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NICC Document

Guidance on CPE Compatibility on NGNs and NGAs

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Foreword

This NICC Document (ND) has been produced by NICC QoS Working Group

1 Scope

This document contains guidance on the potential issues associated with the operation and compatibility of Voice Band Data (VBD) Customer Premises Equipment (CPE) and Voice CPE on Next Generation Networks (NGNs) and Next Generation Access (NGA).

The document also includes guidance on CPE compatibility testing methodologies.

This guidance has been published by the NICC at the request of Ofcom.

The guidance contained in this document is targeted at two audiences:

- Guidance for CPE manufacturers and users, to assist in the design and operation of VBD and Voice CPE and associated applications that are as robust as possible in the presence of any transmission channel impairments, and to assist in appropriate compatibility testing of such CPE against NGN and NGA networks. Further, this document is intended to assist in the procurement of network access and services that meet the operational quality requirements of user VBD systems.
- Guidance for NGN and NGA designers and operators, to assist in the design and operation of NGNs and NGAs that provide transmission channels that are as benign as possible for the operation of VBD CPE and associated applications, and for the connection and use of Voice CPE.

Attention is drawn to the long term trends away from voiceband data and toward use of end to end IP based communication for data services.

2 Informative References

[1] NICC ND1701: "UK National Transmission Plan".
[2] NICC ND1704: "NGN Interconnect e2e Performance".

3 Definitions, symbols and abbreviations

3.1 Definitions

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

MSAN: A generic edge of network device. This may be exchange or cabinet based, depending on network topology

Next Generation Access: A carrier class network used in place of a traditional copper access network, typically carrying multiple services over a converged IP based architecture, with termination equipment in customers premises or street based cabinets.

Next Generation Network: A carrier class network used in place of a traditional switched network, typically carrying multiple services over a converged IP based architecture.

Non Linear Processor: The residual echo suppressor part of a G.168 compliant echo canceller.

Traditional Switched Network: A carrier class PSTN / ISDN network.

Analogue Telephony Adaptor: An electronic device that connects standard analogue telephones to digital telephone systems such as VoIP

3.2 Symbols

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

Bd: Baud; symbol rate
Hz: Hertz
ms: milliseconds

3.3 Abbreviations

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

AJB	Adaptive Jitter Buffer
ATM	Automatic Teller Machine
CLI	Calling Line Identity
CPE	Customer Premises Equipment
CPS	Carrier Pre-Select
DJB	De-Jitter Buffer
DTMF	Dual Tone Multi Frequency
EC	Echo Canceller
EPOS	Electronic Point Of Sale
FTTC	Fibre To The Cabinet
FTTP	Fibre To The Premises
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ITU	International Telecommunications Union
JB	Jitter Buffer
JBA	Jitter Buffer Adaptation
MSAN	Multi Service Access Node
NGA	Next Generation Access
NGN	Next Generation Network
NLP	Non Linear Processor
PDV	Packet Delay Variation
PLC	Packet Loss Concealment
PPM	Parts Per Million
PSTN	Public Switched Telephone Network
RTD	Round Trip Delay
SMS	Short Messaging Service
VAD	Voice Activity Detection
VBD	Voice Band Data

4 Introduction to NGNs and NGAs

NGNs (Next Generation Networks) should be distinguished from NGAs (Next Generation Access) such as FTTC (Fibre To The Cabinet) and FTTP (Fibre To The Premises). NGNs replace the traditional switched network, whereas NGAs replace part or all of the copper pairs in the access network. FTTP based NGAs and potentially FTTC based NGAs will typically be used to provide derived voice services alongside using customer premises sited Analogue Telephony Adapters (ATAs).

Most of the qualitative descriptions in this document relating to voice channel behaviour relate equally to both NGNs and NGAs. However it is more difficult to provide quantitative descriptions of NGA behaviour as can be done for NGNs, as NGA development is at an earlier stage. Moreover with both NGNs and NGAs, the quantitative behaviour of the various potential voice channel impairments can vary significantly from one design and implementation to another: not all NGNs and NGAs are necessarily created equal wrt to compatibility with VBD CPE and voice CPE. However it is reasonable to state that calls over NGA delivered derived voice services are likely to encounter similar NGN-like impairments to the audio transmission channel compared to calls on traditional switched networks, but potentially to a greater degree.

Traditional Public Switched Telephone Networks (PSTNs) are switched circuit networks. When a call is made, a dedicated data path is set up between the two end points which remains in place for the duration of the call. Once this end to end path has been established, then for each direction of transmission, the analogue audio signal received at one PSTN line card is converted into a continuous stream of audio data and sent to the other line card, for conversion back into an analogue audio signal. The audio codec used to convert from analogue audio signal to audio data, and vice versa, is typically the Recommendation ITU-T G.711 codec, which generates a continuous 64kbit/s data stream. It is this 64kbit/s audio data stream which is continuously transmitted in each direction across the dedicated point to point connection set up at the beginning of a call. The propagation delay experienced on any given call on a traditional PSTN should remain constant except when a transmission fault causes an in-call re-route of the transmission path.

NGNs and NGAs are packet based networks. There is no continuous point to point physical connection set up when a call is made. Instead, when a call is made, packets containing the audio data from one voice line card are sent to the other line card by the use of an appropriate address attached to each packet. Packets during a particular voice call on an NGN/NGA will typically be given the same route through the IP network. However, packets, even when on the same route, can be delayed by different times as they encounter different length queues at each network node, though in a well managed IP network, the delay variation should be relatively small. In an NGN/NGA, the audio codec used to convert from analogue audio signal to audio data, and vice versa, may or may not be the same Recommendation ITU-T G.711 codec typically used in traditional switched networks. The audio data stream produced by whichever codec is used, is packetized, sent across the IP network, reassembled at the far end voice line card, and converted back to an analogue audio signal.

There are a number of potential differences in the characteristics of the audio transmission channel, summarised below, which are explained in more detail in later sections.

Calls on NGNs and NGAs are likely to encounter:

- Increased end to end delay compared to traditional switched networks calls. See Section 7 for more detail on e2e delay.
- Audio channel discontinuities caused by jitter buffer adaptations, which do not occur on traditional switched network calls. See Section 8 for more detail on jitter buffer adaptations.
- Use of echo cancellation by default on all voice calls, which may often not be used on traditional switched network calls. See Section 9 for more detail on echo cancellation.

Calls on some NGNs and NGAs may also encounter:

- Use of a different voice codec than the Recommendation ITU-T G.711 codec typically used on traditional switched networks. Such codecs may have lower bitrate or different compression characteristics than G.711 codecs. See Section 10 for more detail on voice codecs.
- Use of DTMF relay, fax relay, or modem relay, which are typically not used on traditional switched networks. See Section 11 for more detail on relay techniques.
- Increased delay to dial tone, post dial delay and post answer delay during peak periods of network traffic. See Section 12 for more details of DDT, PDD, and PAD.

Additionally, CPE on NGNs and NGAs are likely to encounter:

- Reduced maximum off-hook loop current. See Section 13.1 for more details on reduced loop current.
- Balanced ringing rather than unbalanced ringing. See Section 13.2 for more details on balanced ringing.

5 Introduction to Voice Band Data CPE

5.1 Types of VBD CPE

Voice band data CPE are perhaps more widespread than often realised. Any CPE that makes a call and then transfers any sort of data to any sort of receiving CPE is a VBD application.

The following is an indicative list of CPE that use Voice Band Data (VBD) transmission over the PSTN network, with some indications of UK volumes (at time of writing), along with typical data transmission methods used in the UK:

- Satellite TV boxes (~9m). Consumers. V.34 and V.90.
- Fax machines (~2m). Businesses. V.17, V27ter, V29 and V.34
- Dial-up Internet access (~1m? falling). Consumers. V.34, V.90 and V.92.
- Security alarm panels (~1.5m). Buildings and homes. DTMF based protocols and V.34.

- Telecare social alarms (~1m). Local authorities and other telecare providers. DTMF based protocols.
- Lift Emergency Phones (~300k). Every lift. DTMF based protocols and V.34.
- EPOS card sale terminals. (~100-200k) Small to medium sized commercial outlets. V.22bis and V.34.
- Pre-payment card top-up machines. For gas, electricity and mobile pre-payment schemes. V.22bis and V.34.
- Fire alarms (~150k). Non-residential premises. DTMF based protocols and V.34.
- ATM cash machines (~55k). The sort typically sited in non-bank based locations. V.22bis and V.34.
- Telemetry remote monitoring (~50k). Utilities and the Environment Agency. V.22, V.22bis, V23 and V.34.
- SMS phones. Consumers. V23.
- Text Phones. For the hard of hearing. V.21, Baudot.
- Telehealth (eHealth) remote health monitoring . NHS and private health providers. V.34, V.90 and V.92.
- Payphone remote management. Payphone providers. V.34.
- PBX remote management. PBX manufacturers and providers. V.34.
- Custodial tagging devices. Probation services. V.34.
- CLI phones, display boxes and other CPE (~ some millions). Receives transmissions from line card only, not across network. Consumers and businesses. V23.

Most of the above categories of VBD CPE comprise a VBD CPE ‘outstation’ and a ‘central station’ or ‘receiving centre’ that the outstation communicates with, either by reporting from the outstation to the central station, or by polling from the central station to the outstation. A VBD CPE outstations is normally connected to an individual analogue voice line (generally on the legacy PSTN at present), while a VBD CPE central station may connected by one or more analogue voice lines, or may be connected via one or more ISDN30 connections. Individual voice lines will increasingly be subject to potential migration to IP based NGN or NGA voice lines, while ISDN30 connections will be increasingly subject to migration to IP-based SIP trunking products.

