information provided in the following confidential?

☐ No ☐ Yes ☐ Only in the specific sections:

A.1 Core Network

A.1.1 How do you avoid concatenation of geographical distances in cases of multiple interconnections for one call?

A.1.2 Budget for packet loss, packet delays?

A.1.3 Which routing protocols do you use?

A.1.3.1 RIP?

A.1.3.2 OSPF?

A.1.3.3 others, please specify.

A.1.4 Which technologies are used in the core?

A.1.4.1 WDM?

A.1.4.2 SDH?

A.1.4.3 ATM?

A.1.4.4 MPLS?

A.1.4.5 Frame Relay?

A.1.4.6 IP?

A.1.5 Which technologies do you deploy to avoid multiple transit networks?

A.1.6 Convergence time at routers - how much time is spent per re-routing?

A.1.7 Have you implemented one of the methods described in ITU-T Recommendation Y.1542 in order to ensure the compliance with ITU-T Recommendation Y.1541?

A.1.7.1 Static approach?

A.1.7.2 Pseudo-static approach?

A.1.7.3 Signalled approach?

A.1.7.4 Impairment accumulation approach?

A.2 Network Interconnection

A.2.1 Which technical parameters you may have defined in a SLA, please specify.

A.2.2 Have you deployed voice enhancement devices (VED) in your network? Yes/No

A.2.2.1 How do these VED interwork if similar functions are implemented in the mobile station, please specify.

A.2.3 Have you deployed IP-to-IP gateways? Yes/No

A.2.3.1 Do your IP-to-IP gateways impact the voice signal by one the following functions?

A.2.3.1.1 De-jitter buffer, please specify.

A.2.3.1.2 Transcoding, please specify.

A.2.3.1.3 Re-packetization, please specify.

A.2.3.2 Are these functions activated for both directions of transmission?

A.2.3.3 How much additional delay is introduced by your IP-to-IP gateways?

A.2.4 Which QoS parameter values can you (technically) make available at interconnection points?

A.2.4.1 Delay?

A.2.4.2 Delay variation?

A.2.4.3 Equipment impairment factor?

A.2.4.4 Other, please specify.

A.3 End-to-End Performance Objectives

A.3.1 Do you have any end-to-end performance objectives? Yes/No

A.3.1.1 In terms of Mean Opinion Scores (MOS)?

A.3.1.2 In terms of post dial delay (PDD)?

A.3.1.3 In terms of other parameters, please specify

A.3.2 Does your PSTN emulation support ISDN clearmode service? Yes/No

A.3.2.1 By which kind of end-to-end technology is this being achieved?

A.3.2.1.1 Are you using or planning to implement Synchronous Ethernet as per ITU-T Recommendation G.8261

A.3.2.1.2 Are you using or planning to implement packet based methods, e.g. as per IEEE 1588 (TM)?

A.3.3 Does your PSTN emulation support certain dial-up modems (often used for alarm systems or telecare services). These modems require the use of a static dejitter buffer. Some terminals with these modems are built to expect only a few tens of milliseconds of end-to-end delay as seen in the PSTN. Greater delay than expected can result in false alarms when the terminal does not receive a response in time. Problems can also occur when accumulated jitter causes dejitter buffer overruns)? Yes/No

A.3.3.1 By which kind of end-to-end technology is this being achieved?

A.3.3.1.1 Are you using or planning to implement Synchronous Ethernet as per ITU-T Recommendation G.8261

A.3.3.1.2 Are you using or planning to implement packet based methods, e.g. as per IEEE 1588 (TM)?

A.3.4 Does your network architecture provide for unrestricted codec negotiation between user endpoints? Yes/No

A.3.4.1 During call set-up, please specify.

A.3.4.2 During the call itself, please specify.

A.3.5 Which approach do you employ in the case of customer's satisfaction evaluation?

A.3.5.1 End-to-end real networks measurements by accomplished intrusive algorithms, like P.862, etc.?

A.3.5.2 Questionnaire or survey?

A.3.5.3 Non-intrusive approach at some measurements points, like P.563?

A.3.5.4 Other approach, please specify:

A.3.6 Do you support end-to-end codec type/mode negotiation? Yes/No

A.3.7 Do you support Tandem Free Operation? Yes/No

A.3.8 Are you using call admission control as means to guarantee QoS? Yes/No

A.3.8.1 Where are these procedures implemented, please specify?"

A.3.9 Are you using Diffserv? Yes/No

A.3.9.1 In which network segment, please specify:

A.3.9.2 How many classes are actually used, please specify

A.3.10 Are you using or planning to use all or part of the ETSI TISPAN RACS architecture or any similar architecture to enforce QoS?, please specify:

A.3.11 What are your requirements in term of Call Setup delays, please specify:

A.3.12 What are your requirements in terms of equipment resilience / fault-tolerance, please specify:

A.4 Access Networks

A.4.1 Which access technologies do you use for NGN?

A.4.1.1 Fixed, please specify.

A.4.1.2 Wireless, please specify.

A.4.2 Which access technologies do you use for PSTN emulation, please specify:

A.4.3 For wireless access, what is the maximum number of user connected to one wireless access point or base station?

A.4.4 Do you use any advanced techniques to facilitate handover, roaming or cooperation with other types of networks? please specify.

A.4.5 Which is the minimum gross bandwidth (on the application layer) in the access you allocate to each channel of voice communication over NGN?

A.4.6 Do you use any optimization approach between the network and the terminals? Yes/No

A.4.6.1 RTCP-XR?

A.4.6.2 Characterization of VoIP edge devices and gateways over a wide range of network conditions?

A.4.7 Which length of packets and transmission rates (on the IP layer) do you use for your different services (voice, IPTV; data transfer etc.)?

A.4.8 Do use any prioritization in access network? Yes/No

A.4.8.1 Please specify categories for each kind of technology.

A.4.8.2 Please specify the mapping of Class of Service values to the MAC layer parameters or queues:

A.4.9 Do you use any advanced settings of access technology?

A.4.9.1 DSL: Fast or interleaving channel or other techniques according to QoS provisioning?

A.4.9.2 WiFi: Type of access method on MAC layer or other techniques according to QoS provisioning, please specify.

A.4.9.3 WIMAX: Scheduling algorithm or other techniques according to QoS provisioning, please specify.

A.4.10 Do you employ cRTP in your network? Yes/No

A.5 Gateways (edge devices)

A.5.1 Can you provide technical details of the gateway implementations? Yes/No

A.5.1.1 Which types of gateways do you provide? Please specify per type the properties according to subsequent sections:

A.5.1.1.1 Trunking Media Gateways to adjacent ISDN?

A.5.1.1.2 Access Media Gateways for support of non-SIP/RTP UNIs?

A.5.1.1.3 Home Media Gateways installed at customers' premises?

A.5.1.1.4 IP-to-IP Gateways?

A.5.1.1.5 other, please specify:

A.5.1.2 Which codecs are implemented?

A.5.1.2.1 G.711? Yes/No

A.5.1.2.2 other narrowband, please specify:

A.5.1.2.3 wideband, please specify:

A.5.1.2.4 dynamic codec shifting (auto bandwidth), please specify:

A.5.1.2.5 which of these codecs is mainly used?

A.5.1.3 Do you adopt in your gateway a single codec approach (i.e. support only one special codec) or a multiple codec approach?

A.5.1.4 Is packet loss concealment (PLC) implemented and used? Yes/No

A.5.1.4.1 standardized method, please specify:

A.5.1.4.2 proprietary, please specify the principle

A.5.1.5 Can you specify the settings of the codec? Yes/No

A.5.1.5.1 How many frames are per packet?

A.5.1.6 Is a de-jitter buffer implemented in the gateway? Yes/No

A.5.1.6.1 Has the de-jitter buffer

A.5.1.6.1.1 a fixed size?

A.5.1.6.1.2 an adaptive size?

A.5.1.6.1.2.1 Which algorithm is implemented?

- reactive algorithm
- histogram based algorithm
- algorithm that monitors network
- algorithm that optimizes user satisfaction

A.5.1.6.1.2.2 On which principle is your algorithm based?

- per talkspurt
- per packet
- modified version of per talkspurt, please specify
- modified version of per packet, please specify

A.5.1.7 Are speech processing algorithms (e.g. NR, EC, AGC or VAD) implemented and used in the gateway? Yes/No

A.5.1.7.1 Which noise reduction (NR) algorithm is implemented and used, please specify.

A.5.1.7.2 Which echo cancellation (EC) algorithm is implemented and used, please specify.

A.5.1.7.2.1 For electrical echo?

A.5.1.7.2.2 For acoustical echo?

A.5.1.7.3 Which automatic gain control (AGC) algorithm is implemented and used, please specify.

A.5.1.7.4 Which voice activity detection (VAD) algorithm is implemented and used, please specify.

A.5.1.8 Is forward error correction (FEC) implemented and used in the gateway? Yes/No

A.5.1.8.1 Which additional delays are incurred by the FEC?

A.6 Corporate Networks

A.6.1 Do you have requirements for the connection of Corporate Networks (CN)? Yes/No

A.6.1.1 For the interface to your network, please specify.

A.6.1.2 For the delay from the CN terminal to the connection point in your network?

A.7 Terminals

End-to-end Quality of Service is highly influenced by the properties of the telephony terminals connected to and interworking with the network.

A.7.1 Which types of terminals are connected to your network?

A.7.1.1 Classical analogue telephones?

NOTE 1: Traditional telephone sets like the ones used with TDM networks.

A.7.1.2 Digital or ISDN telephones?

A.7.1.3 Cordless telephones including DECT?

A.7.1.4 Mobile telephones?

A.7.1.5 Standalone VoIP telephones?

A.7.1.6 Soft clients?

NOTE 2: E.g. telephony software running on a PC.

A.7.2 Can you provide technical details of the terminal implementations? Yes/No

A.7.2.1 By which mechanism is the terminal profile transferred across the UNI?

A.7.2.2 Which codecs are implemented?

A.7.2.2.1 G.711? Yes/No

A.7.2.2.2 other narrowband, please specify:

A.7.2.2.3 wideband, please specify:

A.7.2.2.4 dynamic codec shifting (auto bandwidth), please specify:

A.7.2.2.5 which of these codecs is mainly used?

A.7.2.3 Is packet loss concealment (PLC) implemented and used? Yes/No

7.2.3.1 standardized method, please specify:

A.7.2.3.2 proprietary, please specify the principle

A.7.2.4 Can you specify the settings of the codec? Yes/No

A.7.2.4.1 How many frames are per packet?

A.7.2.5 Is a de-jitter buffer implemented in the terminal? Yes/No

A.7.2.5.1 Has the de-jitter buffer

A.7.2.5.1.1 a fixed size?

A.7.2.5.1.2 an adaptive size?

A.7.2.5.1.2.1 Which algorithm is implemented?

- reactive algorithm
- histogram based algorithm
- algorithm that monitors network
- algorithm that optimizes user satisfaction

A.7.2.5.1.2.2 On which principle is this jitter buffer algorithm based?

- per talkspurt
- per packet
- modified version of per talkspurt, please specify
- modified version of per packet, please specify

A.7.2.6 To which standards do the electro acoustical parts of the terminals connected to your network comply?

A.7.2.6.1 ES 202 737, ES 202 738, ES 202 739 or ES 202 740?

A.7.2.6.2 ITU-T Recommendation P.1010?

A.7.2.6.3 other, please specify:

A.7.2.7 Are speech processing algorithms (e.g. NR, EC, AGC or VAD) implemented and used in the terminal? Yes/No

A.7.2.7.1 Which noise reduction (NR) algorithm is implemented and used, please specify.

