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NICC Document

End-to-End Network Performance Rules & Objectives for the Interconnection of NGNs

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Foreword

This NICC Document (ND) has been produced by the NICC All-IP Task Group.

Introduction

Previous versions of this NICC Document (ND) were produced by the NICC E2E QoS WG (End-to-End Quality of Service Working Group). Version 3 is a significant update by the NICC All IP TG (All Internet Protocol Task Group) addressing the need to define separate performance requirements for Voice, Voice-band data (VBD) and Clearmode services in NGNs (Next Generation Networks).

This document describes rules and guidelines governing the end-to-end network performance of interconnected fixed and mobile Next Generation Networks providing Voice, VBD and/or Clearmode services.

The network performance planning rules and guidelines described in this document are intended to ensure that end-users experience a high Quality of Service (QoS) over interconnected NGNs. Careful consideration needs to be given to the potential effect on user-perceived QoS (e.g. delay, packet loss, transcoding etc.) and functional aspects (e.g. call failures, outages, call setup time etc.) when planning the deployment of IP based technologies in Next Generation Networks. The rules and guidelines given in this document shall be taken into account in order to minimise the impact on QoS from the introduction of NGNs and any other new technologies.

1 Scope

The present document describes rules and guidelines governing the end-to-end network performance of interconnected fixed and mobile Next Generation Networks providing Voice, VBD and/or Clearmode services.

Future versions of this document may be expanded to include other services (such as multimedia services) and revised to take into account evolution of access and core network technologies.

This document only addresses aspects of the performance of public NGNs, both fixed and mobile. It does not address the network performance or QoS implications of enterprise networks or Best Efforts derived voice services.

2 References

2.1 Normative references

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

[1] ETSI drafting rules Verbal Forms For The Expression Of Provisions (2015-06)

[2] NICC ND1701 Recommended Standard for the UK National Transmission Plan for Public Networks

[3] NICC ND1431 Guidance on CPE Compatibility on NGNs and NGAs

[4] ITU-T Rec. G.811 Timing characteristics of primary reference clocks

[5] ITU-T Rec. G.711 Pulse code modulation (PCM) of voice frequencies

[6] IETF RFC 4040 RTP Payload Format for a 64 kbit/s Transparent Call

[7] ITU-T Rec. G.168 Digital network echo cancellers

[8] ITU-T Rec. Q.115.1 Logic for the control of echo control devices and functions

[9] IETF RFC 4733 RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals

[10] NICC ND1444 DTMF Best Practice Guide (BPG)

2.2 Informative references

[i1] NOT USED

[i2] ITU-T Rec. G.108.2 Transmission planning aspects of echo cancellers

[i3] Broadband Forum TR067 ADSL Interoperability Test Plan

[i4] Broadband Forum TR100 ADSL2/ADSL2plus Performance Test Plan

[i5] Broadband Forum TR114 VDSL2 Performance Test Plan

[i6] ITU-T Rec. T.38 Procedures for real-time Group 3 facsimile communication over IP networks

[i7] ITU-T Rec. V.150.1 Modem-over-IP networks: Procedures for the end-to-end connection of V-series DCEs

[i8] IETF RFC 5194 Framework for Real-Time Text over IP Using the Session Initiation Protocol (SIP)

3 Key words and abbreviations

3.1 Key words

The key words "**shall**", "**shall not**", "**should**", "**should not**", "**may**", "**need not**", "**can**" and "**cannot**", "**will**", "**will not**" in this document are to be interpreted as defined in the ETSI Drafting Rules [1].

3.2 Abbreviations

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

ADC	Analogue to Digital Conversion
ADSL	Asynchronous Digital Subscriber Line
All IP TG	All Internet Protocol Task Group
ATA	Analogue Termination Adapter
ATM	Asynchronous Transfer Mode
CLI	Calling Line Identification
CoS	Class of Service
CP	Communications Provider
CPE	Customer Premises Equipment
DAC	Digital to Analogue Conversion
DECT	Digital Enhanced Cordless Telecommunications
DHCP	Dynamic Host Configuration Protocol
DLM	Dynamic Line Management
DNS	Domain Name Server
DOCSIS	Data Over Cable Service Interface Specification
DSCP	Differentiated Services Code Point
DSL	Digital Subscriber Line
DTMF	Dual-tone Multi-frequency
E2E QoS WG	End-to-End Quality of Service Working Group
EFM	Ethernet in the First Mile
EPOS	Electronic Point of Sale
ETSI	European Telecommunications Standards Institute
FTTC	Fibre To The Cabinet
FTTP	Fibre To The Premises
GoS	Grade of Service
IAM	Initial Address Message
IETF	Internet Engineering Task Force
IP	Internet Protocol
IPDV	IP Packet Delay Variation
ISDN	Integrated Services Digital Network
ISUP	Integrated Services User Part
ITU-T	International Telecommunications Union - Telecommunications Standardization Sector
IVR	Interactive Voice Response
LTE	Long-term Evolution
MBL	Mean Burst Length
MoIP	Modem over Internet Protocol
MOS	Mean Opinion Score
MOS-LQO	MOS - Listening Quality Objective
NGN	Next Generation Network
NICC	Network Interoperability Consultative Committee
NLP	Non-linear Processor
NNI	Network-to-network interface
NTP	Network Termination Point
PBX	Private Branch Exchange
PCM	Pulse-code Modulation
PDD	Post Dial Delay
PESQ	Perceptual Evaluation of Speech Quality
PLC	Packet Loss Concealment

POLQA	Perceptual Objective Listening Quality Assessment
PPP	Point-to-point Protocol
PSTN	Public Switched Telephone Network
QoS	Quality of Service
REN	Ringer Equivalence Number
RFC	Request for Comments
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP-I	SIP ISUP mapping
SMS	Short Message Service
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
ToIP	Text over Internet Protocol
TSG	Technical Steering Group
UK	United Kingdom of Great Britain and Northern Ireland
VBD	Voice-band Data
VCO	Voice Carry Over
VDSL	Very-high-bit-rate Digital Subscriber Line
VLAN	Virtual Local Area Network
VoIP	Voice over Internet Protocol

3.3 Definitions

Clearmode
This term is used to refer generally to a data transfer service providing reliable transmission of an unrestricted 64 kbit/s data stream or more specifically to media sessions using the Clearmode RTP payload type as defined in IETF RFC 4040 [6].

4 General

4.1 Relationship with ND1701

This document describes rules and guidelines governing the end-to-end network performance of interconnected fixed and mobile UK NGNs providing Voice, VBD and/or Clearmode services. NGNs are based on IP technology and the interconnect between them is based on IP.

ND1701 [2] describes rules governing the end-to-end network performance of the UK PSTN (Public Switched Telephone Network) and of public networks evolving towards NGN. The scope of ND1701 [2] covers networks based on TDM (Time Division Multiplexing), ATM (Asynchronous Transfer Mode) and IP (Internet Protocol) technology but, importantly, it only deals with TDM-based interconnect between such networks.