There is considerable inertia in many of the relevant industry segments to moving from VBD implementations to IP-based implementations, and in many of the above areas there are little or no IP-based replacements available to the VBD CPE items. Progress in the migration from VBD CPE to IP based CPE is likely to be a mixed picture.

5.2 Voice Band Data Transmission Method Characteristics

DTMF
Used by security alarm panels, fire alarms, telecare & social alarms, lift emergency phones, some text phones.

Security alarm panel and Fire Alarm protocols include Contact ID, Fast Format (these appear to be the most common two protocols) SIA1, SIA2 and SIA3.

Telecare & Social alarms protocols include BS8521 (new standard).

DTMF based protocols are generally half duplex.

V.21
Used by most text phones. 300bit/s. Full duplex.

V.22
Used by some telemetry equipment. 1200bit/s or 600bit/s Full duplex.

V.22bis
Used by all EPOS terminals, all pre-payment card top-up machines, all ATM machines, and some telemetry equipment. 2400bit/s. Full duplex.

V23
Used by most telemetry equipment. Also used by CLI phones, display boxes and other CLI CPE, and by SMS phones. 1200bit/s Half duplex.

V.34
Used by some dial up internet modems, some fax machines, older satellite boxes, telehealth equipment, and management functions in a wide range of CPE. 33.6kbit/s. Full duplex.

V.90 & V.92
Used by most dial up internet modems, newer satellite boxes and telehealth equipment. 56kbit/s downstream & 33.6kbit/s upstream (V.92 has upstream / downstream trade-off option). Full duplex.

5.3 VBD CPE Discrimination Tones

Voice band data CPE uses a variety of tones that can be recognised by NGN and NGA network edge devices as VBD discrimination tones. Of these, 2100Hz, with and without phase reversals, is the most common, but there are many more. However the effect of most of these VBD tones on jitter buffers and echo cancellers in NGNs and NGAs is not standardised. A fuller list of such tones, along with their likely or possible effect on jitter buffers and echo cancellers, can be found in Section 14.

6 Voice CPE

The range of Voice CPE available is huge, and includes items in many categories, including:

- Corded Phones
- Cordless Phones
    - Analogue Cordless Phones
    - Digital / DECT Cordless Phones
- Answering Machines
- CLI Phones
- CLI Display Boxes
- Audio Conferencing Equipment
- Autodiallers
    - Indirect Access Code Autodiallers
    - Hotline Autodiallers
- Call Barring Devices
- Distinctive Ringing Devices
- REN Boosters
- Extension Ringers
- Payphones
    - Private Payphones
    - Public Payphones
- Multiple Line Phones / Mini PBXs
- Call Loggers
- Test Equipment
- Network Monitoring and Alarm Equipment
- Line Isolating Units (not CPE; actually access network equipment used in electrical hotsites, but included here for completeness)

The scale of compatibility issues faced by Voice CPE when connected to NGNs and NGAs is likely to be relatively small, and most of the potential compatibility issues described in this document relate to Voice Band Data CPE. Nevertheless, there are some potential compatibility issues with NGNs/NGAs that do relate to Voice CPE, and these are described in Section 13.

7 End to End Delay

7.1 End to End Delay on NGNs and NGAs

End to end delay in this context refers to the delay between an electrical signal entering a voice line card at one end of a call, and the corresponding electrical signal exiting the voice line card at the other end of the call. A commonly used measurement is Round Trip Delay (RTD), which is the sum of the delays in each direction.

The RTD on calls over a single traditional UK wide switched network, for normal geographic number routing, is typically between a minimum of 4ms for calls originating and terminating on the same DLE, and a maximum of 50ms for a worst case routing.

On NGNs and NGAs the RTD will normally be higher than on switched networks for the same type of call.

The RTD on calls over a single UK wide NGN, for normal geographic number routing, will typically be between a minimum of at least 40ms for calls originating and terminating on the same edge located MSAN, and a maximum of at least 75ms for a worst case routing. The RTD on similar calls between a switched network and an NGN from the same network operator, will typically be between a minimum of at least 40ms and a maximum of at least 90ms. However the RTDs encountered on an NGN will depend on the specific architecture and design used and may exceed the values given here.

The RTD on calls carried over an NGA will depend on how the call is carried beyond the boundaries of the NGA. For calls carried across an NGA-NGN-NGA call route, the RTD values encountered are likely to be greater than the values described above for NGNs.

In order to understand the potential risk associated with operating a particular item of VBD CPE with a particular RTD point of failure, on an NGN/NGA, it is important to understand the minimum and maximum RTD values that can reasonably be expected to occur on the relevant NGN or NGA in question. This normally requires:

- A bottom up analysis of the minimum and maximum delays associated with each network element in a call route.
- This bottom up analysis to be done for all call route scenarios expected to be encountered, including all relevant combinations of NGA, NGN and TDM network routes
- This bottom up analysis to also be done for all relevant call route scenarios involving call routing or tromboning that may be caused by non-geographic numbering, number portability, carrier preselect or indirect access codes.
- Consideration of the effect of any VBD discrimination tones on the jitter buffers contained in the NGN MSANs, NGA ATAs and NGN/NGA Media Gateways.
- A top down measurement of real e2e delays on all relevant call route scenarios in order to confirm or other wise the limit values obtained from the bottom up analysis.

It is also important to understand that the round trip delay on a call can be greater than these figures for a number of other reasons which include the use of non-geographic numbering, number portability, carrier pre-select, indirect access codes, the use of VBD discrimination tones (see Section 14) and call routing across multiple operators' networks.

The NICC document ND1701 (UK National Transmission Plan) contains guidance on the maximum round trip delay that calls on fixed networks in the UK should encounter. A figure of 300ms is quoted as the recommended maximum in Issue 5 of ND1701, a figure which is currently under review by the NICC and may be revised upwards.

7.2 Minimising End to End Delay on NGNs and NGAs

End to end delay on NGNs and NGAs can be minimised by:

- Using a 10ms packet size for voice data, as normally required by ND1704 for NGNs interconnected by IP.
- Using adaptive jitter buffers (see Section 8), as required by ND1704 for NGNs interconnected by IP.
- An architecture that meets the 35ms limit defined in ND1701 for TDM connected NGNs, and the e2e delay limits for IP connected NGN defined in ND1704. Note that work is ongoing within NICC QoS WG wrt NGA delays.
- An architecture that provides decoding of non-geographic numbers, and avoids tromboning calls out to a TDM network for such decoding.

7.3 Impact of End to End Delay on VBD CPE

Any CPE that communicates data over a call to a receiving centre or other CPE will use some sort of communications protocol. Such protocols may typically have time-outs built into them, which will retry or hang up if some response or other is not received when expected. If such time-outs do not make adequate allowance for network round trip delay (RTD), then protocol time-outs can occur on higher round trip delay calls that would otherwise have been successful. Specifically, if a VBD CPE protocol time-out has been trimmed down to only just allow for the level of RTD typically encountered on traditional switched network, then when such CPE is migrated to an NGN/NGA voice line, it is at high risk of failing. The risk of failure is dependant on:

- the network round trip delay.
- the far end CPE response time.
- the maximum response time allowed by the protocol, i.e. the protocol time-out.

The increased round trip delay encountered in NGNs/NGAs compared to traditional switched networks is described in Section 7.1, and protocol time-outs clearly need to allow for this RTD. Often forgotten though is the need for the protocol time-out to also allow for the far end CPE response time, which is not necessarily a constant. For example the response time of security receiving centre equipment can vary from several tens of milliseconds to several hundreds of milliseconds, and a security alarm panel may work successfully on a given network RTD with one receiving centre equipment, but fail with a second slower responding receiving centre equipment. The same can be true of other categories of VBD CPE.

Even when the relevant protocol time-outs are sufficient to allow successful operation of a particular VBD CPE over a given network RTD and with a given far end CPE, if there is little safety margin, then this combination of VBD CPE is likely to be at increased risk of failure in the future if any aspect of the call routing changes. Such changes that may incur increased RTD on calls made include:

- One or both of the lines being moved from one operator to another, thus incurring additional network hops.
- The introduction of Carrier Pre-Select (CPS) on one or both of the lines, causing call to be carried by another operator's network.
- The introduction of Indirect Access codes on one or both of the lines, causing call to be carried by another operator's network.
- Changes made by any CPS or Indirect Access call re-seller to use spare call capacity on alternative network routes.
- The use of number portability.
- The introduction on a called line of a non-geographic number such as 0870 or 0845, which can introduce call routing 'tromboning' for the decode, due to the decode requiring routing either to a TDM network within the same operator's network, or a different network operator.

Additional network RTD may be incurred on NGNs or NGAs over which a call is routed, if any VBD discrimination tone recognised by any of those NGNs/NGAs is used (see Section 14), that triggers the jitter buffers on that call to transition from adaptive mode to a higher fixed mode length The resulting increase in the RTD on a call can be very significant. Note however that while this jitter buffer behaviour is currently mandated by ND1704, this requirement is currently under review. See Section 8.4 for more information.

7.4 Minimising Sensitivity of VBD CPE to End to End Delay

As indicated in Section 7.3 above, protocol time-outs used by VBD CPE need to adequately allow for both the maximum network RTD expected, and also the maximum far end CPE response time expected.