A.7.2.7.2 Which echo cancellation (EC) algorithm is implemented and used, please specify.

A.7.2.7.3 Which automatic gain control (AGC) algorithm is implementedand used, please specify.

A.7.2.7.4 Which voice activity detection (VAD) algorithm is implemented and used, please specify.

A.7.2.8 Is forward error correction (FEC) implemented and used in the terminal? Yes/No

A.7.2.8.1 Which additional delays are incurred by the FEC?

A.7.3 Are you providing terminals to your customers (e.g. in a shop)? Yes/No

A.7.3.1 Do you impose an approval procedure to these types of terminals? Yes/No

A.7.3.1.1 Does such an approval include QoS requirements (e.g. electro-acoustical testing) Yes/No

A.7.3.2 Do you permit the connection of other terminals than the one provided by you to your network? Yes/No

A.7.3.3 Do you guarantee a specific quality to the customer when using your terminals? Yes/No

A.8 External Standards

A.8.1 Does the performance of your network comply with external standards? Yes/No

A.8.1.1 National performance standards, please specify.

A.8.1.2 International performance standards? Yes/No

A.8.1.2.1 ETSI, please specify

A.8.1.2.3 ITU-T, please specify

A.8.1.2.4 IEEE, please specify

A.8.1.2.5 Others, please specify.

A.9 Traffic characteristics

A.9.1 Which are the nominal values for the design of the network?

A.9.1.1 Activity in Busy Hour Call Attempts (BHCA):

A.9.1.2 Mean holding time:

A.9.1.3 other, please specify:

A.10 Services other than real-time voice communication

A.10.1 Which streaming principle do you use for IPTV, please specify:

A.10.2 Which streaming protocol do you employ for IPTV in your network?

A.10.2.1 IGMP2?

A.10.2.2 RTSP?

A.10.2.3 Network-based video recorder protocol?

A.10.3 Which codecs are mainly used in your IPTV implementation?

A.10.3.1 MPEG1, MPEG2 or MPEG4, please specify:

A.10.3.2 H.261, H.263 or H.264, please specify:

A.10.3.3 SMPTE VC-1?

A.10.4 Which Method is implemented in the case of Fax Service?

A.10.4.1 According to ITU-T Recommendation T.38?

A.10.4.1.1 How do you ensure compatibility between different T.38 devices?

A.10.4.2 According to ITU-T Recommendation G.711 transparent mode?

A.10.4.2.1 How do you ensure that in the G.711 channel speech processing devices (e.g. NR, VAD, VED) are rendered inoperative and that the delay is not variable?

A.10.4.3 According to ITU-T Recommendation V.150.1?

A.10.4.3.1 Which mode of V.150.1 is used?

A.11 Reference Models

In the diagrams of the reference models the following abbreviations are used in addition to those defined in clause 3.

AS	Application Server
ASP	Application Server Process
CSCF	Call Session Control Function
DCS	Dynamic Codec Shifting
DECT	Digital Enhanced Cordless Telecommunications
DSLAM	Digital Subscriber Line Access Module

DSS 1	Digital Subscriber Signalling System No. 1
H.26x	ITU-T Recommendations defining video codecs
I-CSCF	Interrogating Call Session Control Function
ISUP	IDDN User Part
IW	Interworking
LAP D	Link Access Procedure for D-channel
M3UA	Message transfer part 3 User Adaptation layer
MSAN	Multi Service Access Node
MTP	Message Transfer Part
nb	narrowband
NT	Network Termination (used in ISDN)
NVRP	Network-based Video Recorder Protocol
RACS	Resource and Admission Control Subsystem
S-CSCF	Serving Call Server Control Function
SCTP	Stream Control Transmission Protocol
SLA	Service Level Agreement
TEI	Termination Equipment Identifier
UDP	User Datagram Protocol
UPSF	User Profile Server Function
wb	wideband
WiFi	Wireless Fidelity (type of wireless access defined in IEEE 802.11)

A.11.1 Do you adhere to one of the following reference models - as shown in the following sub-clauses - or do you use a different model, please specify:

A.11.1.1 Reference Model IMS to POTS/ISDN Interworking

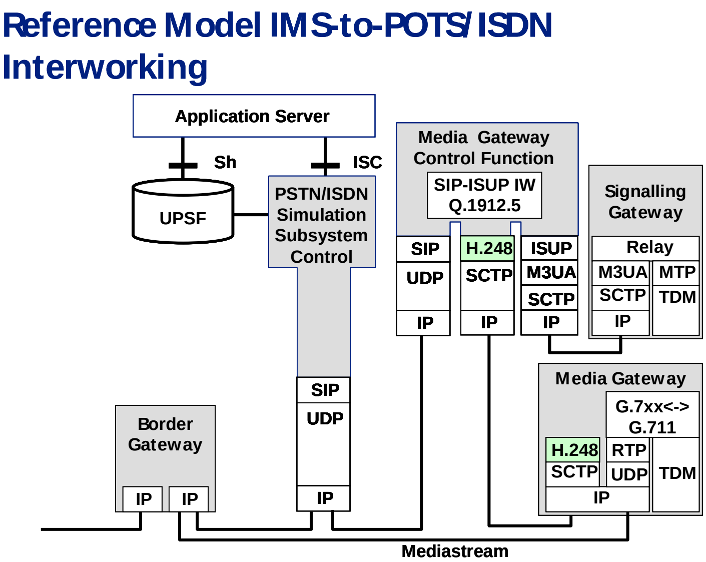


Figure A.1: Reference Model 1

A.11.1.2Reference Model IMS POTS/ISDN Simulation Subsystem

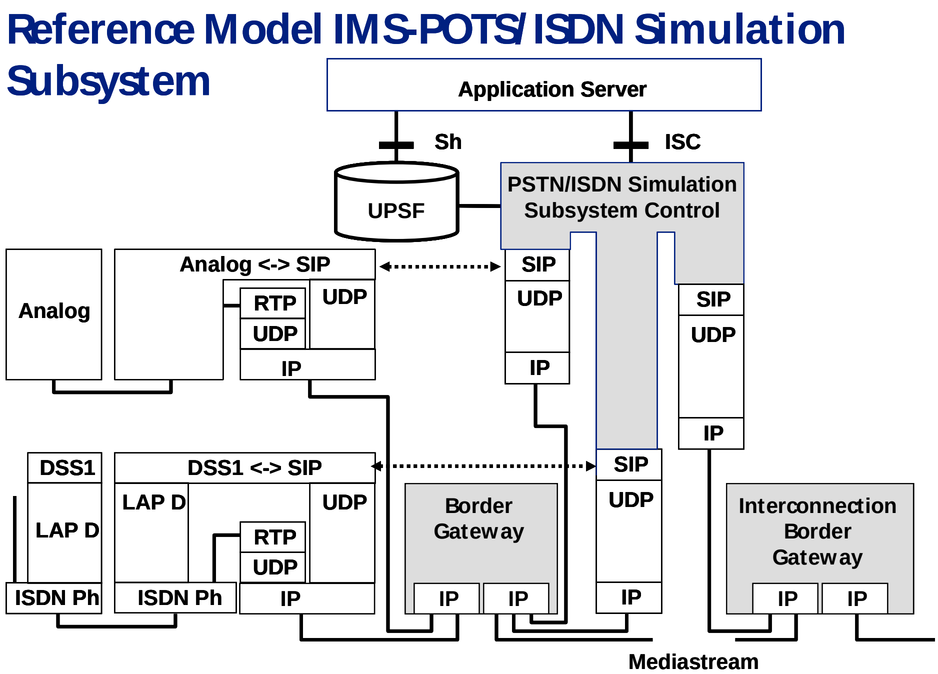


Figure A.2: Reference Model 2

A.11.1.3Reference Model "native IMS"

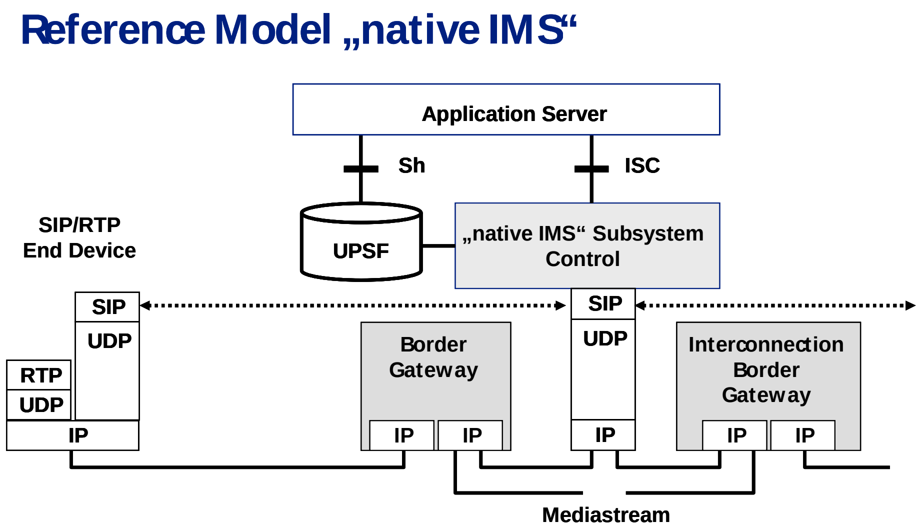


Figure A.3: Reference Model 3

A.11.1.4 Reference Model Decentralized AGCF

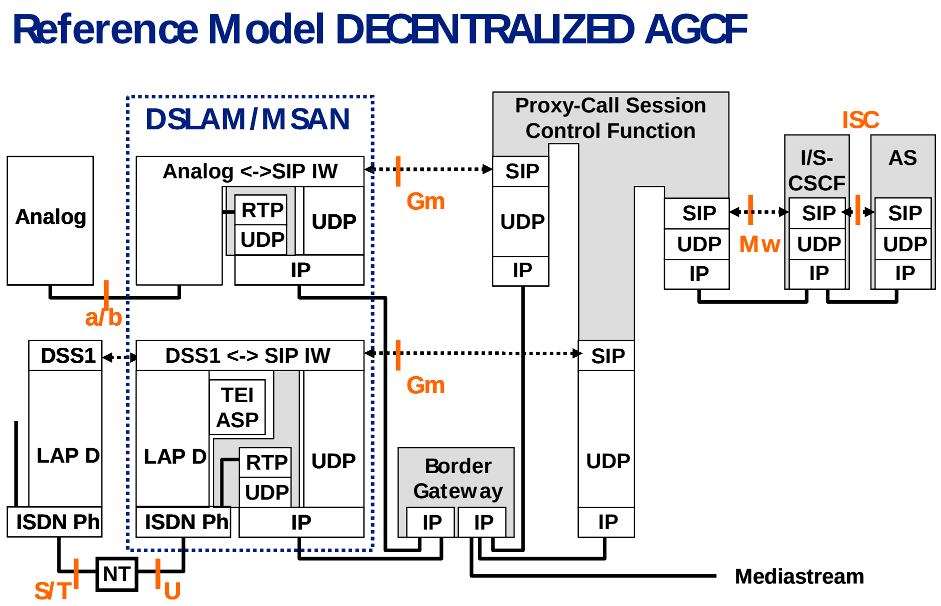


Figure A.4: Reference Model 4

Annex B:
Summary of the replies to the questionnaire

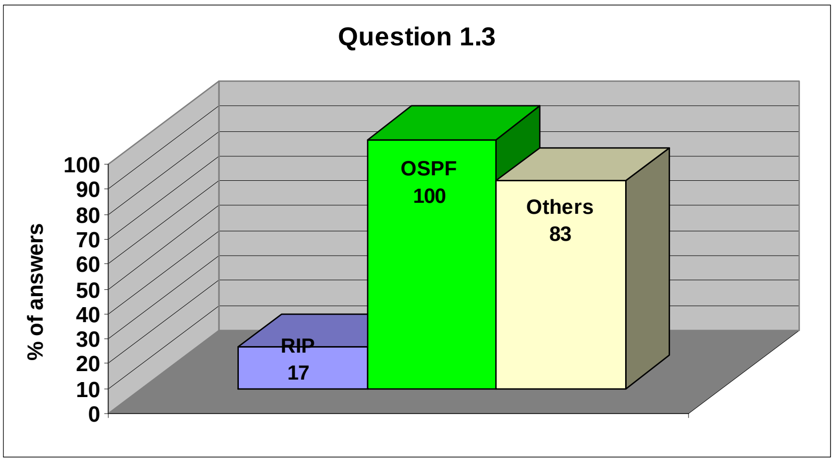
B.1 Core Network

B.1.3 Which routing protocols do you use?