4.2 Service Definitions

NGNs offer fixed and mobile service providers the ability to deliver new and innovative network services that utilise IP connectivity end-to-end. However, over decades of development various applications (broadly classified as Voice-band Data) have emerged that rely on the guaranteed service levels offered by the connection-orientated PSTN that will only be replicated using IP technology if the end-to-end performance characteristics and network component configurations, including the CPE (Customer Premises Equipment) and core network devices that convert to/from IP, are tightly controlled. See also the following from ND1431 [3] regarding VBD CPE support.

> *'The issue of minimising the sensitivity of VBD CPE to the various CPE compatibility failure mechanisms that can be encountered on NGNs and NGAs is primarily a result of having to deal with a legacy installed base of VBD CPE outstations, VBD CPE central stations, and of the continuing production of legacy VBD based CPE designs. Given the widespread and increasing availability of broadband connections in the UK, consideration should be given to the change-out of VBD CPE to IP based equivalent CPE outstations and receiving centres. The change to IP based equipment not only avoids potential VBD incompatibilities, but also provides considerable opportunities for increased functionality and flexibility in design and features offered to both the end user and the service operator.'*

Since the publication of ND1431 [3], some VBD CPE has been, or has begun to be, phased out. ND1704 Annex I provides a summary of known Voice and VBD CPE and applications with proposals for actions to be taken in the migration to NGNs engineered to support only Voice service (see the definitions below). Annex II summarises some common sources of end-to-end performance and service impairments which CPs (Communications Providers) might encounter in the provision of Voice, VBD and Clearmode services over NGNs.

At the same time, there have been significant advances in codec technology for delivering higher quality voice services (e.g. through extension of the audio bandwidth to wideband and super-wideband) and, to an extent, concealing or correcting for IP related degradations whilst minimising bit rate. This means that the performance requirements for Voice services are less stringent than the conditions required to support VBD and Clearmode services.

Therefore, this document sets out different rules and guidelines for three categories of service as given below. CPs **should** consider which service categories apply to their networks and services when determining end-to-end performance requirements.

- Voice service: a service that supports reliable transmission of human speech. Special provisions should be made for ongoing support of some data services such as DTMF (Dual-tone Multi-frequency) for IVR (Interactive Voice Response) purposes. Reliable transmission of VBD (including machine-machine DTMF applications) cannot be guaranteed.

- VBD service: a service that supports reliable transmission of voice-band data via in-band methods (including the machine-machine DTMF applications excluded from the Voice service category).

- Clearmode service: a service that supports reliable transmission of 64 kbit/s data streams.

Rules and guidelines that are associated with a particular definition are highlighted as such, otherwise the stated guidelines apply to all service categories. NGN support for a service category entails compliance with all the applicable rules and guidelines associated with that category.

Support for the Voice service category is the minimum requirement for UK NGNs. Support for VBD and/or Clearmode services is a choice for each individual CP; there is no obligation on a CP to support the more demanding service categories. With the withdrawal of legacy TDM networks, it is likely that only Voice service will be widely supported by UK NGNs.

It should be noted that in practice a CP may choose to implement some measures which are conducive to more successful operation of e.g. VBD service, while not fully supporting the performance necessary to guarantee reliable transmission of the service in all cases. It is likely that ongoing support for VBD service by UK NGNs will be mostly on this kind of “best effort” basis.

It should be further noted that where a call path involves more than one NGN, the end-to-end service performance depends upon all of the CPs involved. Hence, even if a CP has engineered their NGN to support e.g. VBD service, (in the absence of deterministic routeing and explicit inter-CP agreements) reliable transmission may not be guaranteed where the call path involves other networks.

4.3 Assumptions Regarding the Maximum Number of Interconnected NGNs

Certain aspects of network performance (e.g. packet delay and error ratios, de-jitter buffer dimensioning) are a function of the number of CP networks in a call path. It is not technically possible to ascertain the number of CPs involved in a path on a call-by-call basis. In order to make recommendations regarding these aspects of performance this document assumes that a maximum of six CP networks will be involved in any call between interconnected NGNs in the UK.

4.4 Enterprise Networks

Enterprise networks are outside the scope of this document but the existence of enterprise networks cannot be ignored in network performance planning because they will have an effect on end-to-end impairments. Historically the NTP-NTP (Network Termination Point) delay performance of the PSTN was below 15 ms and enterprise network operators have been able to use this low delay to allow the implementation of low bit rate codecs and IP technology, and still achieve ‘acceptable’ quality for calls from the enterprise network to the PSTN/ISDN (Integrated Services Digital Network). As IP technology is deployed in NGNs, enterprise network operators are encouraged to bear in mind the recommendations of this document concerning delay in the public network.

4.5 Reference Connection

This document assumes that end-to-end IP connections between users' terminals are described by a reference connection which in the simplest case is that shown in Fig. 1. The access network, at the left of the figure, is the source access for packets travelling left to right. The access network, at the right of the figure, is the sink access for packets travelling left to right.

It is assumed that inter-CP interconnects are located between the CPs' core networks. The operator of the source core network might own the source access network or might obtain access from an independent access provider, but in either case the connection between the access network and the core network is likely to be both technically and commercially different from the symmetric connection characteristic of CP interconnects. Hence, a CP needs, at least, a vestigial core network to adapt an access-to-core boundary to an inter-CP interconnect. In the diagrams, source and sink access networks are shown bound to core networks.

In Figure 1 the single NNI (Network-to-network interface) marked NNI 1 is both the source NNI and the sink NNI.

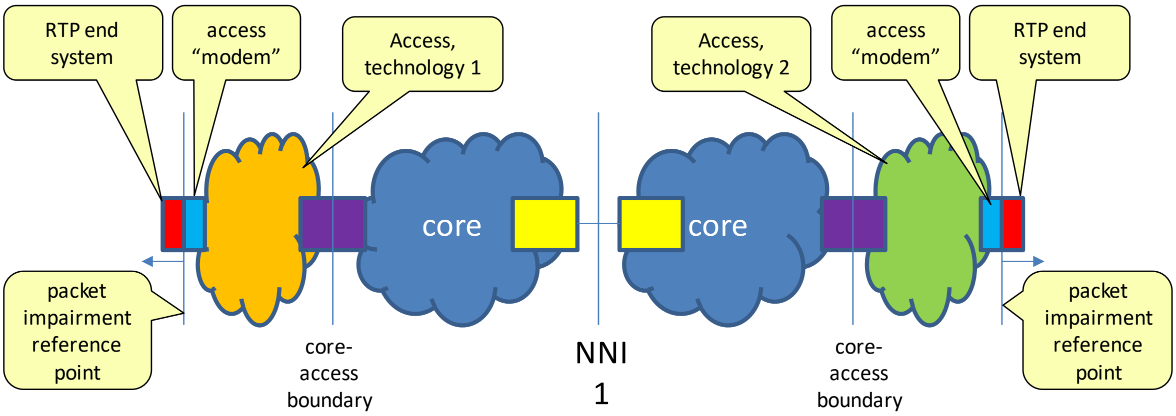


Figure 1: Simplest access-to-access end-to-end IP Connection

Figure 2 shows the most onerous case considered in this document where four transit networks are interposed between the source and sink networks, giving rise to five NNIs marked NNI 1 to NNI 5.