Maximum network RTD
At time of publication, the maximum network RTD expected on calls made over the UK network is defined by NICC document ND1701. This currently states (2006/03 Issue 5) that no calls made over fixed networks in the UK should exceed 300ms RTD. However it should be noted that:

- This figure is a recommendation, which although carries significant force, cannot be guaranteed to be met on particularly complex call routes. Examples exist already, that do not involve NGNs or NGAs, that exceed this figure.
- The maximum RTD may be subject to upward review. Users of this document should consult the latest available version of ND1701.

Maximum far end CPE response time
This can vary, even for a given VBD application, as indicated in Section 7.3 above. Depending on the application, the far end receiving CPE may be tightly defined and paired with the transmitting CPE, or not. And may be owned and operated by the same organisation, or not. If not tightly paired with each other, the operator or operators of each end may need to exercise care, and potentially co-operation, to ensure that CPE response times are understood and allowed for in the relevant protocol time-outs. Industry level co-operation, co-ordination and standardisation may be required.

8 Adaptive Jitter Buffers

8.1 Adaptive Jitter Buffers on NGNs and NGAs

In an NGN/NGA, any deviations from the regular arrival times of packets are referred to as packet jitter, or packet delay variation (PDV) and are dealt with by emptying the received audio data from each packet as it is received into one end of a jitter buffer (JB), sometimes referred to as a De-Jitter Buffer (DJB). The data is continuously streamed from the other end of the jitter buffer, thus smoothing the flow of data into the digital to analogue converter which reproduces the original analogue audio signal. Delays in the arrival times of arriving packets do not therefore cause any discontinuities in the audio data stream, or in the reconstructed analogue audio signal. As packets are not used in traditional switched networks, neither are jitter buffers.

Jitter buffers in NGNs and NGAs are normally required to support two objectives:

- Smoothing the flow of data in the presence of the prevailing level of jitter on a particular call, and
- Minimising the overall end to end delay on that call.

These two objectives potentially conflict in the sense that a larger jitter buffer length will cope with higher levels of jitter but lead to increased end to end delay, whilst a smaller jitter buffer length results in lower e2e delay, but may suffer overflow or underflow when maximum levels of jitter are encountered. To meet these two potentially conflicting objectives, jitter buffers in NGNs and NGAs are normally designed to be adaptive. Adaptive jitter buffers (AJBs) can if necessary dynamically increase in length when the level of prevailing jitter increases during a call, and then decrease back down in length when the level of prevailing jitter decreases during the same call. These changes in jitter buffer length are referred to as ‘jitter buffer adaptations’ (JBAs).

If a jitter buffer adapts upwards in length, ‘padding’ data (typically either silence or repeated data) is inserted into the regenerated 64kbit/s audio data stream prior to being converted back into an analogue audio signal, resulting in an unexpected gap in the reconstructed audio. If a jitter buffer adapts downwards in length, data is cut out of the regenerated 64kbit/s audio data stream prior to it being converted back into an analogue audio signal, resulting in an unexpected cut in the reconstructed audio. Jitter buffer control algorithms vary in their implementation and some may have the ability to wait until a silent period (if this occurs in the audio signal) before an adaptation is triggered, though this is less possible in the case of essential upward adaptations.

8.2 Adaptive Jitter Buffers : An Analogy

A useful analogy for any NGN/NGA or VoIP network, is a road system (=VoIP network), that has road junctions (= network nodes, or routers), where vans (= IP packets), each carry supplies (= audio data) from a starting place (= calling end) to a final destination (= called end).

When traffic is light, vans that are dispatched regularly (ie once every particular period of time) from the starting place will arrive at the final destination like clockwork (ie once every the same period of time). But when traffic builds up, vans may have to queue at junctions and will therefore start to arrive varying amounts of time later than their expected regular arrival time. The worst case van arrival time delay (from its expected arrival time) represents the level of packet jitter being experienced on the network.

The effect of these delays can be eliminated by unloading the supplies from each van into a supplies stockpile (= jitter buffer). Supplies can then be taken from this stockpile at absolutely regular intervals, even though the vans arrive somewhat irregularly. In this situation the size of the stockpile reduces in proportion to the worst case delay. Providing the worst case delay does not exceed a certain amount, the stockpile will not run out. When the level of traffic reduces, the arrival time delays disappear and the vans start arriving on time again, and the size of the stockpile recovers.

Adaptive jitter buffers can be likened in this analogy to a situation where traffic is particularly heavy, resulting in the worst case van delay exceeding the capability of the supplies stockpile to cope, and so the stockpile runs out. To stretch the analogy slightly, supplies still have to flow out of the stockpile, but they will be ‘empty’ (this equates to silence data being inserted into an upwardly adapting jitter buffer which has run out of incoming audio data). Then when the traffic level reduces back down to a low level, the delayed vans finally come streaming in on time, and the size of the stockpile becomes larger than it originally was (this equates to a jitter buffer that has adapted upwards in size, but after the level of jitter has reduced back down to a low value). Then once the arrival times of the incoming series of vans has remained regular for a long enough period of time to satisfy the managers of the stockpile (who equate to the jitter buffer control algorithm) then in order to reduce the length of time that incoming supplies have to sit in the stockpile, some of the supplies in the stockpile are thrown away (this equates to a cut in the audio data being made in a downwardly adapting jitter buffer which is unnecessarily long in length).

8.3 Implementing Adaptive Jitter Buffers on NGNs and NGAs

The steady state operation of adaptive jitter buffers does not introduce any discontinuities in the audio channel, but such discontinuities (insertions or cuts) are introduced when the jitter buffers adapt up or down. To minimise this type of impairment to the transmission channel, the number of jitter buffer adaptations should be minimised.

8.3.1 Mid-Call Adaptations

Jitter buffer adaptations can be avoided on calls made over any one NGN/NGA if the minimum and default jitter buffer lengths is configured sufficiently higher than the maximum level of jitter occurring on that NGN or NGA. This requires:

- The minimum and default jitter buffer lengths to be set to be equal.
- The minimum and default jitter buffer lengths to be set sufficiently higher than the maximum level of jitter expected on the NGN/NGA to provide enough 'headroom' to satisfy the jitter buffer control algorithm's safety margin requirement, and hence avoid any precautionary increases in jitter buffer length.

If the second condition above cannot be met due to the need to not exceed a particular end to end delay figure on an NGN/NGA, then careful choice of the minimum and default jitter buffer lengths can at least minimise the frequency of jitter buffer adaptations on calls made solely over that NGN/NGA.

It is unlikely however that the minimum and default jitter buffer lengths can be configured sufficiently high to avoid adaptations when encountering levels of jitter potentially occurring on calls routed over multiple operators' NGNs. The maximum level of incoming jitter to any NGN is in principle limited by NICC document ND1704, but having the minimum and default jitter buffer lengths set to be large enough to cope with this maximum level without requiring any jitter buffer adaptations would incur undesirably high end to end delays for all calls on that NGN.

8.3.2 Start of Call Adaptations

The behaviour of jitter buffers when encountering fax or dialup modems is currently mandated in NICC document ND1704, which currently states that jitter buffers should then transition upwards into a higher length of 42ms. This requirement is predicated on the assumption that if a VBD application is recognised as such by the NGN/NGA on a call, then the impact of a initial large upwards adaptation but no subsequent adaptations will on average be less than the impact of no initial upwards adaptation but the chance of one or more smaller upwards adaptation occurring later in the call. This is not a black and white judgement, and is currently being reviewed within NICC QoS WG, as CPE compatibility test results to date suggest that the potential advantages of fixing jitter buffers in this way may be, more often than not, outweighed by the detrimental effects of start of call upward adaptations and of resulting increased e2e delay. At the time of writing, a submission to the NICC QoS WG is being progressed that proposes changes to address these considerations.

8.3.3 Synchronisation

The loss of synchronisation at an MSAN will result in regular jitter buffer adaptations, driven by, and proportional to, the magnitude of the difference in clock speed between the incoming packet streams and the MSAN stand alone clock. Similarly, any ATA that is not synchronised to the core IP network will also see regular jitter buffer adaptations proportional to the magnitude of the difference in incoming packet stream clock speed and the ATA clock. For example, a difference in the clock speeds of the edge device clock and the network clock (and therefore a difference between the expected rate of incoming packets and the actual rate of incoming packets) of 50ppm, will produce a regular 5ms jitter buffer adaptation every 100 seconds.

For maximum compatibility with VBD CPE, NGA ATAs should be synchronised to the head end of the NGA network.

8.4 Impact of Jitter Buffer Adaptations on VBD CPE

8.4.1 Effect of Jitter Buffer Adaptations

As is described in more detail in Section 8.1, when a jitter buffer at a particular end of a call adapts upwards in length, any CPE at that end will see an unexpected inserted silent period occur in the reconstructed audio stream. While if that jitter buffer adapts downwards in length, the CPE at that end will see an unexpected cut in the audio data stream. These audio channel discontinuities can affect VBD CPE applications in two ways:

- The data being transmitted can be corrupted, resulting in missing data or errors.
- The CPE can lose timing synchronisation, which depending on the data transmission method may result in errors, and/or a re-train/re-synch, or a retransmission, or a dropped call.

Adaptations occurring during silent periods are less likely to disrupt voice band data signals, however this would not eliminate the effect of the re-timing of received VBD signals.

8.4.2 Frequency and Magnitude of Jitter Buffer Adaptations

The maximum frequency of jitter buffer adaptations on the transmission channel of calls carried across NGNs/NGAs is difficult to predict, and depends on how many networks the call is routed over, the level and distribution of jitter encountered on each network, the traffic loading profile over time, and the exact design and configuration of the adaptive jitter buffers used on each call endpoint.