B.1.3.1 RIP?

B.1.3.2 OSPF?

B.1.3.3 others, please specify.



Note: In case of "others" answers, BGP, MP-BGP,iBGP, eBGP and IS-IS have been mentioned.

B.1.4 Which technologies are used in the core?

B.1.4.1 WDM?

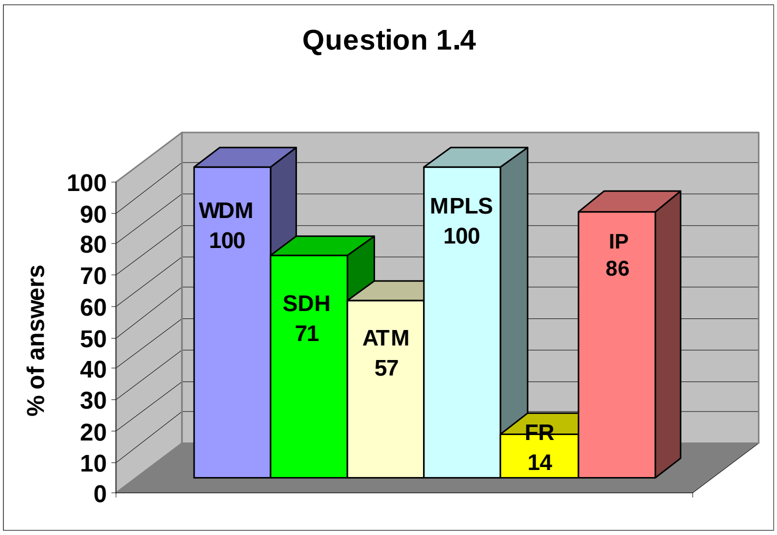
B.1.4.2 SDH?

B.1.4.3 ATM?

B.1.4.4 MPLS?

B.1.4.5 Frame Relay?

B.1.4.6 IP?



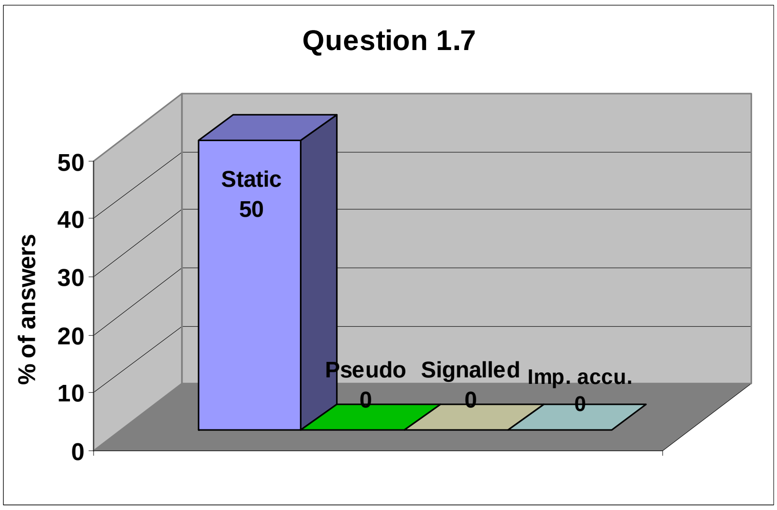
B.1.7 Have you implemented one of the methods described in ITU-T Rec. Y.1542 in order to ensure the compliance with ITU-T Recommendation Y.1541?

B.1.7.1 Static approach?

B.1.7.2 Pseudo-static approach?

B.1.7.3 Signalled approach?

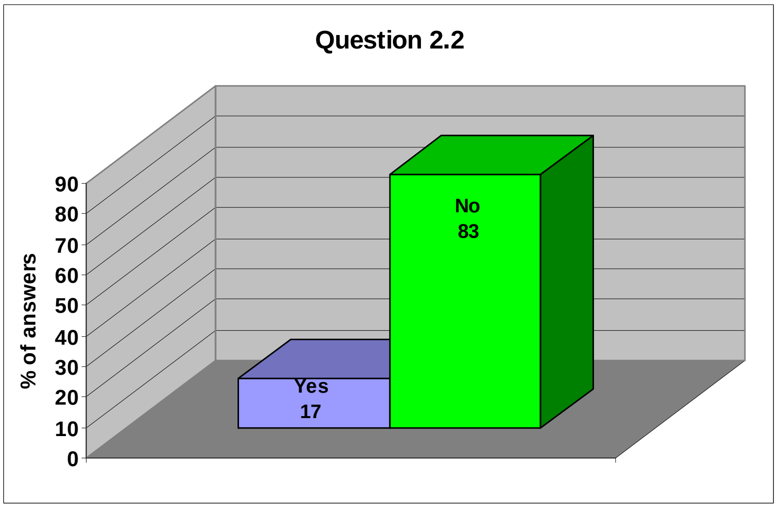
B.1.7.4 Impairment accumulation approach?



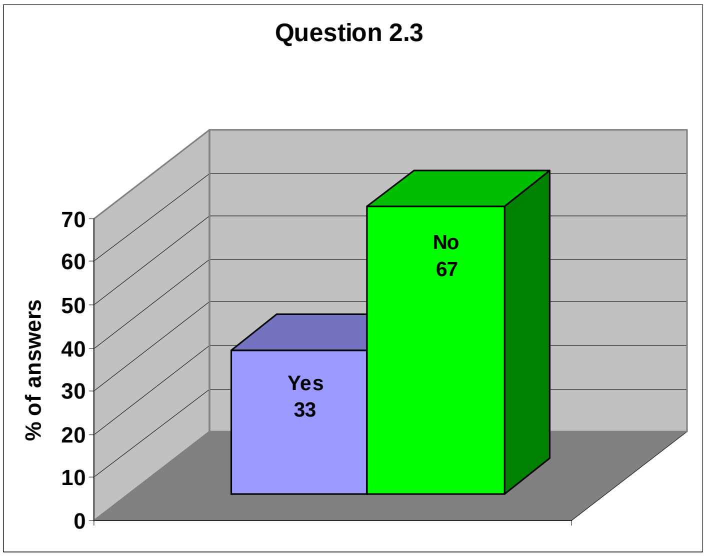
NOTE: In one answer case, we obtained answer, static approach because of government rules.

B.2 Network Interconnection

B.2.2 Have you deployed voice enhancement devices (VED) in your network? Yes/No



B.2.3 Have you deployed IP-to-IP gateways? Yes/No



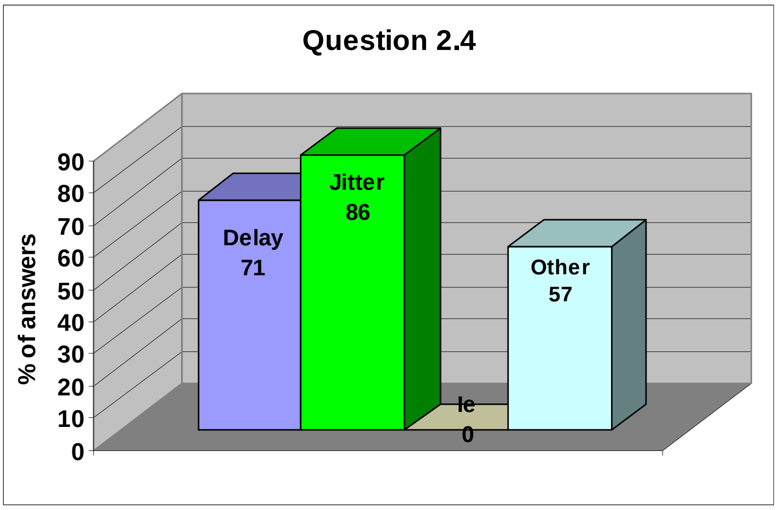
B.2.4 Which QoS parameter values can you (technically) make available at interconnection points?

B.2.4.1 Delay?

B.2.4.2 Delay variation?

B.2.4.3 Equipment impairment factor?

B.2.4.4 Other, please specify.



NOTE: In case of "other" answer: packet loss.

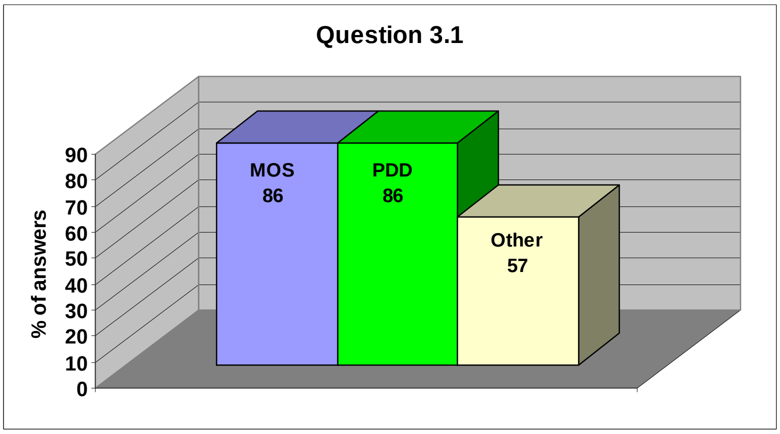
B.3 End-to-End Performance Objectives

B.3.1 Do you have any end-to-end performance objectives? Yes/No

B.3.1.1 In terms of Mean Opinion Scores (MOS)?

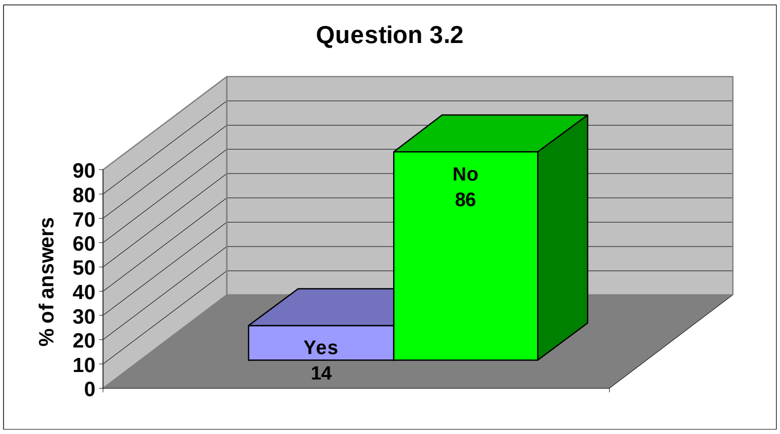
B.3.1.2 In terms of post dial delay (PDD)?