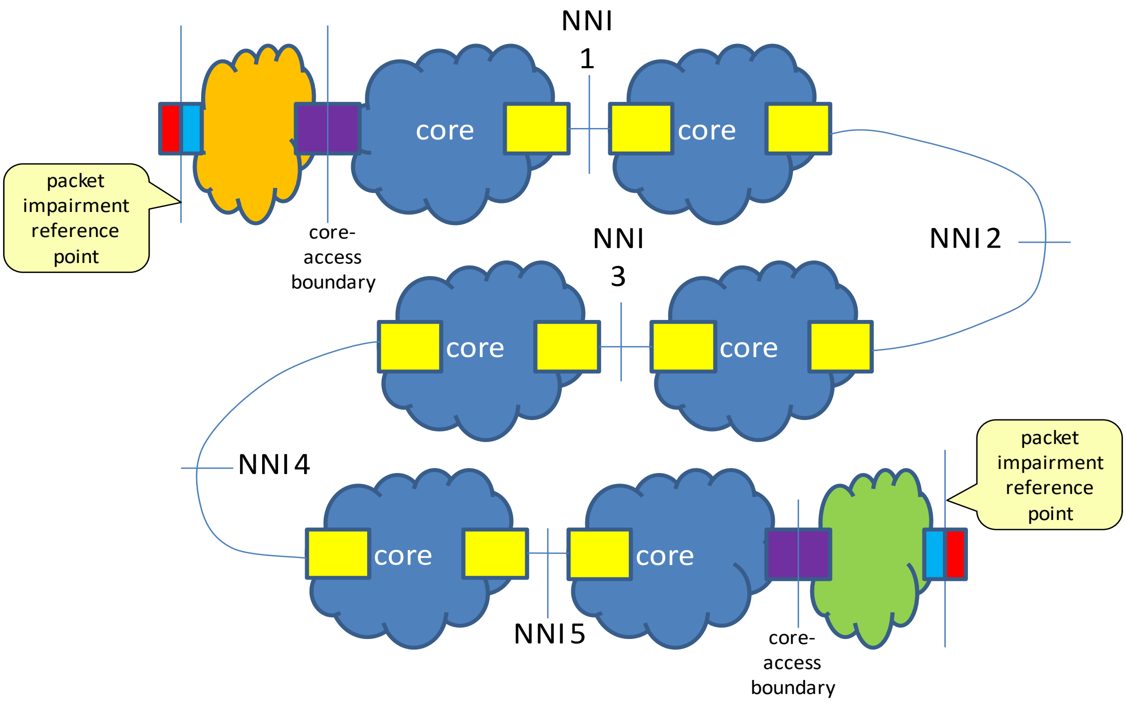


Figure 2: End-to-end IP Connection with Four Transit Networks

An especial focus applies to the accumulated packet stream impairments present at the final NNI crossed by a packet stream, that is, the NNI into the sink core network. This is because the operator of the sink network is responsible for the configuration of the RTP (Real-time Transport Protocol) receiver which, for good service quality, **shall** be compatible with impairments in the packet stream reaching the RTP receiver across the ‘packet impairment reference point’ marked in the diagrams. The operator of the sink network **should** also be aware of the impairment characteristics both of their core network and of the sink access. The impairments in the packet stream reaching the RTP receiver comprise two elements:

- Impairments within the control of the operator of the sink network, that is, impairments in the operator's own core network and the sink access
- Impairments outside the control of the operator of the sink network, that is, impairments already in the packet stream reaching the NNI into their network

A key objective of this document is to inform operators of sink networks about the expected impairment characteristics of packet streams crossing NNIs into their networks for those cases where the service, the source access and intermediate core networks are in scope of this document.

4.6 Access technologies in scope of this document

The following access technologies are in scope of this document.

- Fibre To The Premises (FTTP)
- Very-high-bit-rate DSL (VDSL)
- Asynchronous DSL (ADSL) - for further study
- G.Fast
- DOCSIS (Data Over Cable Service Interface Specification)
- 2G Circuit Switched
- 3G Circuit Switched
- 4G/5G LTE
- Wi-Fi – for further study
- High speed point-to-point

5 Network Performance Rules and Objectives

5.1 Packet Delay and Delay Variation Rules

Rule PD1. Transmission delay **shall** be minimised.

Rule PD2. IP packet delay variation **shall** be minimised.

Rule PD3. IP de-jitter buffer delay **shall** be minimised.

There is a trade-off between de-jitter buffer size (and therefore end-to-end delay) and packet loss, with shorter de-jitter buffer delays potentially resulting in higher packet loss. This **should** be taken into account when optimising de-jitter buffer parameters.

Rule PD4. A de-jitter function **shall** be provided at the point of transition from a packetised voice bit stream to a synchronous voice bit stream. For calls within the UK NGN in the scope of this document, de-jitter functions **should not** be applied on a per network element basis or at the boundary between two IP-based networks interconnected with IP.

A de-jitter buffer **may** additionally be used if required at points where the codec type or packetisation time differs between two IP networks.

Rule PD5. In addition to the de-jitter function being provided in accordance with Rule PD4, additional de-jitter buffers **may** be provided on the source side of an NNI into an ND1704-compliant network to reduce IPDV (IP Packet Delay Variation) for media streams entering the ND1704-compliant network. The provision of this additional de-jitter buffering is under the control of the operator originating the media stream.

Rule PD6. In order that the number of 'packetisations', and associated delays, are kept to a minimum, NGNs in the UK **should** be interconnected using IP.

Rule PD7. In order to help minimise delay (as well as the impairment of voice quality) where the packetisation time between IP networks differs (i.e. transrating is required) but the same codec is used, interworking **should** be performed without transcoding (i.e. the payload **should** be re-sized losslessly).

Rule PD8. The delay introduced by the following functions **shall** be minimised. Each function **should** occur no more than once in the call path, although additional Packet Loss Concealment (PLC) functions **may** be applied when transcoding is employed

- ADC (Analogue to Digital Conversion) and DAC (Digital to Analogue Conversion)
- Echo cancellation
- PLC (Packet loss concealment)

Rule PD9. Each network in a call path **shall** introduce no more than maximum 2*(0.01*Distance, km) ms of propagation delay, where 'Distance' is the crow's flight distance in km between ingress and egress NNIs and the constant 2 is an allowance for situations where the ingress and egress NNIs are co-located but the call path is routed elsewhere.

CPs **shall** design and operate their network in such a way as to minimise total propagation delay (e.g. by appropriate choice of network architecture, interconnects and routeing).

Rule PD10. The overall effect across all networks of propagation delay, ADC, DAC, echo cancellation and PLC **should** be less than 10 ms delay for all UK calls.