Additional and regular upward or downward jitter buffer adaptations will occur if the ATA on an NGA is unsynchronised to the head end of the NGA network, proportional to the magnitude of the difference in clock speed between the incoming packet streams and an ATA stand alone clock.

The size of upward and downward adaptations of an adaptive jitter buffer on NGNs/NGAs is dependant on the exact jitter buffer design, but will typically be in the order of some milliseconds. One particular chipset available for example has a minimum upward adaptation step size of 5ms, and a minimum downward adaptation step size of 2.5ms, the asymmetry allowing more aggressive upward adaptations and more cautious downward adaptations.

Inserted silent periods and cuts in the audio data stream will never be less than the minimum adaptation length of a jitter buffer, but may be multiples of that length. The minimum adaptation length of a jitter buffer in the upwards direction may be different to that in the downwards direction.

8.4.3 When Jitter Buffer Adaptations Occur

CPE may encounter audio channel discontinuities caused by jitter buffer adaptations in the following circumstances:

Adaptations early in the call

- If the VBD CPE call is recognised as such by the network, then depending on the exact network implementation, a relatively large inserted silent period may be encountered by CPE at one or both ends of the call, either at the beginning of the call, or shortly after any VBD discrimination tone is transmitted by either end CPE and recognised by each of the network endpoints. This is caused by the jitter buffers at each end of the call potentially adapting upwards as they transition into fixed mode, as described in Section 8.3.2. Any such upward adaptations, and the resulting silent periods inserted, are likely to occur at approximately the same time, but will not actually be precisely synchronised, as each jitter buffer and its VBD discrimination tone detection system is autonomous at each end of a call. And it is possible either that one endpoint may detect a VBD discrimination tone, and the other endpoint does not, or that one endpoint jitter buffer is configured to transition to fixed mode on VBD detection, and the other endpoint jitter buffer is not. In either case a large inserted silent period might occur at one end only, and so in one direction of transmission only. The use and length of jitter buffers in interconnected NGNs/NGAs in fixed mode is currently defined in ND1704, though this is currently being reviewed. See Section 8.3.2 for more information.
- If the prevailing level of jitter at the start of a call is higher than the default/initial jitter length can cope with, then one or more initial upwards adaptations will occur, until the jitter buffer control algorithm decides that there is enough ‘headroom’, or safety margin, between the adapted jitter buffer length and the prevailing level of jitter.

Mid-call adaptations

- If the level of jitter during the call increases significantly during any part of the call, then one or more upwards adaptations may occur, until the jitter buffer control algorithm decides that there is enough ‘headroom’, or safety margin, between the adapted jitter buffer length and the increased level of jitter.
- If the level of jitter during the call decreases significantly during any part of the call, then one or more downwards adaptations may occur, until the jitter buffer control algorithm decides that the ‘headroom’, or safety margin, between the adapted jitter buffer length and the decreased level of jitter, is not excessive.

8.5 Minimising Sensitivity of VBD CPE to Jitter Buffer Adaptions

As indicated in Section 8.4.2, the size of jitter buffer adaptations, and therefore the cuts and inserted silent periods caused to the audio stream, are normally in the order of some milliseconds. Using the example given, a 5ms upward adaptation is a very long period compared to many commonly encountered symbol rates, equating to 100% of a 200bd symbol length, or 20% of a 40bd symbol length. Clearly this is more than enough to cause potential synchronisation problems in any synchronous transmission, and data corruption problems in both synchronous and asynchronous transmissions.

Protection against audio channel discontinuities caused by jitter buffer adaptations can be provided by the VBD CPE application using:

- transmission methods that can detect any loss of synchronisation, and trigger a re-synchronisation sequence
- transmission coding that incorporates redundancy and error correction
- transmission methods that incorporate error detection and retransmission requests
- transmission protocols that do not interpret gaps in transmission of several tens of ms (caused by upwards adaptation of jitter buffers into fixed mode) as a permanent failure in transmission, whether at the start of any handshaking negotiations, or during data transmission.
- Pre-emptive early transmission of 2100Hz tone, to cause any upwards adaptation of jitter buffers into fixed mode to occur at a point that does not corrupt any handshaking or data transmission.
- retry calling after any error serious enough to cause the call to be dropped

Whether or not there is active protection against errors caused by jitter buffer adaptations, it may be advantageous for any failed calls to be logged and alarmed by VBD application. This, particularly if the call log includes the CLI of the calling line, may enable remedial action to be taken, and is particularly important for life and limb critical VBD CPE applications. Pre volume deployment testing is recommended to identify the likelihood or otherwise of operational issues relating to jitter buffer adaptations. The guidance for such testing is at Clause 15 of this document.

9 Echo Cancellation

9.1 Echo Cancellation on NGNs and NGAs

Echo cancellation is used, where necessary, in order to avoid callers hearing unwanted echo from the far end of a call.

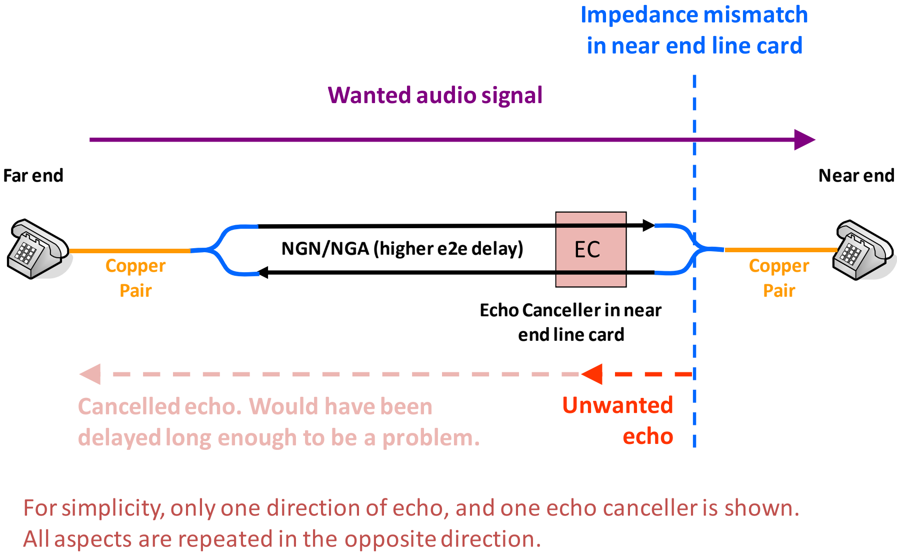
Echo cancellers used on both traditional switched networks and NGNs/NGAs will typically be built to meet the requirements of Recommendation ITU-T G.168. Such ECs comprise two separate parts:

- a linear echo canceller, which cancels most of the echo, and
- a Non Linear Processor, which suppresses the remaining low level residual echo.

Recommendation G.168 contains the statement ‘NLPs may affect the transmission of data through an enabled echo canceller’. This can occur when the beginning of a wanted return signal is clipped by an active echo-suppressing NLP prior to the NLP deactivating, triggered by the detection of the wanted signal.

Echo cancellation has been used for very many years on traditional switched networks for international calls, and in more recent years for some national calls. The potential for the signal clipping described above to occur exists on any call made over a switched network call that encounters echo cancellation.

Echo cancellation will typically be used by default on all voice calls on NGNs and NGAs due to the increased end to end delay encountered by such calls. There will therefore typically be an increased potential for the signal clipping describe above to occur on calls made over NGNs or NGAs



It is important to note that echo cancellers are contained in the NGN/NGA edge device (eg MSAN or ATA) but also in the Media Gateway that interconnects to TDM based legacy networks. The designs and operation of these two echo cancellers may well be different.

The operation of echo cancellers on calls can be controlled by the use of 2100Hz and other VBD discrimination tones (see Section 14).

9.2 Optimising Echo Canceller Implementation on NGNs and NGAs

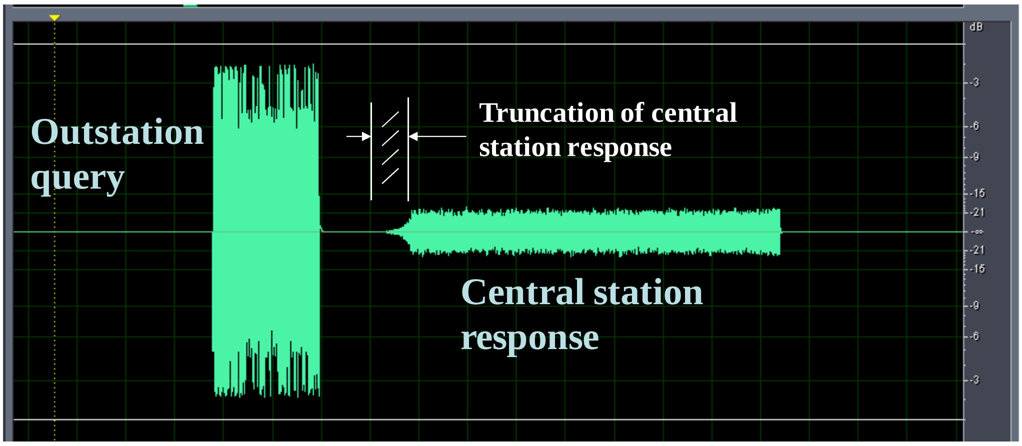
The implementation of echo cancellers in digital networks is described in the following document: G.168 ‘Digital Network Echo Cancellers’, which should be adhered to when implementing any echo canceller used in both traditional switched networks and NGNs/NGAs. One piece of text contained in Appendix I ‘Guidance for Application of Echo Cancellers’ is particularly relevant and is therefore quoted here:

> “PSTN network planners were expected to continue to evolve the network in such a way that it would not knowingly prevent the continued carriage of a permissive voice band data/facsimile service.”