B.3.1.3 In terms of other parameters, please specify



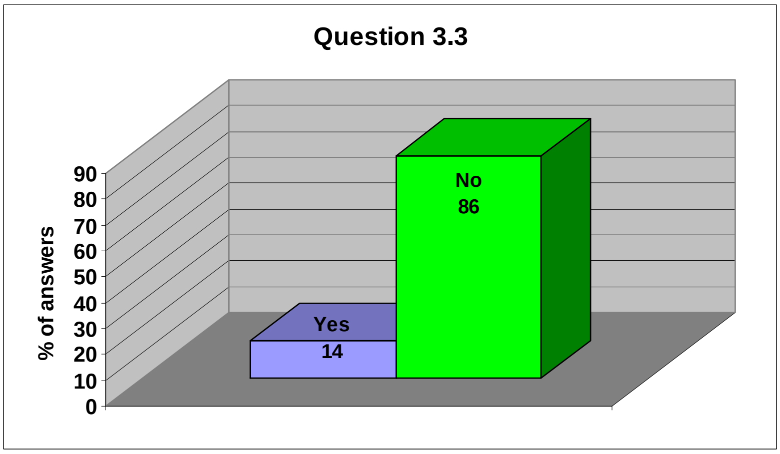
The possibility "other" covers these parameters: delay, call setup delay, call blocking ratio (government rule) and in case of mobile network there are TCH Blocked ratio, SCCH Success ratio, Drop call ratio and availability.

B.3.2 Does your PSTN emulation support ISDN clearmode service? Yes/No



It is mainly supported by IEEE 1588.

B.3.3 Does your PSTN emulation support certain dial-up modems (often used for alarm systems or telecare services). These modems require the use of a static dejitter buffer. Some terminals with these modems are built to expect only a few tens of milliseconds of end-to-end delay as seen in the PSTN. Greater delay than expected can result in false alarms when the terminal does not receive a response in time. Problems can also occur when accumulated jitter causes dejitter buffer overruns).? Yes/No



In case of "yes" answer, the technology was not specified.

B.3.4 Does your network architecture provide for unrestricted codec negotiation between user endpoints? Yes/Noi

The negotiation during call setup is mainly used.

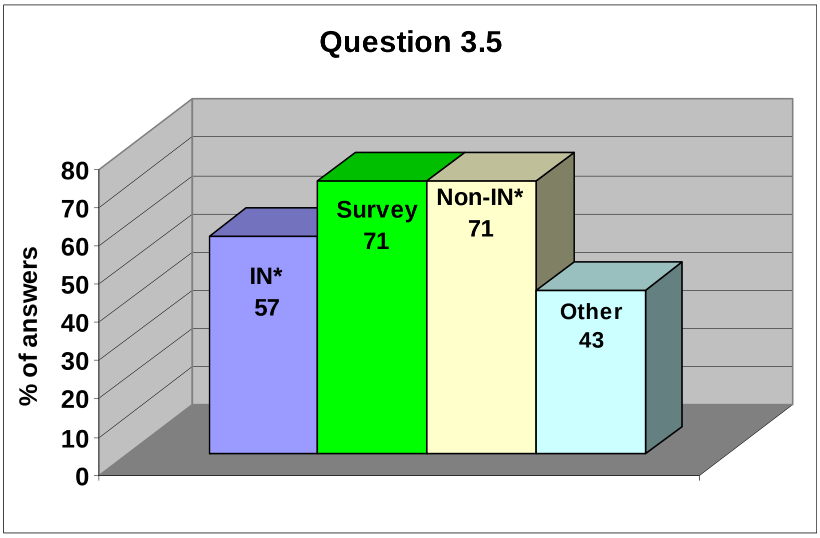
B.3.5 Which approach do you employ in the case of customer's satisfaction evaluation?

B.3.5.1 End-to-end real networks measurements by accomplished intrusive algorithms, like P.862, etc.?

B.3.5.2 Questionnaire or survey?

B.3.5.3 Non-intrusive approach at some measurements points, like P.563?

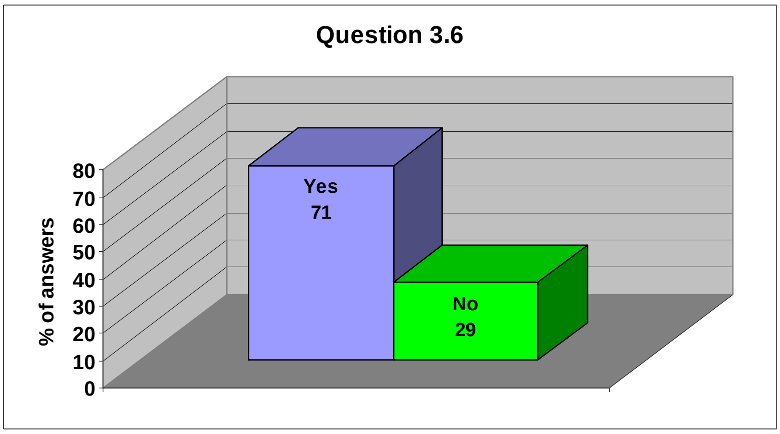
B.3.5.4 Other approach, please specify:



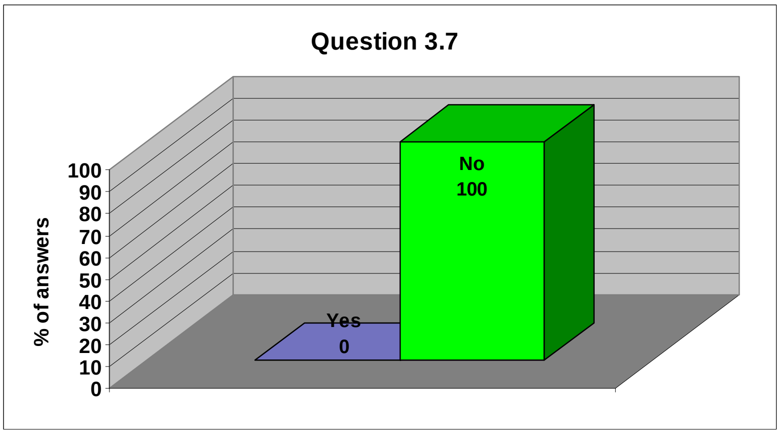
NOTE: *IN-Intrusive

Others: Network parameters measurements, solutions based on counters provided by the network equipment, Call details records, Call statistics, monitoring approaches based on RTCP

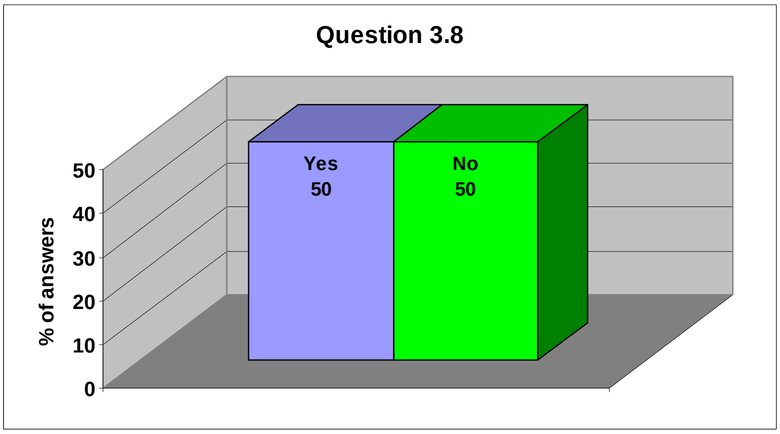
B.3.6 Do you support end-to-end codec type/mode negotiation? Yes/No



B.3.7 Do you support Tandem Free Operation? Yes/No

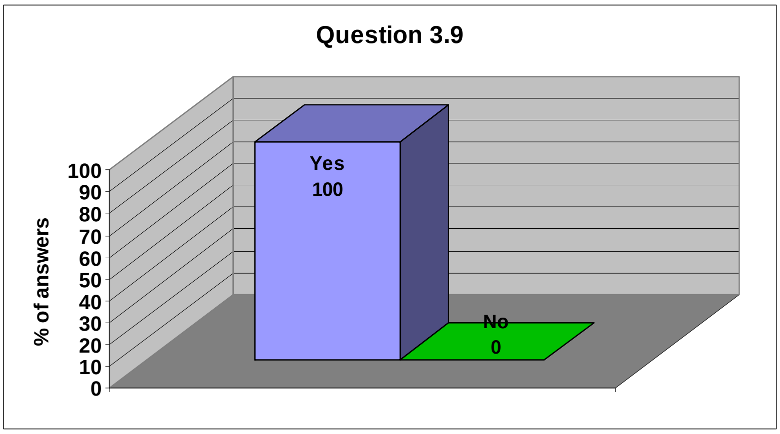


B.3.8 Are you using call admission control as means to guarantee QoS? Yes/No



This approaches is used in Access networks.

B.3.9 Are you using Diffserv? Yes/No



B.3.9.1 In which network segment, please specify: access and core networks

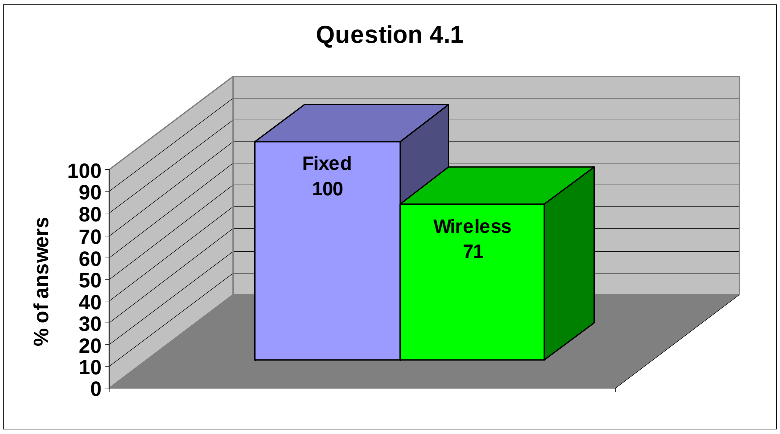
B.3.9.2 How many classes are actually used, please specify: mainly 4 classes, namely voice (Expedited Forwarding), video, data and best effort. background traffic.

B.4 Access Networks

B.4.1 Which access technologies do you use, for NGN?

B.4.1.1 Fixed, please specify: xDSL (ADSL, SDSL, VDSL2, ADSL2), Ethernet, FTTH, HFC (EuroDOCSIS 2.0)

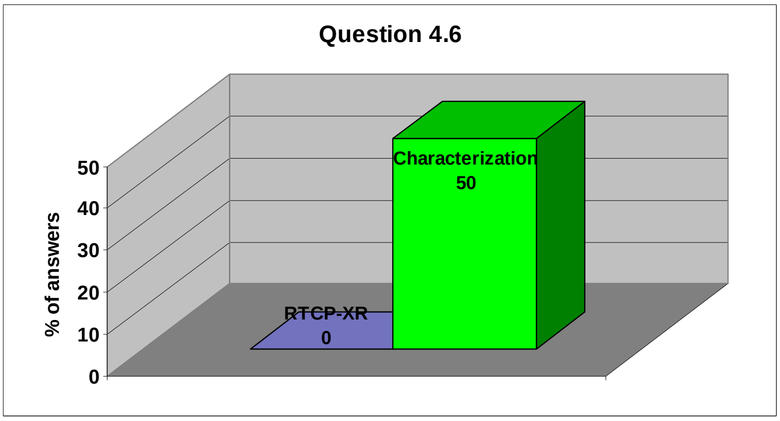
B.4.1.2 Wireless, please specify: CDMA, WiMAX, UTRAN 3G, HSDPA, HSUPA, IEEE 802.11 [i.66], GSM



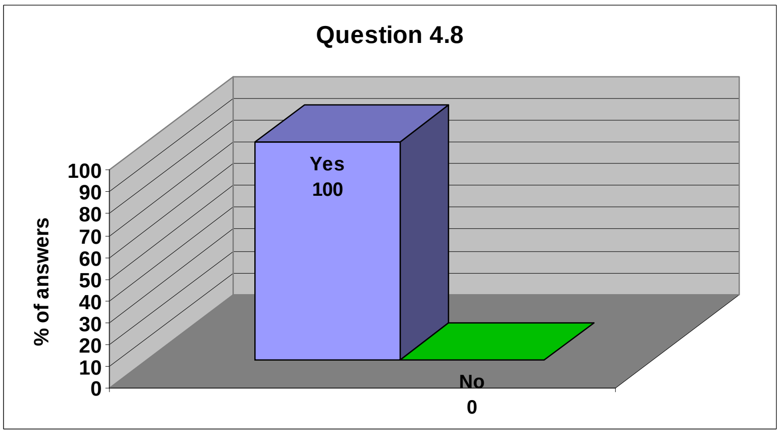
B.4.6 Do you use any optimization approach between the network and the terminals? Yes/No

B.4.6.1 RTCP-XR?