Rule PD11. <u>Voice</u>. In order to help minimise delay, adaptive de-jitter buffers **shall** be used for voice service and **shall** be dimensioned by the worst-case source access network and by the specific sink access network to handle the delay variation introduced to the packet flow by the sink network and all preceding networks (up to a maximum of six ND1704 compliant core networks in total).

Adaptive de-jitter buffer algorithms vary and might sometimes add a constant delay above the best-case minimum, or might add delay proportional to the spread of the delay variation they are handling. Such a buffer **may** adapt to apply a minimal additional delay to the maximally delayed packet.

Rather than assume 'perfect' adaptive de-jitter buffer behaviour, it is assumed for the purpose of the ND1704 Performance Calculator (Annex IV) that an adaptive de-jitter buffer might typically size itself at 150% of the observed IPDV or less.

Rule PD12. VBD. When the presence of a call involving voice-band data is detected, a fixed de-jitter buffer **shall** be used. The de-jitter buffer **shall** remain fixed even if the call reverts to VCO (Voice Carry Over) when it would normally operate in adaptive mode.

See also ND1431 [3] for information regarding VBD detection.

Rule PD13. VBD. Source clocks, network clocks and sink clocks **should** be compliant with ITU-T Rec. G.811 [4]. This includes ensuring CPE clocks are synchronised.

Excessive packet wander/jitter and/or synchronisation differences between the source clock, network clocks and the sink clock may lead to packet loss due to de-jitter buffer overruns or under-runs. The contribution of such losses to overall packet loss will be negligible if ITU-T Rec. G.811 [4] clocks are employed. Where it is not possible to ensure CPE is synchronised, a CP will not be able to guarantee a VBD service meeting the criteria defined in this document.

Rule PD14. Clearmode. When signalling indicates the use of Clearmode a fixed de-jitter buffer **shall** be used.

Quantitative objectives for delay and delay variation are incorporated in the ND1704 Performance Calculator (Annex IV).

5.2 Codec Rules

Rule C1. VBD. When the presence of a call involving voice-band data (e.g. fax, modem etc.) is detected, the ITU-T Rec. G.711 A-law codec [5], without silence suppression **shall** be supported end-to-end for interconnecting NGNs (without transcoding or transrating) and 10 ms packetisation time **should** be used to minimise delay.

10 ms packetisation time is recommended for VBD (and Clearmode) services where media path delay must be tightly constrained to avoid causing issues for associated CPE and/or receiving centres (see ND1431 [3] for more information).

Rule C2. Voice. The ITU-T Rec. G.711 A-law codec [5], without silence suppression, **shall** be supported as the default codec for interoperability, however, the choice of codecs which **may** be negotiated for voice calls is not limited to ITU-T Rec. G.711 A-law. For example, the use of modern codecs that provide increased audio bandwidth, low bit rates and (to an extent) resilience to errors and packet loss is encouraged to improve perceived call quality. Where possible, codec negotiation **should** also aim to avoid delay and/or coding impairments caused by:

- Multiple packetisations
- Transcoding
- Silence suppression

Rule C3. VBD. An offer/answer model to establish and update multimedia sessions using the Session Description Protocol (SDP) **shall** be employed.

Rule C4. Clearmode. Interconnecting NGNs **shall** support the Clearmode RTP payload type as defined in IETF RFC 4040 [6]. When signalling indicates the use of Clearmode, 10 ms packetisation time **should** be used to minimise delay.

Rule C5. When choosing to employ a Codec Translation Function (i.e. transcoding) the speech quality resulting from the delays and choice of codec **must** be considered in order that end-to-end impairments are minimised.

Codec translation within the call media path is undesirable unless it is to enable a call connection that would otherwise be impossible due to incompatible codecs. The Codec Translation Function introduces delay and potential conversion impairment. The recommended national delay objectives for connections as per ND1701 [2] are applicable to a call subject to codec translation. For such calls it is recommended that the routeing of a call is as determinate as possible to ensure that end to end call delay maxima are not exceeded.

A CP may introduce a Codec Translation Function in their own fixed or mobile network, or direct calls to a transit network which provides such a function, based on known routeing. Where provision of the Codec Translation Function in an applicable fixed network exceeds the delay allocation indicated by the ND1704 Performance Calculator (Annex IV) it is recommended that the total delay introduced by the CP's network **should** not exceed a delay objective of 35 ms. Exceptionally, this figure may be exceeded where the call routeing is so determinate that the call will meet the appropriate recommended national end to end delay objective given in ND1701 [2].

5.3 Packet Loss Rules

Rule PL1. VBD. The support of VBD requires more stringent packet loss objectives than the support of voice traffic. A packet loss ratio of around $1x10^{-6}$ per CP, that is characteristically non-bursty, **shall** be achieved for voice-band data.

As noted in PD13, excessive packet wander/jitter and/or synchronisation differences between the source clock, network clocks and the sink clock may lead to packet loss due to de-jitter buffer overruns or under-runs. The contribution of such losses to overall packet loss will be negligible if ITU-T Rec. G.811 [4] clocks are employed. Where it is not possible to ensure CPE is synchronised, a CP will not be able to guarantee a VBD service meeting the criteria defined in this document.

Rule PL2. Clearmode. The support of Clearmode in NGNs requires more stringent packet loss objectives than the support of VBD and voice traffic. NGN packet loss for support of Clearmode **shall** be minimised and **should not** exceed a packet loss ratio of $1x10^{-7}$ per CP.

Rule PL3. Voice. CPs **should** establish suitable packet loss thresholds based on the information provided in the ND1704 Performance Calculator (Annex IV) or use their own data to establish suitable thresholds for their Voice service networks.

The support of voice only services in NGNs requires less stringent packet loss objectives than the support of VBD and Clearmode services. The subjective tolerance to specific loss ratios and bursty profiles is codec dependent (including the provision of error correction and/or packet loss concealment techniques).

Annex III provides details of an approach to measuring voice quality under different burst conditions for different codecs. MOS-LQO (Mean Opinion Score – Listening Quality Objective) measurements for a range of codecs based on this approach are incorporated in the ND1704 Performance Calculator (Annex IV).

Rule PL4. Voice. Packet Loss Concealment (PLC) **should** be applied. In NGNs the CP terminating the multi-CP IP connection segment **shall** be responsible for PLC. Even though the terminating CP network meets its individual objective for packet loss, the contribution from preceding networks in the multi-CP NGN connection segment might result in an overall level of packet loss which makes PLC necessary.

Quantitative objectives for packet loss are incorporated in the ND1704 Performance Calculator (Annex IV).

5.4 Echo Control Rules

Rule EC1. Echo cancellation **shall** be employed in NGNs as end-to-end delay will be above 25 ms. Echo cancellers **should** be co-located with the codec functions.

Rule EC2. Echo cancellers compliant with the requirements of ITU-T Rec. G.168 [7] **shall** be employed.