Annex B of G.168 contains a description of a reference NLP. A parameter that is of particular relevance to the potential signal clipping referred to in Section 9.1 is the ‘NLP Operate’ time for the Z/W transition 3, as defined in Figure 34, which has a recommended maximum of 5ms. This parameter describes how quickly the NLP should go from active (ie cancelling an unwanted residual echo) to inactive (ie allowing a wanted signal through) after the onset of a wanted signal at the Sin input of the echo canceller. An NLP that has an operate time for this Z/W transition of significantly greater than this recommended maximum is likely to have a significantly increased potential to clip data signals.

9.3 Impact of Echo Cancellation on VBD CPE

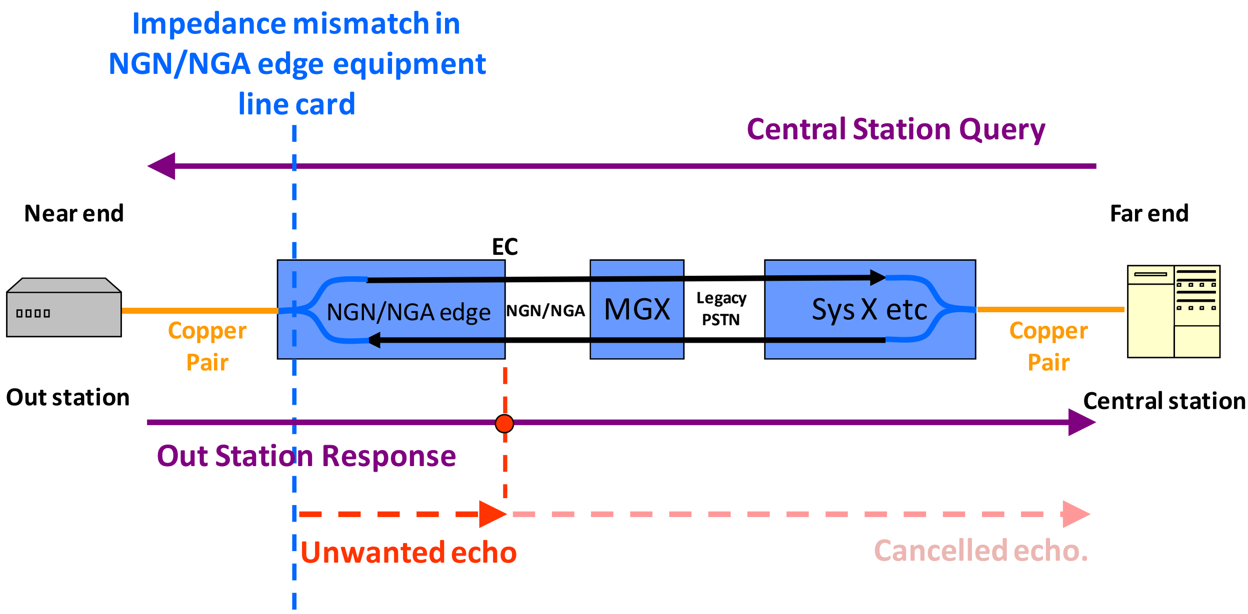
G.168 states 'NLPs may affect the transmission of data through an enabled echo canceller'. This refers to the possibility of the 'clipping' of an isolated data transmission burst that can occur with some echo cancellers under certain conditions. This has the potential to affect the operation of some voice band data CPE. An example of a VBD transmission truncated by a relatively slow NLP as described in Section 9.2 is shown in the following audio trace:



Clipping can in principle occur in two directions, as described in the next two sections.

9.3.1 Potential impact of MSAN/ATA echo canceller operation

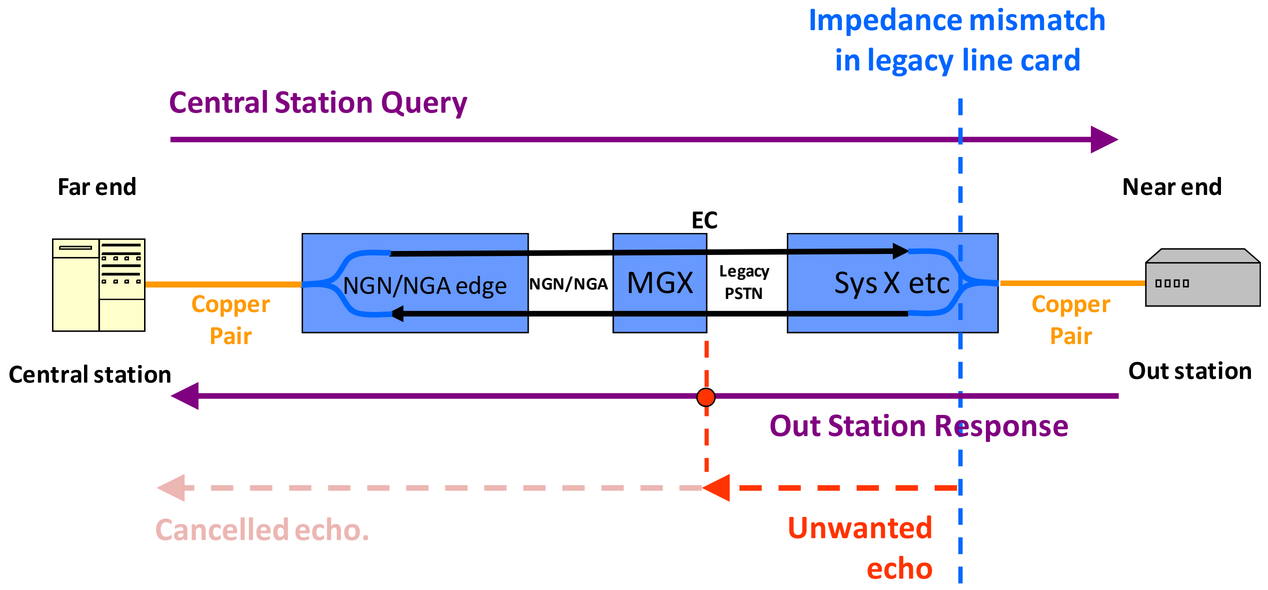
The echo canceller contained in the NGN/NGA edge equipment can potentially truncate a very fast response from a CPE connected to the edge equipment line card. For example, a fast responding EPOS terminal or telemetry out station connected to an NGN/NGA line card, could have its response truncated by the NLP on that line card. (The central station may be on either legacy PSTN or an NGN/NGA.) This is shown in the following diagram:



In the above diagram, the unwanted echo from the NGN/NGA edge equipment line card is cancelled by the echo canceller in that line card. A fast responding out station CPE at the NGN/NGA end may have its response clipped by the NLP operation in the MSAN / ATA echo canceller. Only one direction of echo is shown.

9.3.2 Potential impact of Media Gateway echo canceller operation

The Media Gateway echo canceller can potentially truncate a very fast response from a CPE item connected to a legacy PSTN line card. For example, a fast responding EPOS terminal or telemetry out station connected to an legacy PSTN line card, has its response truncated by the NLP on the relevant Media Gateway trunk card. (The central station is on an NGN/NGA.) This is shown in the following diagram:



The positions of the out station and central station are now reversed from that described in Section 9.3.1. In the above diagram, the unwanted echo from the legacy line card is cancelled by the echo canceller in the Media Gateway. A fast responding out station CPE at the legacy PSTN end may have its response clipped by the NLP operation in the Media Gateway echo canceller. Only one direction of echo is shown.

9.3.3 Confirming sensitivity to echo cancellor operation

If a failure of a VBD CPE occurs on an NGN/NGA line in either of the above two configurations described above, the capture of audio traces at the far end and near end two-wire points will help to confirm whether the failure mechanism is sensitivity to echo canceller operation: If the echo canceller can be configured to be permanently disabled at either the NGN/NGA edge equipment, or the Media Gateway, then this will also help confirm the cause of failure.

VBD CPE needs to be tested for sensitivity for NLP clipping or any other aspect of echo canceller operation by making test calls on an actual NGN/NGA line, as this is the only way that the effect of the actual NGN/NGA echo canceller will be seen. See Section 15 for guidance on CPE testing.

9.4 Minimising Sensitivity of VBD CPE to Echo Canceller Operation

Guidance relevant to the use of CPE in the presence of echo cancellers in the network is contained in the following document: G.168 'Digital Network Echo Cancellers'. The relevant guidance is contained in Appendix I 'Guidance for Application of Echo Cancellers', specifically the following two sections:

- Section I.2.3 'Responsibilities of modem manufacturers and end users'
- Section I.4 'Effect of cancellers on voice and data services'.

Certain parts of the text are particularly relevant and are therefore quoted here:

"It is the responsibility of the modem manufacturers and end users to understand the characteristics of the network-based echo canceller fully and decide whether the echo cancellers should be enabled or disabled. If the modem manufacturers and end users decide that the network-based echo canceller functionality should be disabled, they should ensure that the terminal uses the appropriate approved methods, defined in Recommendations, to disable cancellers."

G.168 makes repeated reference to G.161 'Interaction Aspects of Signal Processing Network Equipment', and Section I.4 mentioned above specifically references G.161. Particularly relevant sections include:

- Section 5.2.1 'Interaction of echo cancellers with facsimile transmission'
- Section 5.2.2 'Interaction of echo cancellers with modems'

Certain parts of the text are particularly relevant and are therefore quoted here:

"The V.27 ter and V.17 modulation scheme employed by ITU-T Rec. T.30 are protected against the mutilation of the training sequence by echo suppressors (by using an unmodulated carrier prior to the training signal)."