B.4.6.2 Characterization of VoIP edge devices and gateways over a wide range of network conditions?



B.4.8 Do use any prioritization in access network? Yes/No



B.4.8.1 Please specify categories for each kind of technology: classical approach based on diff serv, DSCP values

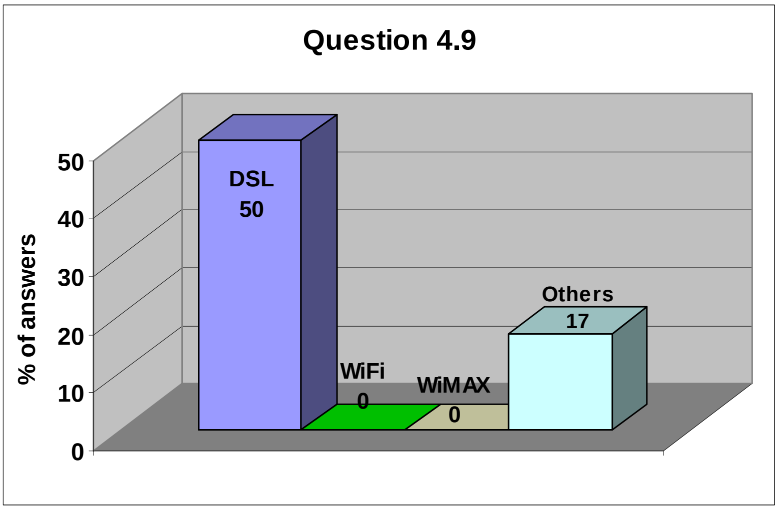
B.4.8.2 Please specify the mapping of Class of Service values to the MAC layer parameters or queues:

B.4.9 Do you use any advanced settings of access technology?

B.4.9.1 DSL: Fast or interleaving channel or other techniques according to QoS provisioning?

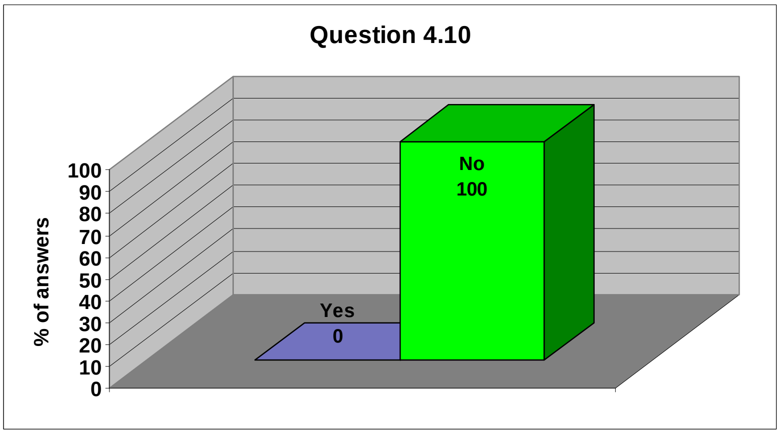
B.4.9.2 WiFi: Type of access method on MAC layer or other techniques according to QoS provisioning, please specify.

B.4.9.3 WIMAX: Scheduling algorithm or other techniques according to QoS provisioning, please specify.



Others: EuroDOCSIS 2.0 in case of cable networks

B.4.10 Do you employ cRTP in your network? Yes/No



B.5 Gateways (edge devices)

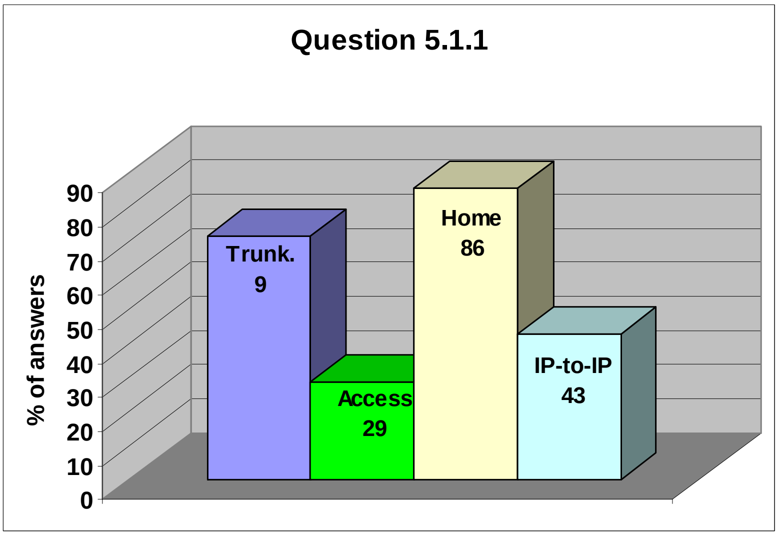
B.5.1.1 Which types of gateways do you provide? Please specify per type the properties according to subsequent sections:

B.5.1.1.1 Trunking Media Gateways to adjacent ISDN?

B.5.1.1.2 Access Media Gateways for support of non-SIP/RTP UNIs?

B.5.1.1.3 Home Media Gateways installed at customers' premises?

B.5.1.1.4 IP-to-IP Gateways?



B.5.1.2 Which codecs are implemented?

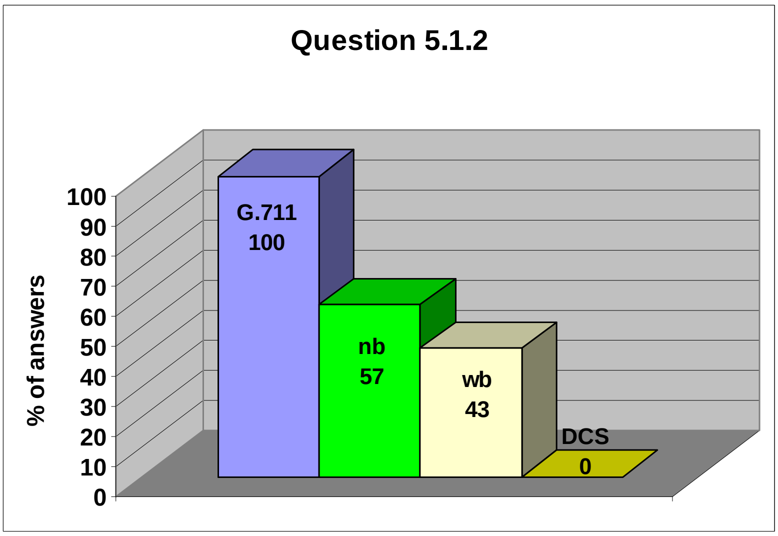
B.5.1.2.1 G.711? Yes/No

B.5.1.2.2 other narrowband, please specify: G.729, G.723, G.726

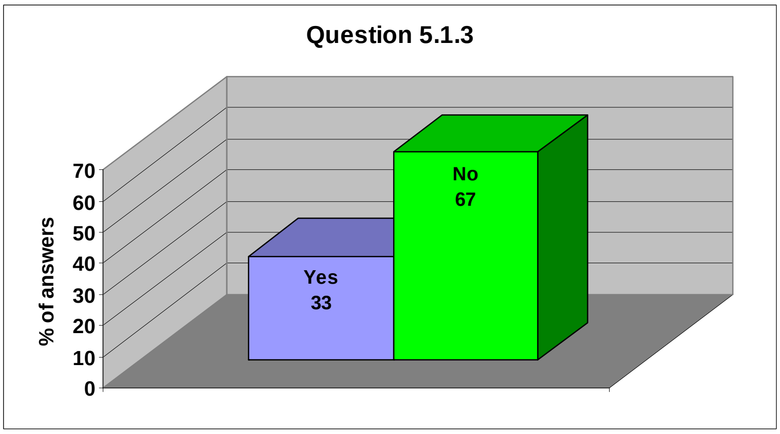
B.5.1.2.3 wideband, please specify: G.722

B.5.1.2.4 dynamic codec shifting (auto bandwidth), please specify:

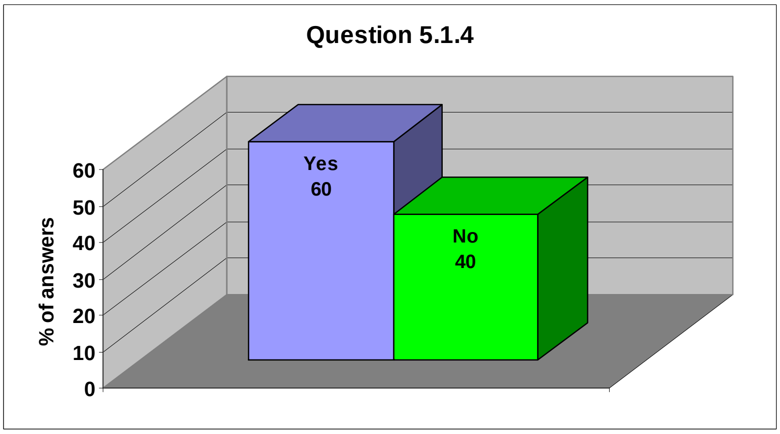
B.5.1.2.5 which of these codecs is mainly used? G.729, G.711



B.5.1.3 Do you adopt in your gateway a single codec approach (i.e. support only one special codec) or a multiple codec approach?