Rule EC3. The presence of an echo canceller in a call path **shall** be signalled to adjacent networks in accordance with ITU-T Rec. Q.115.1 [8] where the signalling system allows. Ideally, all echo cancellers **should** be disabled with the exception of the two closest to the two potential sources of echo. Where explicit signalling information about the presence/location of echo cancellers is not available, the decision to enable/disable local echo cancellers **should** be based on appropriate assumptions about the behaviour of the other networks in the call path.

Where explicit signalling information about the presence/location of echo cancellers is not available there is potential for the presence of more than one echo canceller working in the same direction of a call path. If G.168 [7] compliant echo cancellers are used then this will not degrade speech quality.

Rule EC4. In general, the tail capacity **shall** be at least 64 ms to prevent any potential failure to cancel echo in some routeing scenarios. As an exception to this, the tail capacity may be smaller when the echo path is known to be less than 64 ms.

Rule EC5. VBD. For Non ITU-T modulation schemes that do not have answer tone, a robust means of signal classification **shall** be employed (see ND1431 [3] for more information). In the absence of ability to detect answer tones, CPs **should** attempt to mitigate risk associated with the de-jitter buffer not being made fixed, through the optimisation of the adaptive settings and ensuring that the echo canceller NLP (Non-linear Processor) remains in-circuit.

Note that much detail regarding the use of echo cancellers is provided in ITU-T Rec. G.108.2 [i2].

5.5 Post Dial Delay

NGN calls are based on different technology to that used in the TDM-based PSTN. From a service perspective it is important that customer perception of connection processing performance parameters, such as Post Dial Delay (PDD) in NGNs, **should** be comparable to that of the PSTN.

5.5.1 Definition of Post Dial Delay

Post Dial Delay is experienced by the originating customer as the time from the sending of the final dialled digit to the point at which they hear ring tone or other network tones and announcements. Where the originating network is required to play an announcement before completing the call then this definition of PDD excludes the duration of such announcements.

Each network that forms part of the call path contributes to the overall experience. Although more complex routeings are potentially possible, the representative ‘worst-case’ call set-up used for the apportionment model below involves six network ‘stages’, of which two involve location-related services (for example: ‘Find me anywhere’ or mobile telephony). Under this assumption, all calls will naturally involve both an Originating and Terminating stage.

5.5.2 Assumptions

The PDD apportionment model makes the following assumptions:

1. Overlap signalling is not used. In general terms SIP-I (Session Initiation Protocol - Integrated Services User Part mapping) ‘overlap signalling’ is strongly discouraged due to the risks of imposing additional processing load on signalling devices arising from increased messages per call.

2. Normal PSTN operation is assumed, where the return media path is established based on the forward signalling and ring-tone is generated at the far end. Local (‘near end’) ring-tone generation is technically possible. It is noted that, for example, SIP defaults to local application of ring-tone. It is important that operators of NGNs consider their SIP configuration, particularly when interworking with legacy networks, to ensure that callers do in fact receive ring-tone (ideally from only one source).

3. In general, a particular network has no knowledge or control over the performance of the other networks involved in delivering a call.

5.5.3 Apportionment Model

It is appropriate to consider the behaviour of interconnected networks at the network boundaries. It is recognised that there are many possible approaches to the implementation of signalling, however it is expected that the following simplified model **will** cover the majority of cases (and the impact of other cases is noted for information).

The apportionment model is that, as call set up progresses, there are clearly defined handovers of call setup control in each of the forward and backwards directions at each network boundary.

The Forward PDD element of a network **should** be viewed as the sum of delays for processing all messaging up to the point the far end is able to alert the calling party. The Backward PDD element **should** be viewed as the sum of delays for processing all messages up to the point that the calling party receives the network response. The total PDD contribution of a network is the sum of all Forward and Backward elements.

The target forward and backward contributions for each network are based upon the functions that each network is expected to perform in a given call path and are identified as the time difference between Input and Output signalling events. It is assumed that optimal routeing is used. The recognised functions are shown in Tables 1 and 2.

Table 1: PDD Functions Associated with the Forward Path

Network	Input Event	Output Event	Network Functions
Originating	End of customer dialling	Initial Address Message (IAM)/Invite to next network	Authentication; Routeing
Transit	IAM/Invite from preceding network	IAM/Invite to next network	Routeing (simple) Note: ‘simple’ means that the destination was known when the call was passed to the transit network
Terminating	IAM/Invite from preceding network	Alerting destination	Identify customer; internal routeing; alerting.
Mobility	Service trigger	Instruction to serving network	Locate mobile terminal
Number Translation Service	Service trigger	True Destination	Number translation services

Table 2: PDD Functions Associated with the Backward Path

Network	Input	Output	Network Functions
Terminating	Alerting destination	Return ring-tone to preceding network	Connect media path; generate ring-tone
Transit	Ring-tone	Ring-tone	Connect media path
Originating	Ring-tone	Ring-tone	Connect media path

5.5.4 Targets

The PDD target is in two parts: the 95th percentile of the PDD distribution (i.e. representing the situation that when meeting the target no more than 5% of calls will experience longer than the specified PDD); and the target maximum value that any call should experience. This approach recognises that the PDD will depend upon the (variable) instantaneous load.

CPs could be expected to demonstrate that they meet the target 95th percentile across the mix of calls under normal operation; the maximum figure is considered a design target and in the event that a particular call exceeds this target there is no requirement to abort the establishment of that call. A CP that exceeds their target allowance **will** be expected to bring their contribution back to the specified target by appropriate engineering methods (e.g. equipment improvements or architectural changes).

Network designers **shall** take account of the fact that under adverse load conditions (overload) it is vital to manage the network in order to constrain PDD and prevent worsening of the overload due to high customer abandonment rates.

Clearly, PDD **will** increase in instances of overload but failure to constrain PDD risks high levels of customer abandonment and repeat attempts, which could actually lead to worsening of the overload by increasing the level of ineffective processing.

Overload is effectively the situation in which the processing load on elements of the network is high enough that it is likely that normal load PDD targets (identified in Table 3) **will** be breached, in which case it is reasonable that higher targets (for further study) **should** apply.

Table 3: Normal Load PDD Targets

Network 'stage'	**PDD (95th Percentile) (Sum of forward and backward path contributions)**	**PDD (Target maximum all calls) (Sum of forward and backward path contributions)**
Originating	250 ms	500 ms
Transit	100 ms	200 ms
Mobility Service		10 s (See Note 1)
Number Translation Service		1 s
Termination (fixed line)	200 ms	400 ms
Note 1: Whilst it is recognised that mobility services **will** in general require more time for PDD than Number Translation Services, the additional time in the target is intended only to allow the CP the necessary allowance to locate the destination customer/terminal and is not intended as a general consent to longer PDD without good cause.		

Some complex services may be out of scope of these targets. Such services will be characterised by being of interest to 'sophisticated' end users who will recognise that the service complexity will result in comparatively long set-up times, might be subscription only and may not be 'mass market'.

5.6 Stability & Reliability Rules

Rule SR1. Care **shall** be taken to ensure that the performance of a service is not adversely affected by other traffic types. NGNs **shall** preserve Grade of Service (GoS) even under high or rapidly fluctuating load and/or mix of other traffic types.