"Most modem manufacturers feel that network echo cancellers should be disabled for modems with integrated echo cancellers (e.g., ITU-T Recs V.32 and V.34), because an active network echo canceller operating in conjunction with the integral echo canceller in the modem may cause undesirable phenomena…"

"In the early 1980s, data showed that some echo cancellers did improve the operation (i.e., reduce or eliminate bit errors) for low-speed modems designed according to ITU-T Recs V.21, V.23, V.26 (alternative B), V.27 ter and V.29. Therefore, it was accepted that these modems benefited from an active echo canceller and a disabled echo suppressor."

"Data have indicated that certain combinations of modems with different protocols and modulation schemes (V.18, V.21 and V.23, DTMF and some V.34 implementations) and some echo cancellers, in various simulated network configurations and in the network, exhibit degraded performance when the echo cancellers are enabled. Therefore it is strongly recommended to use the 2100 Hz Answer Tone with phase reversals as mandatory in any modem and protocol Recommendation where the presence of a network echo canceller is likely to affect the performance of the modem transmission."

The operation of VBD CPE may be able to be protected against NLP clipping by one or more of the following:

- The use of 2100Hz tone to disable the NLP immediately prior to the start of full duplex data transmission, or at the start of each data burst in half duplex transmission. More detail of the effect of VBD discrimination tones can be found in Section 14. Note however, that NLP disabling on detection of 2100Hz tone is optional in the relevant standard (G.168) and will depend on the exact configuration of the relevant NGN or NGA.
- Use of a Full Duplex transmission modulation, as once disabled, the NLP and/or echo canceller will remain disabled, in contrast to Half Duplex, which is likely to allow the NLP and/or echo canceller to re-enable following any period of 100ms-400ms of bidirectional silence (see Section 14 for more information).
- The use of dummy data bits prior to the start of actual data transmissions in a full duplex system, or at the start of each data burst in a half duplex system.

10 Voice Codecs

10.1 Voice Codecs on NGNs and NGAs

On any digital telephone network, an audio codec is used on the PSTN line card to convert the analogue audio signal to a stream of audio data in the transmitting direction, and vice versa on the receiving direction.

On traditional switched networks, the codec used is typically one meeting the requirements of Recommendation ITU-T G.711 codec, which produces a 64kbit/s stream of audio data. There are three options in G.711 which are of relevance:

- Whether the ‘A-law’ or ‘u-law’ method is used to improve the handling of the wide dynamic range found in speech. The u-Law method is used North America and Japan, while A-law is used in Europe and the rest of the world.
- Whether packet loss concealment technique is used (G.711 Appendix 1). This is a method of masking the effect of any lost packets by replacing the missing data with data extrapolated from data received immediately previously. This is not applicable to traditional switched networks, which do not use packet transmission.
- Whether Voice Activity Detection technique is used (G.711 Appendix 2). This is a method of saving bandwidth in packet based networks which involves detection of silent periods, and transmitting silence length indicators to the far end, which then inserts comfort noise back into the audio path. This is not applicable to traditional switched networks, which do not use packet transmission.

In European traditional switched networks, the A-law method of handling audio dynamic range is used. Neither packet loss concealment nor voice activity detection are applicable to traditional switched networks.

NGNs and NGAs may typically use the same G.711 codec used in traditional switched networks, but other codecs may be used in place of G.711. If an NGN/NGA uses a G.711 codec, then in

Europe A-law will be used, as in a switched network. Packet loss concealment may or may be used; G.711 standard PLC adds 3.75ms of delay and so may not be preferred, although other methods of PLC exist that add zero delay. Voice activity detection may or may not be used.

10.2 Choice of Voice Codecs on NGNs and NGAs

If the design aim of an NGN or NGA is to offer a quality of transmission channel as close as possible to a traditional switched network, then the G.711 codec or better should be used, with appropriate choice of u-Law or A-law, and with packet loss concealment enabled, but with voice activity detection disabled.

The use of G.711 codec is mandated for NGN interconnect by NICC document ND1704, and the use of the same G.711 codec internally on an NGN will avoid the need for any transcoding at interconnect points.

Using alternative lower bitrate codecs will results in reduced transparency of the transmission channel compared to traditional switched networks, and an increasing risk of CPE incompatibilities. Tandeming of different codecs, and transcoding between them, should also be avoided if possible. If lower bitrate codecs are used as default, then the use of codec upspeeding on calls detecting as VBD should be considered.

Where an NGN/NGA does not use a G.711 codec,. VBD device users must satisfy themselves as to the appropriateness of services that incorporate non G.711 codecs in their end to end voiceband connection. Use of wideband codecs to provide improved speech are also possible, but they will lead to substantially increased risk of degredation of voiceband data.

10.3 Impact of Voice Codecs on VBD CPE

The types of audio channel distortions caused by the use the different types of audio codecs available is a large and complex subject. If the same G.711 audio codec is used on an NGN/NGA as is used on traditional switched networks, then VBD CPE designed for use on traditional switched networks should encounter no difference in the quality of the audio channel wrt audio coding (echo canceller operation and jitter buffer adaptations aside).

If lower bitrate codecs are used, then problems may occur, and increasingly so with higher compression codecs and higher bit rate VBD CPE applications. If the transmission quality of an audio channel is limited by a lower bit rate codec than G7.11, then some modem standards might be able to negotiate a lower and error free transmission rate, but many types of VBD CPE using modem standards such as V.22bis or V23 may be unable to negotiate a lower rate, and may therefore fail. Some very low bit rate audio codecs are not even able to support the transmission of DTMF sufficiently for accurate digit recognition (for example in mobile networks, DTMF relay is used to transmit DTMF digits) however such codecs are less likely to be encountered on typical NGNs/NGAs.

10.4 Minimising Sensitivity of VBD to Voice Codec Type

The aspects of a VBD CPE application that may give protection against audio channel discontinuities caused by jitter buffer adaptations listed in Section 8.5 will not necessarily give any protection against the use of lower bit rate codecs, as such codecs cause audio distortions that are constantly present in the audio channel, although such features may help if the quality of the audio channel is borderline for a particular VBD CPE application.

VBD CPE applications may be more likely to successfully operate across an audio channel using a lower bit rate codec if they:

- Use lower symbol rate transmission methods rather than higher.
- Use modem standards that can gracefully degrade by virtue of error detection and subsequent rate re-negotiation to a lower symbol transmission rate.
- Use DTMF rather than modem transmission standards.
- Use 2100Hz VBD discrimination tone in order to trigger potential codec up-speeding in the NGN/NGA network.

11 DTMF Relay, Fax Relay and Modem Relay

11.1 DTMF Relay, Fax Relay and Modem Relay on NGNs and NGAs

DTMF Relay, Fax Relay and Modem Relay are all methods of transmitting voice band data signals across a network via network messaging protocols rather than in-band. In each case, the in-band signal is detected, identified and terminated, then transmitted across the network using a dedicated message protocol, and the voice band data signal regenerated as an in-band signal at the far edge. An example of this type of out of band DTMF transmission is on mobile networks, where keypad presses are transmitted by the handset across the mobile network not as in-band Dual Tone Multi Frequency signals, but as ‘User pressed key 1’ and ‘User pressed key 2’ messages etc, with (for a mobile call to the PSTN) the in-band DTMF then regenerated at the far edge of the mobile network at the mobile-PSTN gateway. In a similar way, a VoIP network can be equipped to recognise and terminate DTMF, fax or modem signals at the edge of the network, transmit the underlying data across the network in IP packets, and regenerate the in-band DTMF, fax or modem signals at the far edge of the VoIP network. Each of these relay techniques can be implemented individually, together, or not at all.

11.2 Implementing DTMF Relay, Fax Relay and Modem Relay on NGNs and NGAs

If the design aim of an NGN or NGA is to offer a quality and transparency of transmission channel as close as possible to a traditional switched network, then the use of any relay technique probably needs to be avoided. However, an NGN/NGA that minimises bandwidth by the use of lower bit rate codecs may need to employ modem relay, fax relay or in the extreme even DTMF relay to ensure successful carriage of such voice band data traffic, depending on the compression used in the lower bit rate codec chosen. Implementations of such relay techniques can be complex though, and, and vary in the number of modem, fax or other protocol standards that are supported. This can increase the risk of protocol mis-recognition or mis-termination, and result in an increased risk of CPE incompatibilities.

11.2.1 Use of DTMF Relay

DTMF relay may be used if lower bit rate codecs are used. This is already the case for mobile networks, and may be encountered on some NGNs/NGAs. If implemented, then the scope of signals to be recognised, is of course tightly defined, and so problems with digit recognition at the near edge and digit regeneration at the far edge are perhaps less likely. DTMF relay is standardised in RF4733 (previously by RFC2833).

11.2.2 Use of Fax Relay

Fax relay is most commonly implemented according to the T.38 standard, which standardises the carriage of standard T.30 fax across an IP network. For example, a T.30 fax machine on a VoIP voice line on a network equipped with T.38 dials the far end fax number as normal, but the T.30 fax call is terminated on the near edge of the network, and the 14,400kbit/s (or lower rate) fax data is extracted and transmitted across the network in IP packets in a T.38 compliant way. At the far edge, the full T.30 fax signals are regenerated and transmitted to the far end fax machine.