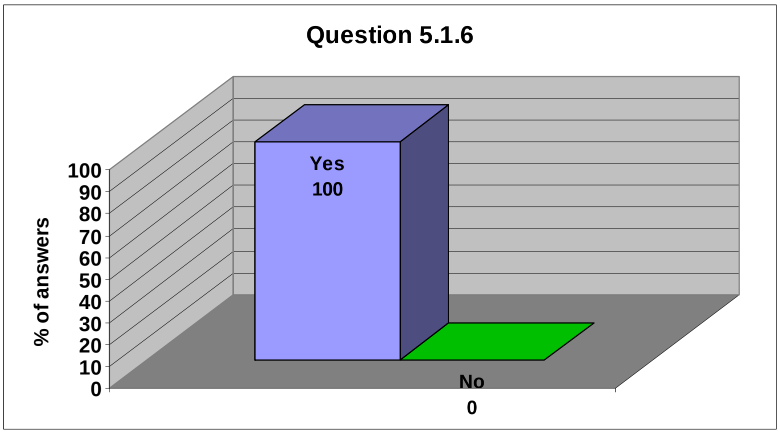
B.5.1.4 Is packet loss concealment (PLC) implemented and used? Yes/No



B.5.1.4.1 standardized method, please specify: No detailed information

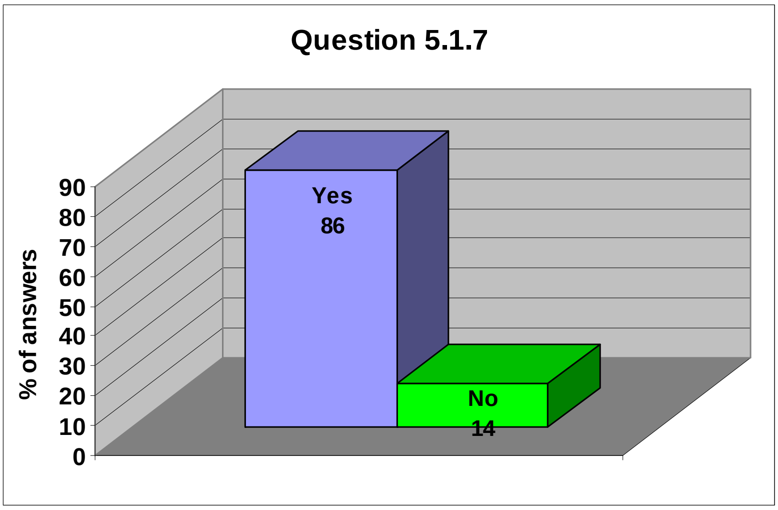
B.5.1.4.2 proprietary, please specify the principle

B.5.1.6 Is a de-jitter buffer implemented in the gateway? Yes/No



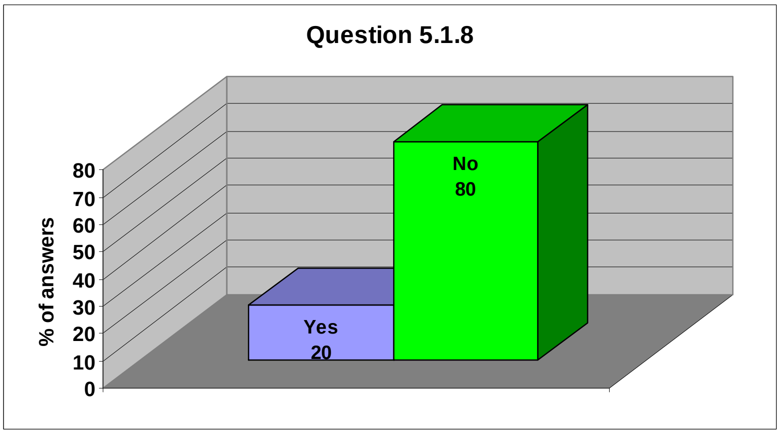
The operators are using fixes as well as adaptive sizes, mainly additional parameters have not been specified but the algorithm that monitors network based on per packet solution has been pointed out in one case.

B.5.1.7 Are speech processing algorithms (e.g. NR, EC, AGC or VAD) implemented and used in the gateway? Yes/No



EC is only implemented in that case.

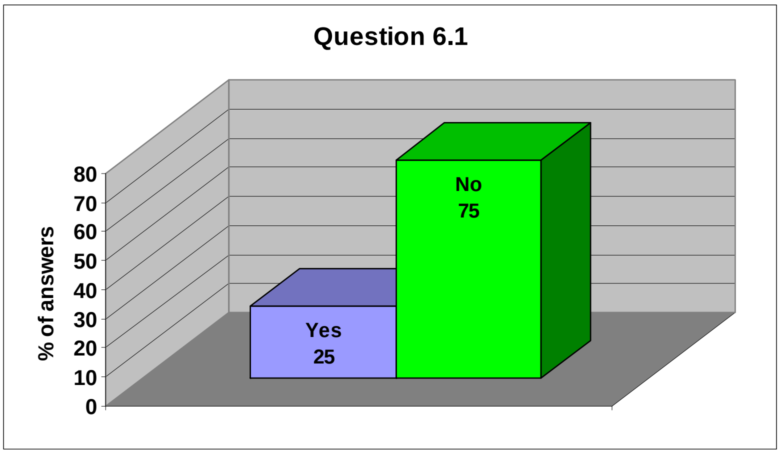
B.5.1.8 Is forward error correction (FEC) implemented and used in the gateway? Yes/No



No info about additional delays incurred by the FEC

B.6 Corporate Networks

B.6.1 Do you have requirements for the connection of Corporate Networks (CN)? Yes/No



No detailed information about CN has been provided by operators.

B.7 Terminals

End-to-end Quality of Service is highly influenced by the properties of the telephony terminals connected to and interworking with the network.

B.7.1 Which types of terminals are connected to your network?

B.7.1.1 Classical analogue telephones?

NOTE 1: Traditional telephone sets like the ones used with TDM networks.

B.7.1.2 Digital or ISDN telephones?

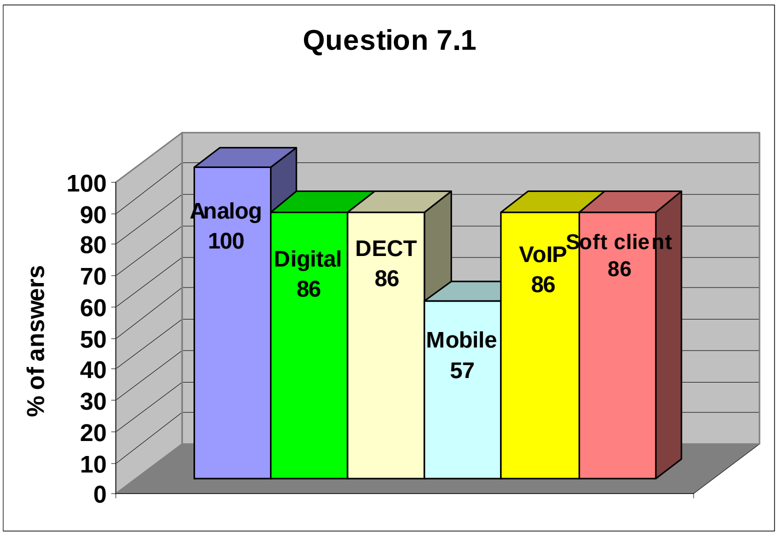
B.7.1.3 Cordless telephones including DECT?

B.7.1.4 Mobile telephones?

B.7.1.5 Standalone VoIP telephones?

B.7.1.6 Soft clients?

NOTE 2: E.g. telephony software running on a PC.



B.7.2.2 Which codecs are implemented?

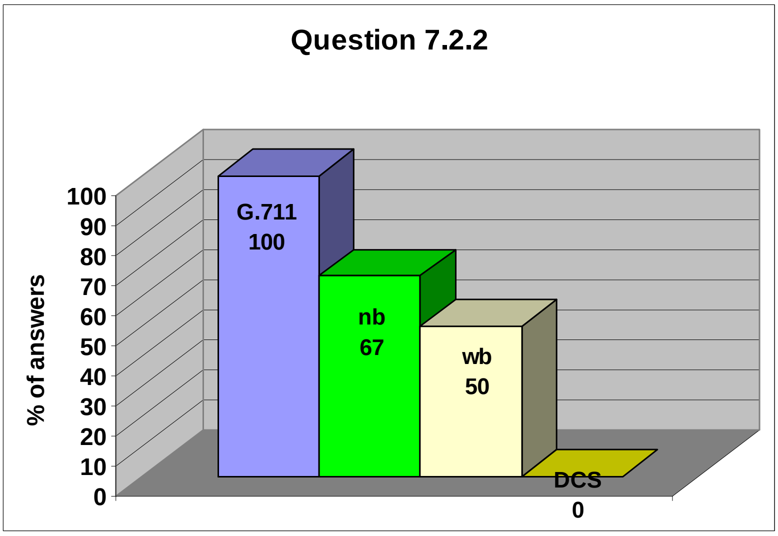
B.7.2.2.1 G.711? Yes/No

B.7.2.2.2 other narrowband, please specify: G.729, G.726, G.723

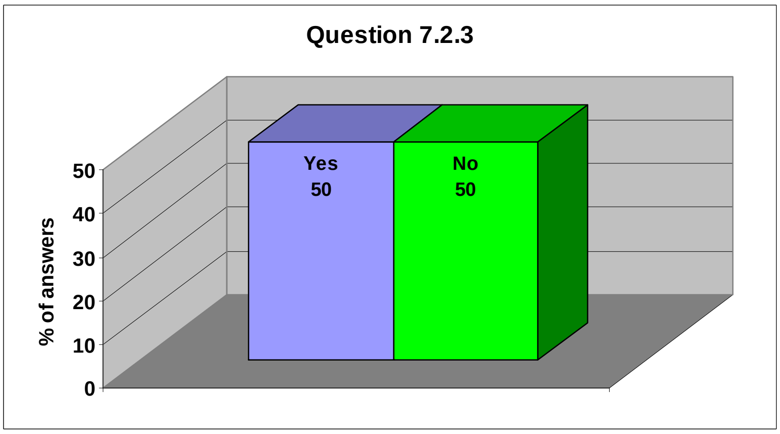
B.7.2.2.3 wideband, please specify: G.722

B.7.2.2.4 dynamic codec shifting (auto bandwidth), please specify:

B.7.2.2.5 which of these codecs is mainly used? G.711, G.729

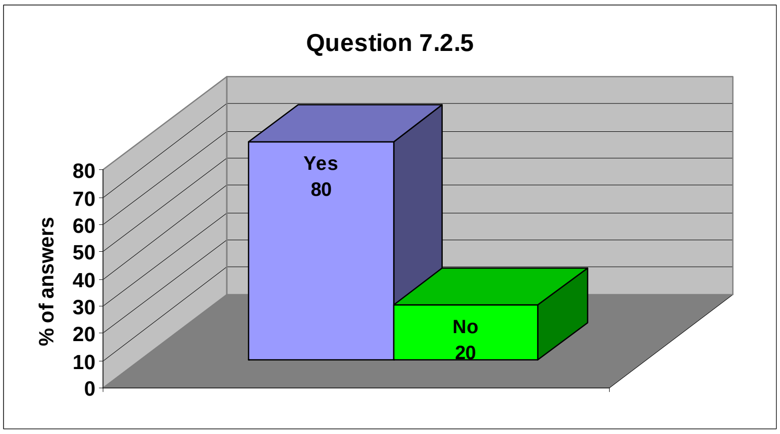


B.7.2.3 Is packet loss concealment (PLC) implemented and used? Yes/No



Without detailed description of the method used.

B.7.2.5 Is a de-jitter buffer implemented in the terminal? Yes/No



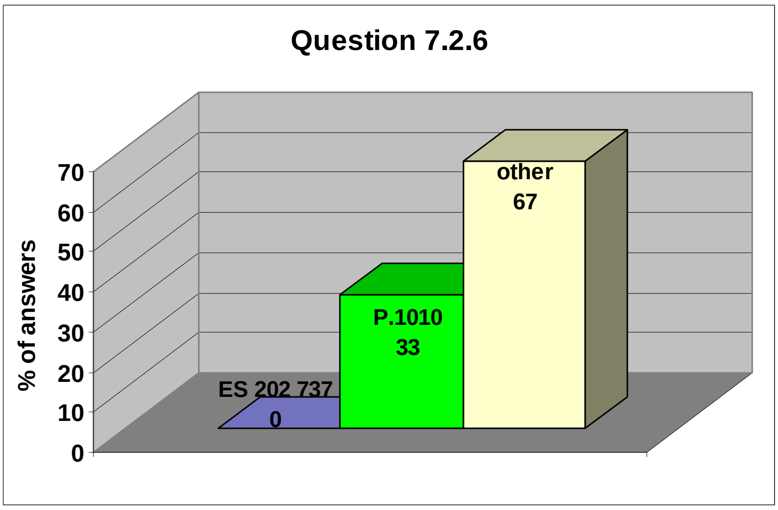
The operators are using adaptive size, mainly additional parameters have not been specified but the algorithm that monitors network based on per packet solution has been pointed out in one case.