Achieving this end-to-end, potentially across multiple NNIs and access networks that are owned by different operators, requires universal agreement that signalling and voice traffic is prioritised accordingly. For example, between operator NNIs this may involve appropriate VLAN (Virtual Local Area Network) tagging, priority queuing of layer 3 DSCP (Differentiated Services Code Point) marked packets and/or a mapping between the two. Across source and sink access networks this may involve the priority queuing of layer 2 CoS (Class of Service) and layer 3 DSCP marked packets and/or mapping between the two.

Rule SR2. Operators **shall** minimise re-synchronisations and other mechanisms, such as ATA (Analogue Termination Adapter) and/or modem power cycling, that cause momentary service outages (see also SR3). Care **shall** be taken to ensure that any forced events occur during a period of inactivity in order to minimise user impact.

Operators may consider running pre-migration commissioning tests on a per user basis to ensure (a) basic end-to-end service availability and (b) that access performance is fit for purpose. Such tests, particularly in the case of (b), could be considered essential where migration involves the move from analogue to digital signal transmission over the source and/or sink access networks. Such lines could suffer from performance anomalies that were problematic for previous analogue transmission (e.g. resulting in a 'noisy' line) that translate to catastrophic issues (e.g. loss of service) for digital transmission.

It might not be practicable for CPs to guarantee the long-term frequency of events leading to loss of service (for example, due to faulty terminal equipment). Users should be made aware of these limitations and they should be prepared to use an alternative means of communication in the event of an emergency (such as a mobile phone) operators may consider building in back-up options (such as fixed line modems that also include mobile network access).

Rule SR3. When physical layer re-synchronisation occurs on ADSL, VDSL or G.Fast broadband access lines, this **should** occur within the time limits specified in the relevant Broadband Forum Test Documents (for example Broadband Forum TR067 [i3] specifies 60 seconds for ADSL, TR100 [i4] specifies 60 second (single mode) and 120 seconds (auto mode) for ADSL2/2plus and TR114 [i5] specifies 90 seconds for VDSL2). The delay associated with higher layer processes involved in the provision of service, such as PPP (Point-to-point Protocol), DHCP (Dynamic Host Configuration Protocol), DNS (Domain Name Server) etc. is expected to be in the order of milliseconds.

Operators **should** consider running ‘stable’ DLM (Dynamic Line Management) profiles when supporting Voice, VBD or Clearmode services in order to minimise forced and unforced synchronisations. Whilst this might be at the expense of absolute instantaneous throughput (e.g. versus profiles optimised for ‘speed’) a stable profile may be necessary for optimising the performance of real-time services, and could result in better long-term performance overall.

Rule SR4. Where access services rely on mains powered customer premises equipment, operators **should** either provide a battery backup solution or **should** make users aware that service will be lost in the event of a power-cut and they should be prepared to use an alternative means of communication in the event of an emergency (such as a mobile phone).

Rule SR5. Asymmetric routeing **shall** be avoided.

5.7 DTMF and Voice-band Data Considerations

Rule VBD1. VBD. DTMF, fax and data **should** be carried in-band using G.711 (see Rule C1) to reliably handle time sensitive, machine-machine transmission of DTMF, fax and data.

Rule VBD2. Voice. RTP Telephony Events **should** be used to carry DTMF (in preference to in-band transmission) by all VoIP (Voice over Internet Protocol) terminals to provide resilience to packet loss and remove reliance on the use of G.711 coding.

Rule VBD3. RTP Telephony Event support **shall** include dialled digits 0-9, A-D, *, #. Conversion between different types of DTMF (e.g. in-band to/from RTP Telephony Events) **should** be avoided where possible, but where interworking is required, timing and volume information **should** be accurately conveyed to avoid subsequent issues with digit detection.

Although out of band methods exist for supporting fax, text telephony and other data (such as ITU-T Rec. T.38 [i6], ITU-T V.150.1 MoIP (Modem over IP) [i7], IETF RFC 5194 ToIP (Text over IP) [i8] etc.), such applications are discouraged in favour of native IP data solutions as outlined in Annex I.

Whilst IETF RFC 4733 [9] is the main Internet standard for RTP Telephony events, also refer to ND1444 [10] for DTMF best practice.

5.8 Quantitative Performance Objectives

The ND1704 Performance Calculator (Annex IV) allows the end-to-end impairments (fixed delay, delay variation and packet loss) to be estimated for different source and sink access configurations, source packetisation times and transit network configurations. For Voice service networks it also includes MOS-LQO measurements for a range of codecs under bursty packet loss conditions.

Annex I: Voice Only Services - Voice/VBD CPE & Applications Impact and Migration Options

Table A1: Voice/VBD CPE & Applications Impact and Migration Options

CPE/Application	Modulation (if known)	Source	Classification	Formal Support	Mitigation Options	Notes
Satellite TV boxes	V.34, V.90	ND1431	VBD	No	Migrate to native Ethernet solutions	Being phased out
Fax machines	V.17, V27ter, V.29, V.34	ND1431	VBD	Partial	Support with fax relay (reliability not guaranteed/typically low speed only)	Encourage migration to more suitable alternative (e.g. secure email)
Dial-up Internet access	V.34, V.90 and V.92	ND1431	VBD	No	Migrate to native Ethernet solutions	Being phased out
Security alarm panels	DTMF, V.34	ND1431	VBD & DTMF	No	Migrate to native Ethernet solutions	DTMF relay is not recommended for time critical machine-machine applications, but is the preferred mechanism for human-machine applications for robustness to packet loss.
Telecare social alarms	DTMF	ND1431	DTMF	No	Migrate to native Ethernet solutions	
Lift Emergency Phones	DTMF, V.34	ND1431	VBD & DTMF	No	Migrate to IP phones	
EPOS (Electronic Point of Sale) card sale terminals	V.22bis, V.34	ND1431	VBD	No	Migrate to native Ethernet solutions	
Pre-payment card top-up machines	V.22bis, V.34	ND1431	VBD	No	Migrate to native Ethernet solutions	
Fire alarms	DTMF, V.34	ND1431	VBD & DTMF	No	Migrate to native Ethernet solutions	DTMF relay is not recommended for time critical machine-machine applications, but is the preferred mechanism for human-machine applications for robustness to packet loss.
Automated Teller (cash) Machines	V.22bis, V.34	ND1431	VBD	No	Migrate to native Ethernet solutions	
Telemetry remote monitoring	V.22, V.22bis, V.23, V.34	ND1431	VBD	No	Migrate to native Ethernet solutions	
SMS (Short Message Service) phones	V.23	ND1431	VBD	No	Alert users to loss of service	No options available - feature **will** be obsolete on applicable CPE.
Text Phones	V.21, Baudot	ND1431	VBD	No	Migrate to native Ethernet solutions	E.g. Apps on tablets/smartphones
Telehealth (eHealth) remote monitoring	V.34, V.90, V.92	ND1431	VBD	No	Migrate to native Ethernet solutions	
Payphone remote management	V.34	ND1431	VBD	No	Migrate to native Ethernet solutions	
PBX (Private Branch Exchange) remote management	V.34	ND1431	VBD	No	Migrate to native Ethernet solutions	
Custodial tagging devices	V.34	ND1431	VBD	No	Migrate to native Ethernet solutions	
CLI (Calling Line Identification) phones, display boxes and other CP	V.23	ND1431	VBD	Yes	SIP-V.23 managed locally by ATA	
Corded Phones		ND1431	Voice	Partial	Basic voice transmission supported in all cases. DTMF to be converted to/from RTP Telephony Events by ATA.	Slow migration to IP phones. DECT (Digital Enhanced Cordless Telecommunications) remains applicable to air interface.
Cordless Phones		ND1431	Voice			
Analogue Cordless Phone		ND1431	Voice			
Digital / DECT Cordless Phones		ND1431	Voice			
Answering Machines		ND1431	Voice			