11.2.3 Use of Modem Relay

Modem relay is implemented in a similar manner to fax relay, and is standardised by V.150, V.151 and V.152. The most likely to be encountered is probably support for V.34 modem standard, which may also support a V.90/92 modem that trains down to the lower V.34 rate. Support on modem relay gateways for older lower rate modem standards such as V.22, V.22bis and V23, is less likely. Support for modems can also be provided by ‘modem pass-through’ which is when a network recognises that a call is a modem call of some sort, and switches to the use of a full rate G.711 voice codec, and transmits the modem tones inband.

11.3 Impact of Relay Techniques on VBD CPE

11.3.1 Impact of DTMF Relay

Given the tightly defined and limited scope DTMF tones, problems with DTMF relay should not be encountered frequently. However, DTMF relay problems are sometimes encountered if:

- the tolerance on the ‘twist’ of DTMF tones (the ratio of volumes of the two tones used for any digit) are so tight on the digit detection side that DTMF tones from some slightly out of tolerance CPE, or edge of tolerance CPE, can fail to be properly detected.
- information is transmitted by using differing DTMF tone spacings or differing DTMF tone lengths, which may not be accurately transcribed by the DTMF relay function.

11.3.2 Impact of Fax Relay

T.30 fax and T.38 fax relay are both complex standards, and problems with faxes not communicating successfully with T.38 gateways can be encountered. Additionally, other fax handling methods can also be encountered, including proprietary fax relay protocols, fax store and forward, and fax ‘upspeeding’. A full description of possible problems is outside the scope of this guidance document, and significant external information is available on this subject.

11.3.3 Impact of Modem Relay

Implementation of modem relay techniques is again a complex area, and potential for incompatibilities with modem CPE is significant. If an NGN/NGA does employ lower bit rate codecs, then support for standard V.34 and trained down V.90/92 modems, is most likely to be available, with support for other modem standard rather less likely. Support for other modem types would then be most likely by Modem Pass-Through, as described in Section 11.2.3, however problems can still be encountered such as failure to detect a particular modem type, or slow detection causing problems with a particular modem protocol.

11.4 Maximising Compatibility of VBD with Relay Techniques

The compatibility of VBD CPE with relay techniques can be maximised by, if possible, the use of:

- fully compliant DTMF tone generation within DTMF CPE; relevant standards being ETSI 201 235 and ITU Q.23.
- fully standard T.30 fax devices.
- modern modem standards such as V.34 and V.90/92.

12 Delay to Dial Tone, Post Dial Delay and Post Answer Delay

12.1 Delay to Dial Tone, Post Dial Delay and Post Answer Delay on NGNs and NGAs

12.1.1 Delay to Dial Tone on NGNs and NGAs

Delay to dial tone is the time between a line going off-hook, i.e. a loop being placed on a line, and dial tone being applied onto the line, and is typically some fraction of a second on traditional switched networks. This delay is primarily affected by the operation and loading of the call server that is handling the call signalling for MSANs on NGNs. In NGAs, dial tone will typically be applied locally by SIP based ATAs.

12.1.2 Post Dial Delay on NGNs and NGAs

Post dial delay is the time between the last dialled digit and the application of ringing tone on the line. This delay is the sum of the relevant call signalling times between the two end points of a call, which may involve multiple operators' networks. This delay can be up to some number of seconds on traditional switched networks, depending on the number of network hops involved.

12.1.3 Post Answer Delay on NGNs and NGAs

Post answer delay is the time between a called line going off hook, i.e. a loop being applied to a line, and the opening of both directions of transmission on the line between the two endpoints. This delay primarily comprises the sum of the time to validate the off hook condition on the called line

plus the response time of the call server that is handling the call signalling for the relevant MSAN / ATA. This delay typically totals some fraction of a second on traditional switched networks.
All of these delays are affected to one degree or another by the handling time of the call server, which may result in higher delays when the call server is very heavily loaded, depending on the design and dimensioning of the NGN/NGA network architecture.

12.2 Minimising Delay to Dial Tone, Post Dial Delay and Post Answer Delay on NGNs and NGAs

Of these delays, it is probably the delay to dial tone on NGNs that has the most potential to impact the operation of automated CPE. In NGAs, dial tone will typically be applied locally by SIP based ATAs.

To match the performance of traditional switched networks in respect of these delays, the following design aspects need to be considered;

- The performance of the NGN MSAN PSTN line card or NGA ATA telephony interface, which needs be able to match those of line cards used in traditional switched networks.
- The design and dimensioning of the relevant NGN/NGA call server, which needs particularly careful consideration to avoid undesirable increase in one or more of these delays.

12.3 Impact of Delay to Dial Tone, Post Dial Delay and Post Answer Delay on VBD CPE

In general, testing for sensitivity to the level of delays that occur under lightly loaded network conditions is straightforward, -simply test the VBD CPE in question on the NGN/NGA network in the general manner described in Section 15.3. Testing for sensitivity to any potential increases in these delays that may occur on a heavily loaded NGN/NGA is more problematic, as this condition is unlikely to be available on a test network, although it may be available on an existing large scale deployment, if this existed. A practical alternative may be to request the likely maximum values of these delays from the network operator, and to simulate this behaviour on a suitable test rig.

12.3.1 Impact of Delay to Dial Tone on VBD CPE

The effect of any significant increase in delay to dial tone will depend on whether an item of VBD CPE waits for dial tone before proceeding, or whether it simply goes off hook and assumes dial tone will be present after a short fixed period. In the latter case, an increase in this delay may cause a VBD CPE call to fail. Subsequent retries may also fail if the longer delay to dial tone persists. In the former case, whether any problem will occur depends on how long the time-out may be on the wait-for-dial-tone state. A time-out of a few seconds or more is very likely to be sufficient for any normal distribution of delays to dial tone, barring network fault conditions.

12.3.2 Impact of Post Dial Delay on VBD CPE

Given that post dial delays already occur in legacy networks, it is probably unlikely that VBD CPE designed to cope with existing delays would have a problem on any NGN/NGA routed calls. The likelihood of any problems occurring will be dependent on the time-out on the waiting-for-answer-tone state or similar, that is built into the CPE in question.

12.3.3 Impact of Post Answer Delay on VBD CPE

Increased post answer delay may potentially cause clipping of any transmission from a VBD CPE that answers an incoming call and responds particularly fast, if the transmission channel is not opened quickly enough. Whether this causes a problem, even if it occurs, will depend on the nature and duration of the initial transmission.

12.4 Minimising sensitivity to increases in DDT, PDD & PAD on VBD CPE

Connection related time-outs of the sort described in Section 12.3 is primarily avoided by the use of:

- Appropriate dial tone detection before dialling out
- Sufficiently ‘patient’ time-outs in each call state.

13 Other Potential Compatibility Issues

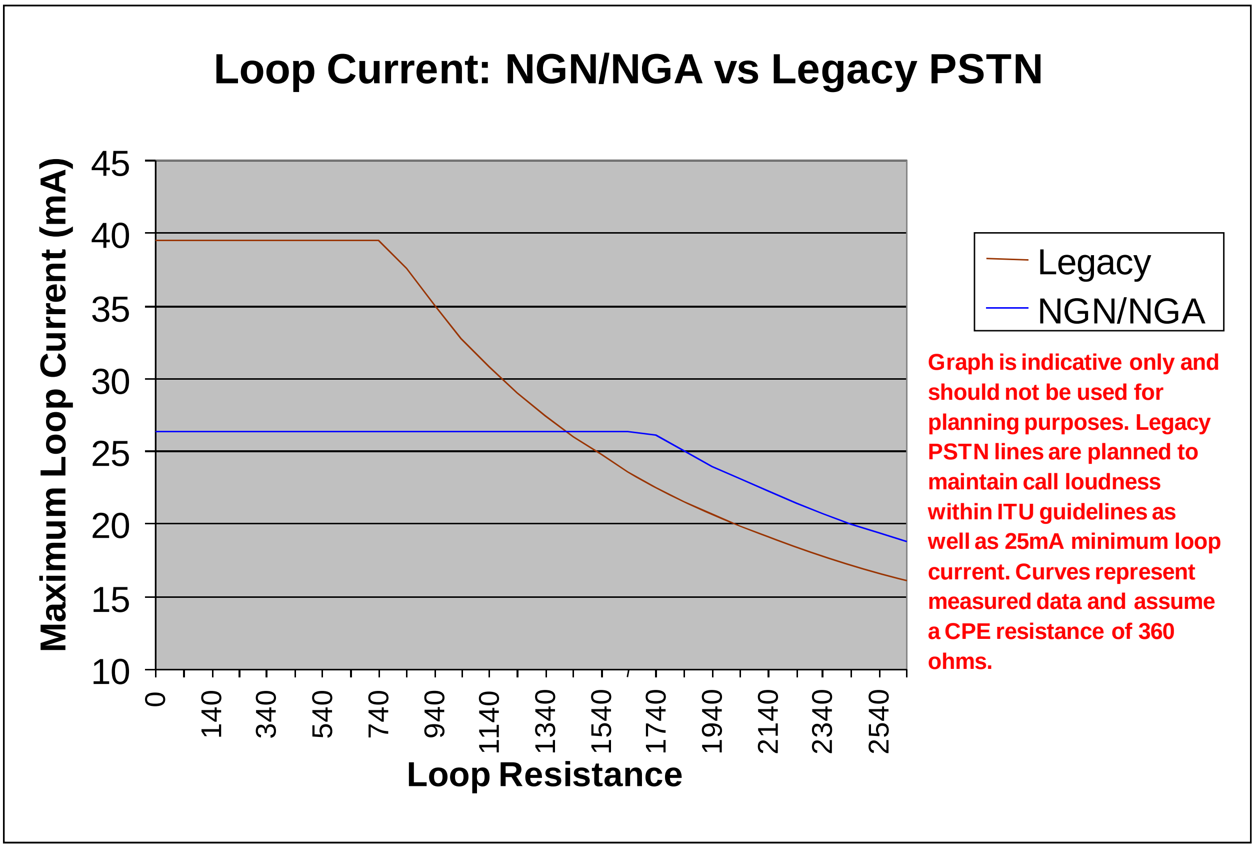
This section describes a number of additional potential compatibility issues that may affect both Voice Band Data CPE and Voice CPE.