B.7.2.6 To which standards do the electro acoustical parts of the terminals connected to your network comply?

B.7.2.6.1 ES 202 737 [i.5], ES 202 738 [i.6], ES 202 739 [i.7] or ES 202 740 [i.8]?

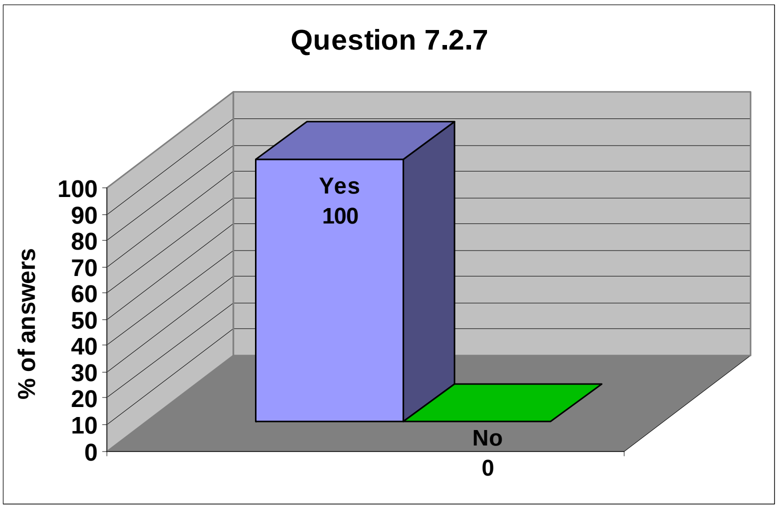
B.7.2.6.2 ITU-T Recommendation P.1010 [i.63]?

B.7.2.6.3 other, please specify:



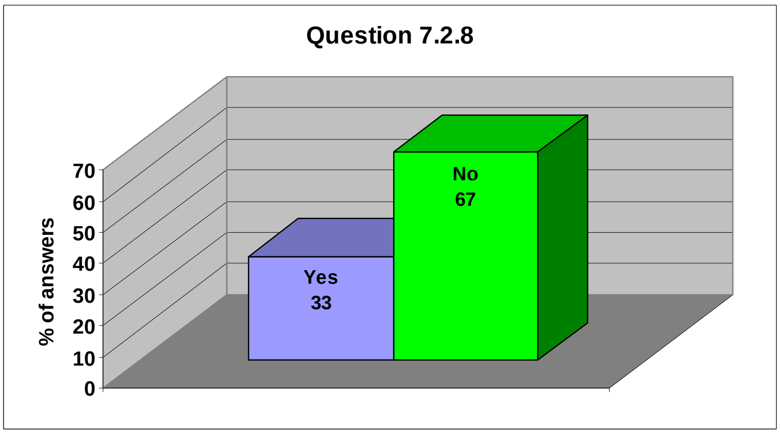
In case of "other" answer, the possibilities have not been described.

B.7.2.7 Are speech processing algorithms (e.g. NR, EC, AGC or VAD) implemented and used in the terminal? Yes/No



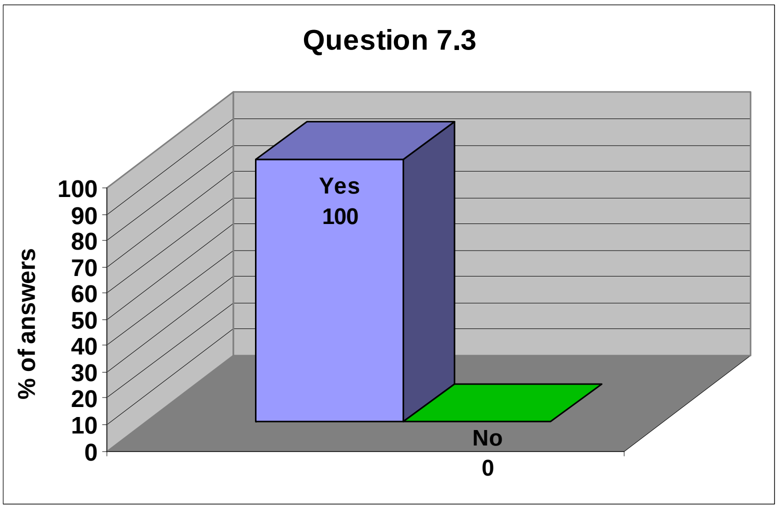
EC is mainly deployed.

B.7.2.8 Is forward error correction (FEC) implemented and used in the terminal? Yes/No



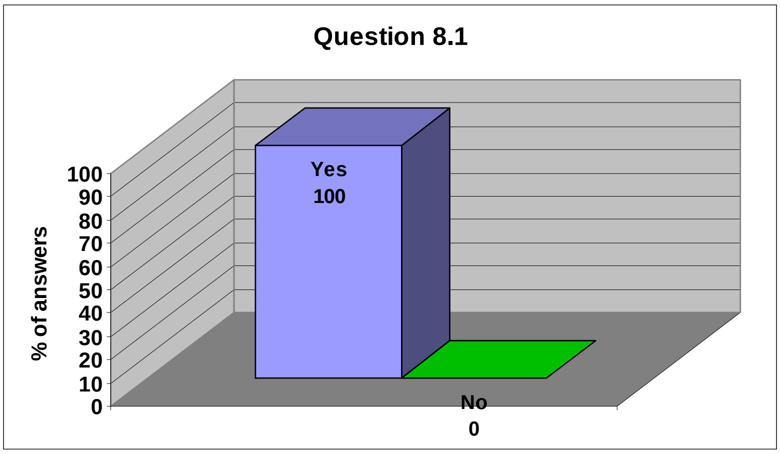
No detailed info about delays in case of this implementation.

B.7.3 Are you providing terminals to your customers (e.g. in a shop)? Yes/No



B.8 External Standards

B.8.1 Does the performance of your network comply with external standards? Yes/No



B.8.1.1 National performance standards, please specify. SR 784.101.113/1.2, TTC Stanards, Japanese Government rules, TKG 1TR110

B.8.1.2 International performance standards? Yes/No

B.8.1.2.1 ETSI, please specify: TISPAN ES 282 003, TR 102 775, TS 101 909-18, ES 201 970, TBR 021, EG 201 120, ES 201 235-3

B.8.1.2.3 ITU-T, please specify: Y.1541, Y.1542, Q.543, G.114, G.121, H.248.1 (megaco), Y.1221(Token policer), Q.552, G.109, G.107, P.862, G.131, G.168

B.8.1.2.4 IEEE, please specify

Others, please specify. RFC4566(SDP), Cable networks standards like DOCSIS

B.9 Void

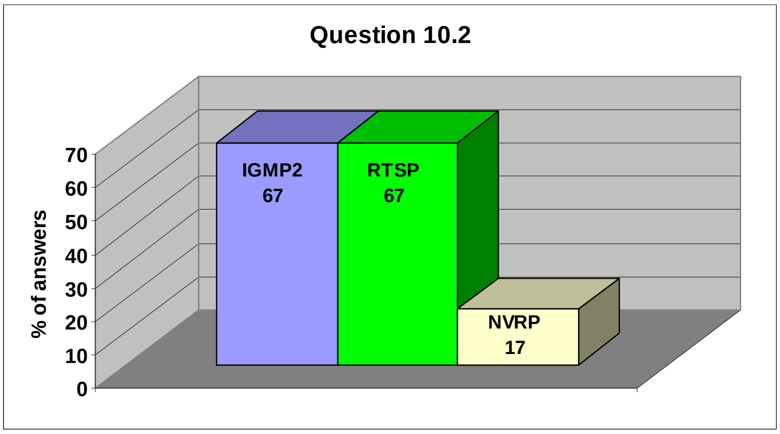
B.10 Services other than real-time voice communication

B.10.2 Which streaming protocol do you employ for IPTV in your network?

B.10.2.1IGMP2?

B.10.2.2RTSP?

B.10.2.3Network-based video recorder protocol?

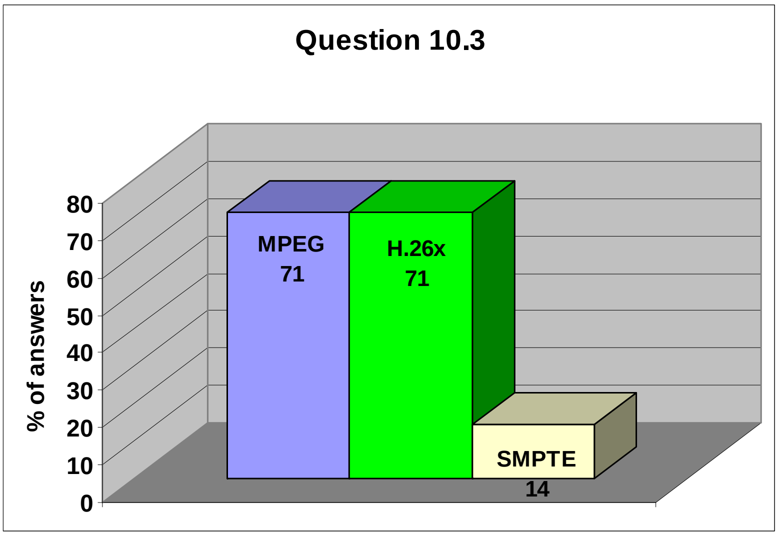


B.10.3 Which codecs are mainly used in your IPTV implementation?

B.10.3.1 MPEG1, MPEG2 or MPEG4, please specify: MPEG 2 and 4, MPEG1 layer II for audio purposes

B.10.3.2 H.261, H.263 or H.264, please specify: H.264, H264 AVC over MPEG2, H.263

B.10.3.3 SMPTE VC-1?



B.10.4 Which Method is implemented in the case of Fax Service?

B.10.4.1 According to ITU-T Recommendation T.38?

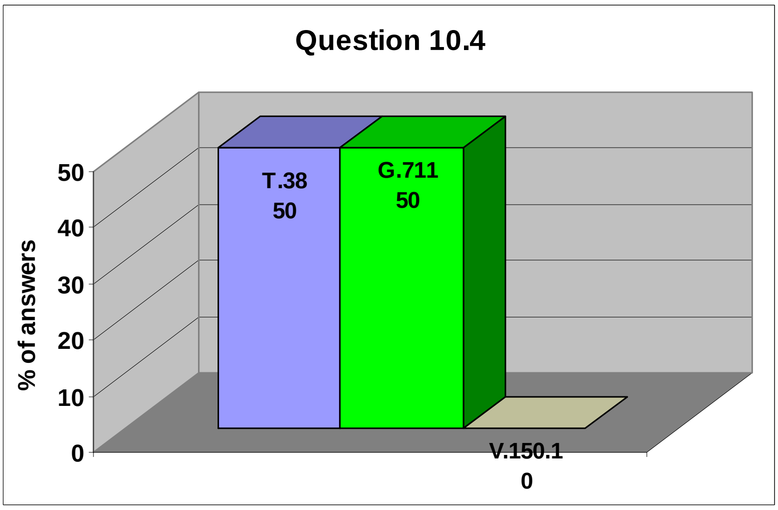
B.10.4.1.1 How do you ensure compatibility between different T.38 devices? Up to now not ensured

B.10.4.2 According to ITU-T Recommendation G.711 transparent mode?