CPE/Application	Modulation (if known)	Source	Classification	Formal Support	Mitigation Options	Notes
CLI Phones		ND1431	Voice			
CLI Display Boxes		ND1431	Voice			
Audio Conferencing Equipment		ND1431	Voice			
Auto diallers		ND1431	Voice			
indirect Access Code Auto diallers		ND1431	Voice			
Hotline Auto diallers		ND1431	Voice			
Call Barring Devices		ND1431	Voice			
Distinctive Ringing Devices		ND1431	Voice			
REN (Ringer Equivalence Number) Boosters		ND1431	Voice			
Extension Ringers		ND1431	Voice			
Payphones		ND1431	Voice			
Private Payphones		ND1431	Voice			
Public Payphones		ND1431	Voice			
Multiple Line Phones / Mini PBXs		ND1431	Voice			
Call Loggers		ND1431	Voice			
Test Equipment		ND1431	Voice			
Network Monitoring and Alarm Equipment		ND1431	Voice			
Line Isolating Units		ND1431	Voice			
Phones with Pulse Dialling		NEW	Voice	No	Alert users to loss of service	No options available (ATAs do not typically support rotary dial CPE) therefore this type of CPE will be obsolete.

Annex II: Sources of Performance and Service Impairment

The table below summarises common sources of performance and service impairments CPs might encounter in the provision of Voice, VBD and Clearmode services over NGNs. This is not an exhaustive list and it deliberately excludes ‘edge-case’ fault conditions, but it serves to highlight the wide range of contributing factors that might impact service performance and service availability.

Table A2: Performance and Service Impairments associated with NGN networks and terminal equipment

	SOURCE				SINK		
	RTP End System	**Access Modem**	**Access Technology 1**	**Core & NNI**	**Access Technology 2**	**Access Modem**	**RTP End System**
DELAY/DELAY VARIATION	A-D & encoding	Queuing/buffering	Retransmission/interleaving	Transcoding	As per source	As per source	De-jitter buffering
	Packetisation		Queuing/buffering	De-jitter buffering			De-packetisation
	Encryption		Propagation	Propagation			Decryption
	Serialisation			Queuing/buffering			Decoding & D-A
	Source/scheduling jitter						
PACKET LOSS	Imperfect premises wiring	LAN congestion[5]	Aggregator congestion	Network Congestion	As per source	As per source	Unsynchronised source/network/sink
		RF interference	Imperfect wiring/joints				Imperfect premises wiring
			Cable loss, exp(-d)				De-jitter buffer discards
			Cable propagation impairments				
			DLM Trade-offs				
			Radio loss, $1/d^2$				
			Radio propagation impairments				
FUNCTIONAL ISSUES	Premises power-cut	Premises power-cut	Loss of sync between modems	Limited/no VBD discrimination	As per source	As per source	As per source
	Recoverable fault/crash	Recoverable fault/crash		No detection of ISDN			
	Irrecoverable fault	Irrecoverable fault					
	Limited/no VBD discrimination[1]						
	No detection of ISDN[2]						
	SDP mismatch[3]						
	Unsupported CPE[4]						

[1] E.g. failure to fix de-jitter buffers, negotiate G.711 coding, configure echo canceller etc. in accordance with NICC ND1431 Guidance on CPE Compatibility on NGNs and NGAs

[2] E.g. failure to meet IETF RFC4040 RTP Payload Format for a 64 kbit/s Transparent Call

[3] E.g. failure of endpoints to negotiate and/or network to adequately interwork different p-times, DTMF, fax and/or modem in-band/relay methods etc.

[4] E.g. pulse dialling from rotary telephones

[5] Including overload/misuse of priority (e.g. EF) queue

Annex III: Voice Services - A method for objective voice quality measurement of bursty packet loss

A3.1 Introduction

Average packet loss (typically expressed as the ratio of lost packets to transmitted packets converted to a percentage) is a common way of reporting packet loss for a given IP network or service. This is a useful metric when packet loss is random (i.e. where there are singular loss events at random and sufficiently spaced time intervals). However, when losses occur concurrently, or with near concurrence, this is not reflected in the average loss statistics alone.

It stands to reason that for voice services, a degree of random loss could be more easily concealed through interpolation or error correction than bursty loss. However, qualitative data that accounts for the performance of specific packet loss concealment algorithms and error correction techniques is not widely available. This is a barrier to setting meaningful acceptability thresholds for packet loss in general.

This Annex describes two different models for defining packet loss burstiness along with a method for measuring the associated voice quality impact for different codecs with different loss resilience and recovery mechanisms (such as packet loss concealment and forward error correction).

This is applied in the ND1704 Performance Calculator (Annex IV) for the purposes of generating indicative voice quality measurements. The inclusion of other codecs and concealment techniques, as well as different approaches to modelling burstiness and measuring the impact on voice quality, is for further study.

A3.2 Models for packet loss burstiness in VoIP RTP streams

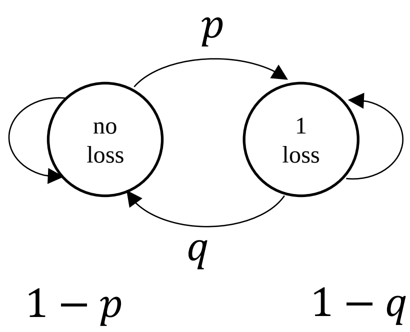
Various models have been developed to measure the burstiness of packet loss in VoIP RTP streams. Two examples are IETF RFC 3611 (RTP Control Protocol Extended Reports - RTCP XR) and Burst Ratio (Burst ratio - BurstR: a measure of bursty loss on packet-based networks. U.S. Patent 6,931,017, McGowan, J.W., Lucent Technologies Inc. 2005).

IETF RFC 3611 classifies different periods of an RTP stream depending on the proportion of lost and discarded packets in bursts or gaps, where a burst is a period with a high proportion of lost and/or discarded packets and gaps are periods with a low proportion of lost and/or discarded packets.

A burst is usually defined in terms of a value, Gmin (gap threshold) as the longest sequence that (i) starts with a lost or discarded packet, (ii) does not contain any occurrences of Gmin or more consecutively received (and not discarded) packets and (iii) ends with a lost or discarded packet. Anything that is not a burst in an RTP stream is essentially classified as a gap. The recommended value of Gmin according to IETF RFC 3611 is 16 (i.e. gaps will have a minimum distance of 16 successfully received packets between losses whereas bursts are any periods with less than 16 successfully received packets between losses).

BurstR is the ratio of the average length of observed bursts in a packet arrival sequence over the average length of bursts expected for a random loss, packet-based network. It is usually represented in the context of a 2-state Markov model as shown in Figure A1.

Figure A1: Markov 2-state model



The Markov 2-state model has two distinct states: no loss and 1 loss, which respectively highlights when a packet has been successfully received or lost in an RTP stream. The model is fully characterised by two values, p and q where p is the probability of transition between the no loss state to the 1 loss state and q is the probability of transition between the 1 loss state to the no loss state. Given the values of p and q, BurstR can be defined as follows:

$$BurstR = \frac{1}{p+q}$$

According to the definition of BurstR, when BurstR < 1, it means that the loss is sparse or scattered. When BurstR = 1, this means that loss is random and when BurstR > 1, loss is considered bursty.

An alternative to IETF RFC 3611 and Burst Ratio (BurstR) is to simply measure the Mean Burst Length (MBL) in RTP streams. A burst in this context is any consecutive loss sequence in the stream and the length of that burst is the number of packets lost in that sequence. It is noted that this definition of burst includes individually lost packets and the length in this case will equal 1.

The following is an example of an RTP loss profile that shows how the MBL can be calculated.

1 1 0 1 1 1 1 0 0 1 1 0 0 0 1 1 1 1 1 0 1

A 1 indicates a packet is received successfully and a 0 indicates a lost or discarded packet. In this example there are 4 bursts and their length is 1, 2, 3 and 1. In this case MBL is given by the following equation:

$$Mean\ Burst\ Length = \frac{\sum burst\ lengths}{number\ of\ bursts}$$

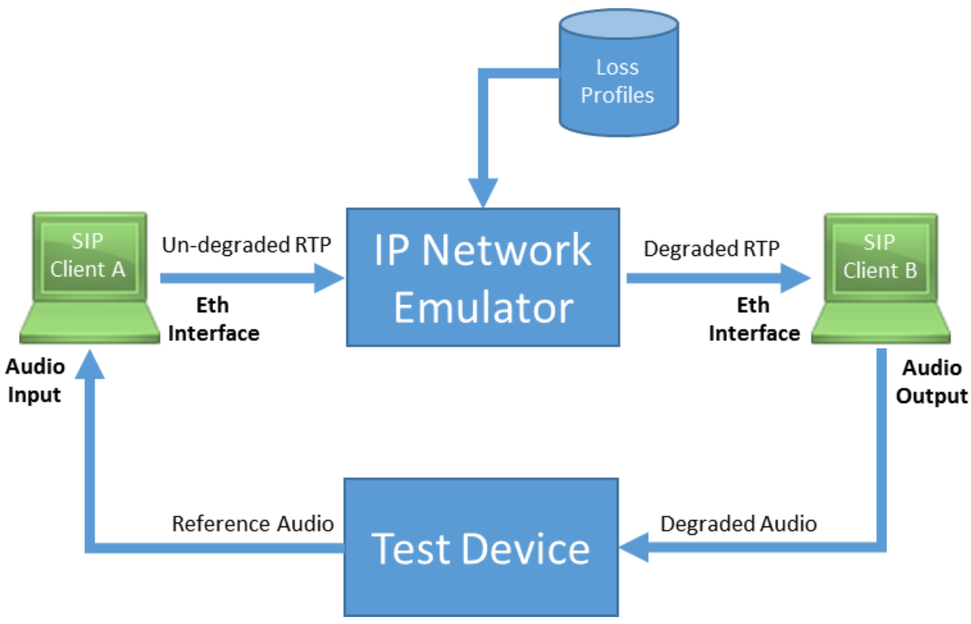
Hence, for the above example, the MBL is 1.75.

By measuring the packet loss on real-world networks, bursty profiles can be generated that are classified by their associated MBL and applied in an emulation environment allowing the impact on voice quality to be measured.

A3.3 Voice quality measurement configuration

The set-up shown in Figure A2 can be used to carry out voice quality measurements using a Full Reference algorithm such as ITU-T P.863 POLQA (Perceptual objective listening quality assessment) or ITU-T P.862 PESQ (Perceptual evaluation of speech quality), whilst degrading the RTP stream according to a specific burst profile.

Figure A2: Test configuration for measuring voice quality of a degraded RTP stream



A test call is established from SIP client A to SIP client B (for the codec under test). A test device is used to inject a reference speech signal into the audio interface of SIP client A. A network emulator is used to intercept the packet path between SIP client A and SIP client B and discard packets according to the loss profile under test. The degraded RTP stream is received at client B and the associated degraded audio received by the test device allowing the voice quality to be measured.

MOS-LQO measurements for a range of codecs based on this approach are incorporated in the ND1704 Performance Calculator (Annex IV).

Annex IV: ND1704 Performance Calculator (Normative)

Please see associated spreadsheet

History

Document history		
V1.1.1	March 2008	Approved by NICC E2E QoS WG
V1.1.2	September 2008	Editorial corrections approved by NICC TSG
V1.2.1	March 2009	Section 5.2.1 Rule PD5 amended as a result of a Change Request approved by NICC E2E QoS WG and TSG.
V1.2.2	May 2009	Editorial minor change.
V1.2.3	September 2011	Major changes to provide guidance (rather than rules) for network operators to dimension their networks to handle the expected delay, jitter and packet loss introduced by interconnected NGNs. Rules C1 and C2 changed to mandate support of both 10 ms and 20 ms packetisation times for the G.711 codec.
V1.2.4	October 2011	Approved by NICC E2E QoS WG ready for consensus approval
V2.1.1	November 2011	Approved following Consensus approval process and uplifted to ideal version numbering
V3.1.1	August 2018	Approved following Consensus Approval process containing the All IP TG updates
V3.1.2+ma	October 2020	Proving clarification for the Service Definitions and Rules in the following sections - Introduction; section 4.2 (Service Definitions); section 5.1 to 5.8 (Rules); Annex 1; Annex 3 - for TG approval
V3.1.4	April 2021	Comments with regards to the removal of the reference to ND1647 10 ms p-time requirement in Rule C2 since this is no longer mandated in ND1647 (V1.2.2).
V4.1.1	July 2021	Published with changes to - aligning the Foreword, Introduction and Scope; align with the Performance Calculator; changes to the Service Definitions and Rules in the following sections - Introduction; section 4.2 (Service Definitions); section 5.1 to 5.8 (Rules); Annex 1; Annex 3.
V4.2.1	May 2022	Published to include to the agreed change to rule C2 to specify the requirement for G.711 coding for default interoperability as required by ND1035 and ND1647.