B.10.4.2.1 How do you ensure that in the G.711 channel speech processing devices (e.g. NR, VAD, VED) are rendered inoperative and that the delay is not variable? By measurements

B.10.4.3 According to ITU-T Recommendation V.150.1?

B.10.4.3.1 Which mode of V.150.1 is used?



B.11 Reference Models

B.11.1 Do you adhere to one of the following reference models - as shown in the following sub-clauses - or do you use a different model, please specify:

B.11.1.1 Reference Model IMS to POTS/ISDN Interworking

In case of one answer, these differences have been pointed out:

Partially applied:

- Media gateway control is via MGCP (RFC 3445) instead of H.248/Megaco
- There is no (external) application server applied, but ISC is supported
- For the signalling gateway, there are two different implementations:
    - The signalling gateway is integrated with the PSTN media gateway (software option),
    - The signalling gateway is a separate entity (as pictured).

Reference Model IMS-to-POTS/ISDN Interworking: see Figure A.1 in clause A.11.

B.11.1.2 Reference Model IMS POTS/ISDN Simulation Subsystem

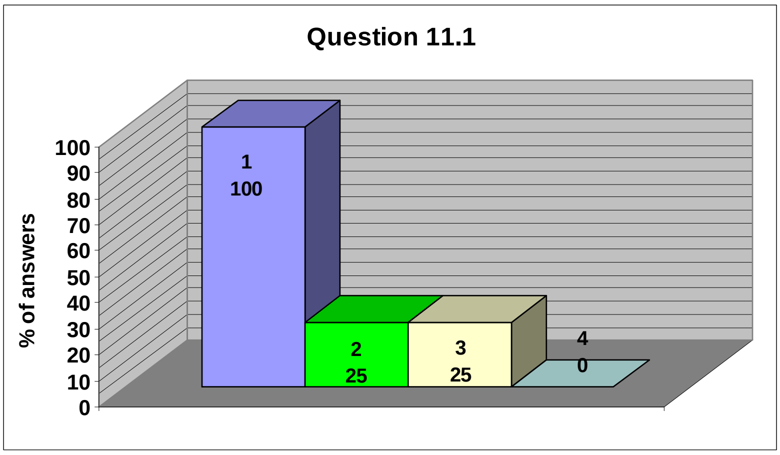
Reference Model IMS-POTS/ISDN Simulation Subsystem: see Figure A.2 in clause A.11.

B.11.1.3 Reference Model "native IMS"

Reference Model "native IMS": see Figure A.3 in clause A.11.

B.11.1.4 Reference Model Decentralized AGCF

Reference Model "decentralized AGCF": see Figure A.4 in clause A.11.



AGC	Automatic Gain Control
AGCF	Access Gateway Control Function
ATM	Asynchronous Transfer Mode
BHCA	Busy Hour Call Attempts
CN	Corporate Network
cRTP	compressed Realtime Transport Protocol
DECT	Digital Enhanced Cordless Telecommunication
DSL	Digital Subscriber Line
EC	Echo Canceller
FEC	Forward Error Correction
IEEE	Institute of Electrical and Electronics Engineers
IGMP2	Internet Group Multicast Protocol, version 2
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IPTV	Internet Protocol Television
ISDN	Integrated Services Digital Network
ITU	International Telecommunication Union
ITU-T	ITU Telecommunication Standardization Sector
MAC	Medium Access Control
MOS	Mean Opinion Score
MPEG	Moving Picture Experts Group
MPLS	Multi-Protocol Label Switching
NR	Noise Reduction
OSPF	Open Shortest Path First
PDD	Post Dial Delay
PLC	Packet Loss Concealment
POTS	Plain Old Telephone System
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTSP	Real-Time Streaming Protocol
RIP	Routing Information Protocol
RTCP-XR	Real-Time Transport Control Protocol Extended Reports
RTP	Realtime Transport Protocol
SDH	Synchronous Digital Hierarchy
SIP	Session Initiation Protocol
SMPTE VC-1	Society of Motion Picture and Television Engineers Video Codec, version 1
TM	Trade Mark
UNI	User-Network Interface
VAD	Voice Activity Detection
VED	Voice Enhancement Device
VoIP	Voice over Internet Protocol
WDM	Wavelength Division Multiplex
WIMAX	Worldwide Interoperability for Microwave Access

Annex C: Critical issues: Number Portability and call routing

Number Portability in a PSTN/ISDN segment and NGN are not completely compatible. The procedures of EN 302 097 [i.2] are compatible with EN 383 001 [i.1] except for:

1) The procedures in Annex A of EN 302 097 [i.2] "Procedures for the Concatenated Addressing method" are compatible with the specifications for NGN if the Request-URI remains unchanged or the modifications retain the original ported directory number. No such specifications are known to exist.

2) The procedures in Annex B of EN 302 097 [i.2] are not compatible with the specifications for NGN. The ported Directory Number remains in the Called Party Number parameter and is mapped into the To and Request-URI fields. This Annex should be removed or additional procedures should be defined in EN 383 001 [i.1].

If the donor network is a legacy PSTN/ISDN network and deploys onward routing, undesirable multiple transcoding may happen as illustrated in Figure C.1. Although when selecting a clearchannel code for RTP/IP the data loss and delay variations might impact QoS less.

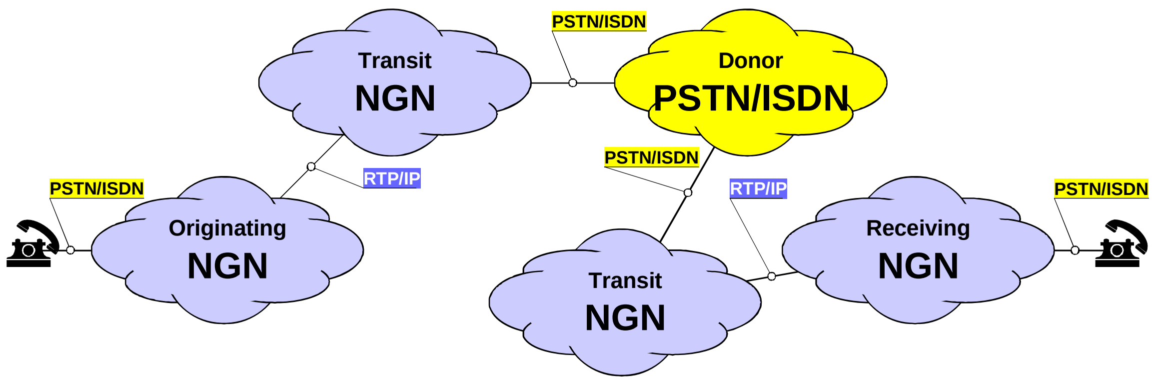


Figure C.1: Undesirable multiple transcoding with onward routing by donor

Under unfavourable conditions, up to four IP islands are in the bearer path. This is illustrated in Figure C.2.

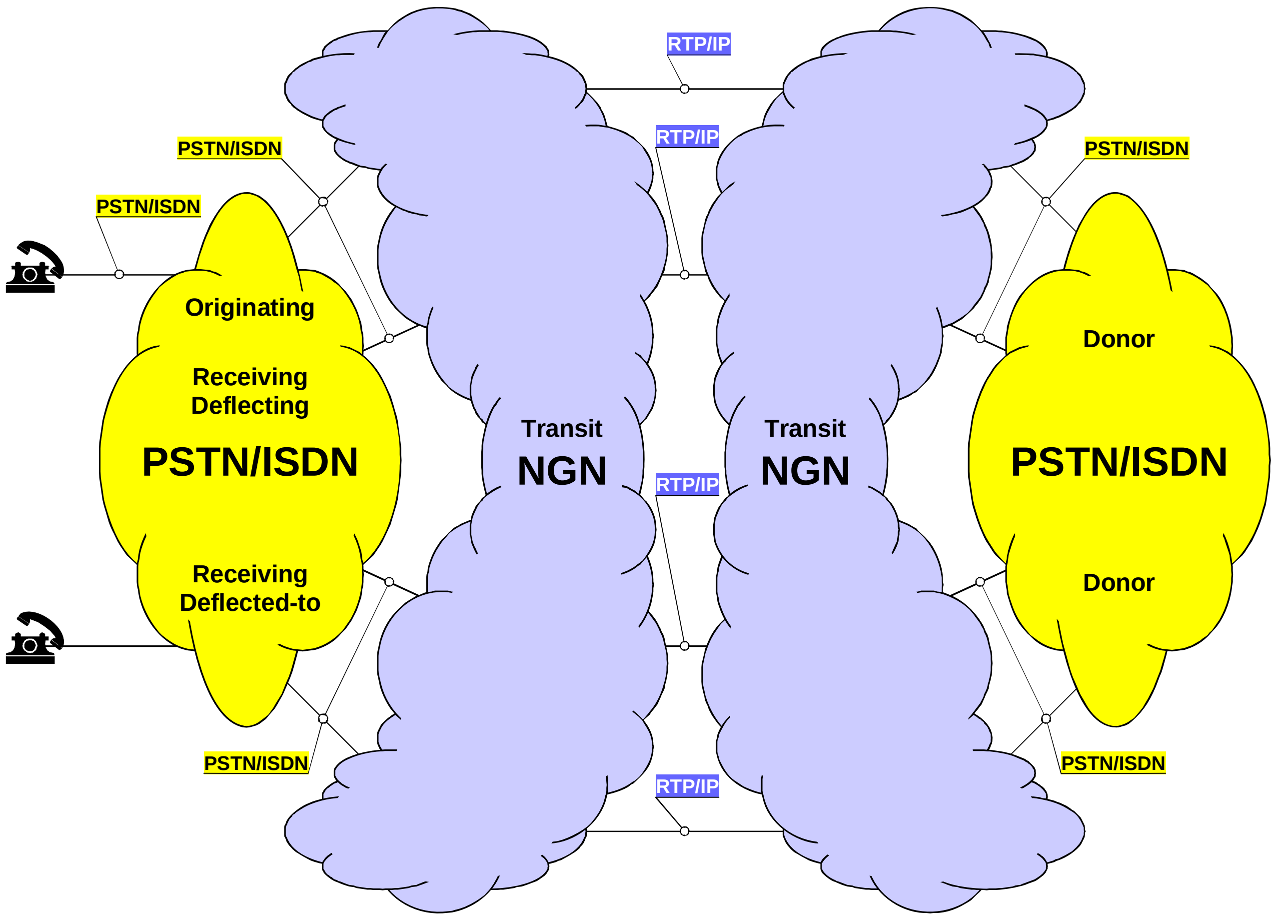


Figure C.2: Undesirable multiple transcoding with onward routing by donors and deflection

As onward routing may lead to undesirable multiple transcoding situations, it has been recommended in TR 183 014 [i.3] that the QoR (query on release) method according to Annex C of EN 302 097 [i.2] be deployed in the NGN PSDN/ISDN Emulation Service.

NOTE: This clause is an excerpt from TR 183 014 [i.3].

Annex D: Bibliography

ETSI EN 300 347-1 (1999): "V interfaces at the digital Local Exchange (LE); V5.2 interface for the support of Access Network (AN); Part 1: V5.2 interface specification".

ETSI TR 102 430: "Speech Processing, Transmission and Quality Aspects (STQ); Basic Issues concerning the Quality of Speech over Packet Technology (both Internet and Next Generation Networks)".

ITU-T Recommendation P.800.1 (2006-07): "Methods for objective and subjective assessment of quality".

History

Document history		
V1.1.1	October 2009	Publication