13.1 Reduced Maximum Loop Current

Loop current is the current that flows when a CPE goes off hook, and is determined by the maximum voltage available from the exchange line card, the current control implemented on the exchange line card, the loop resistance of the copper line, and the DC characteristics of the CPE when off hook. UK legacy PSTN (System X and AXE 10) exchange line cards have constant current controls implemented which aim to deliver approximately 40mA constant current. On lines with low loop resistance, only a relatively low exchange voltage is required to deliver this. Lines with higher loop resistance required correspondingly higher exchange voltage, until the upper limit on the exchange voltage is reached. At this point, lines with still higher loop resistance will receive lower than the 40mA maximum, on a I=V/R basis. Given the exchange voltage available, and the range of loop resistance encountered in the access network, all lines will receive a minimum of at least 25mA loop current.

The maximum loop current available on NGNs and NGAs may replicate the characteristics of the UK legacy PSTN, but more typically will be limited to a lower figure of 25mA, as this is consistent with network design trends in Europe and elsewhere, which allows reduced exchange power consumption and air conditioning load, and improves line card packing density. Compared with voice lines on legacy PSTN, voice lines on NGNs/NGAs may therefore still provide the 25mA minimum, but may not provide a higher 40mA maximum.

The difference between the loop current available on legacy PSTN, and the likely loop current available on NGNs and NGAs, is shown in the following graph. It should be noted that this is not a difference that affects longer lines with higher loop resistance, but one that affects shorter lines with lower loop resistance.



The impact of this reduced maximum loop current on CPE is very low, as the vast majority of VBD and voice CPE should be, and is, designed to work with 25mA. Any CPE unable to operate correctly with 25mA loop current would only be found on shorter / lower loop resistance lines on the existing legacy network, as they would fail to operate on longer legacy lines. Testing compatibility for this characteristic can be undertaken on the NGN/NGA voice line itself, or on longer legacy voice lines of sufficiently high loop resistance such that only 25mA loop current is available, which can be easily checked with a simple multimeter.

13.2 Balanced Ringing

Most of the existing UK legacy PSTN uses ‘unbalanced ringing’, where AC ringing voltage is applied to one wire of the copper pair with respect to earth, but not the other wire. Voice lines on NGNs/NGAs may provide unbalanced ringing, but are more likely to provide balanced ringing, which applies the AC ringing voltage across the pair, with half the voltage applied across each of the wires, but out of phase. Thus with balanced ringing the AC voltage across the pair is the same as that for unbalanced ringing, but the voltage with respect to earth is less. This form of balance ringing is in fact already used in parts of the UK legacy PSTN, and is in widespread use in many parts of the world, as it simplifies exchange line card design, and because it is also more ‘broadband friendly’, as it generates less impulse noise.

The impact of the use of balanced ringing is low, as the very large majority of CPE has been designed to detect both forms of ringing. Any CPE that has been designed to only detect unbalanced ringing, will not ring, and/or not detect the incoming ringing signal. Testing for this characteristic is best done on the relevant NGN/NGA voice line itself, although a balanced ringing generator can be used as a portable alternative, providing that the generator exactly emulates both the AC ringing characteristics and the DC bias characteristics of the ringing implementation of the relevant NGN/NGA voice line.

13.3 On-Hook Voltage

The on-hook voltage of voice lines on the UK legacy PSTN network is typically approximately 50V, although a maximum of 70V is specified. Some CPE exists that incorporate over voltage detectors, that trigger substantially below 70V. Any NGN/NGA voice line that provide an on-hook voltage of significantly greater than 50V may trigger such CPE over voltage alarms.

13.4 CLI Delivery Performance

The successful operation of Calling Line Identity (CLI) on voice lines is dependant on both the network implementation of CLI being correct, and also of the relevant CPE being fully conformant with the relevant terminal equipment specifications. In reality however, there are many voice CPE in the installed base that are not fully conformant to the relevant terminal specifications, and this can have a significant impact on the CPE in question, depending on how well engineered the CLI transmission is at the network end. Performance degradation under these circumstances is not necessarily line length dependant, and so it is possible that CLI performance issues may be evident on either customer sited or street sited NGA equipment.

14 2100Hz and Other VBD Discrimination Tones

14.1 The effect of VBD discrimination tones on NGNs and NGAs

Various VBD CPE discrimination tones may be recognised by NGNs and NGAs as valid triggers for:

- Disabling the NLP component of the echo canceller, particularly if this is required to avoid data transmissions being clipped by the NLP. See Section 14.2.
- Fully disabling all echo canceller functionality i.e. the NLP component of the echo canceller and the linear echo canceller itself. This is done for high speed modems (V.32 and above) which contain their own built-in echo cancellers which might conflict with any network echo cancellers left enabled. See Section 14.3.
- Triggering adaptive jitter buffers to transition to fixed mode. See Section 14.4.
- Upspeeding the codec used to G.711 from a lower bit codec, if such lower bitrate codecs are used by default.

There is no standardised list of VBD discrimination tones; see Section 14.6 for a fuller list of such tones that may be recognised by NGNs or NGAs.

The most common VBD tone is 2100Hz tone, used in higher speed ITU V series modem standards. However, the use of 2100Hz tone in other VBD CPE tends to be rather haphazard. Some security equipment receiving centre equipment issues 2100Hz tone, though most do not. Some EPOS terminal receiving centre platforms issue 2100Hz tone, though most do not. Some telemetry equipment uses 2100Hz tone, though most do not.

14.2 The effect of plain 2100Hz (& other VBD discrimination tones) on NLPs

Plain 2100Hz tone was originally used by CPE to disable network echo suppression devices. Such echo suppression devices have mostly been replaced by G.168 compliant echo canceller devices, which have a Non Linear Processor (NLP) component which suppresses residual echo not cancelled by the linear canceller component. The NLP part of G.168 echo cancellers are optionally disabled by plain 2100Hz tone, but if so will re-enable after 100ms to 400ms of bidirectional silence (ie may re-enable at 100ms-400ms; will re-enable at 400ms or more). This is significant for V23 modem applications, which although sometimes issue 2100Hz, are half duplex and therefore allow the NLP to re-enable at the end of the first data burst.

NICC document ND1704 mandates the use of echo cancellers that are compliant to Recommendation ITU-T G.168, but note that removal of the NLP on detection of 2100 Hz tone is optional in G.168 and therefore may not be implemented in all networks.

Other VBD discrimination tones may have the same effect on NLPs as plain 2100Hz, though this behaviour is not standardised. See Section 14.6 for a fuller list of VBD discrimination tones.

14.3 The effect of 2100Hz with phase reversals (& other VBD discrimination tones) on NLPs & ECs

2100Hz tone with phase reversals is used by CPE to fully disable network G.168 echo cancellers. This is used by any modem which includes its own echo canceller function, specifically V.32, V.32bis, V.34, V.90 and V.92 modems. These modems are all full duplex, and so the echo canceller will remain disabled for the duration of the call.

This echo canceller behaviour is mandated in NICC document ND1704 by reference to ITU document G.168.

Other VBD discrimination tones may have the same effect on NLPs & ECs as 2100Hz with phase reversals, though this behaviour is not standardised. See Section 14.6 for a fuller list of VBD discrimination tones.

Note that some network implementations may also remove the echo canceller on detection of amplitude modulated 2100 Hz tone (ANSam) as defined in ITU-T V.8.

14.4 The effect of 2100Hz (& other VBD discrimination tones) on adaptive jitter buffers

The behaviour of jitter buffers when encountering fax or dialup modems is currently mandated in NICC document ND1704, which currently states that jitter buffers should then transition upwards into a higher length of 42ms. This requirement is predicated on the assumption that if a VBD application is recognised as such by the NGN/NGA on a call, then the impact of a initial large upwards adaptation but no subsequent adaptations will on average be less than the impact of no initial upwards adaptation but the chance of one or more smaller upwards adaptation occurring later in the call. This is not a black and white judgement, and is currently being reviewed within NICC QoS WG, as CPE compatibility test results to date suggest that the potential advantages of fixing jitter buffers in this way may be, more often than not, outweighed by the detrimental effects of start of call upward adaptations and of resulting increased e2e delay. At the time of writing, a submission to the NICC QoS WG is being progressed that proposes changes to address these considerations.

With current ND1704 compliant NGNs/NGAs therefore, when 2100Hz tone (with or without phase reversals) is recognised, the jitter buffers on a call will increase in length up to 42ms. This will increase RTD by the difference between the initial length of the jitter buffer on the call and this final value, time the number of jitter buffers on a call. The number of jitter buffers on a call carried over a single NGN/NGA, or any NGNs/NGAs interconnected via IP, will be one in each direction of the call (one at each edge of the NGN/NGA domain), i.e. two in the round trip. However it is possible for calls to be routed via NGN-TDM-NGN or NGA-NGN-TDM-NGN-NGA type routings, which will therefore have a set of jitter buffers for each IP ‘island’, and will therefore result in a higher RTD increase than a single NGN/NGA case when all the jitter buffers become fixed.

Other VBD discrimination tones may have the same effect on jitter buffers as 2100Hz, though this behaviour is not standardised. See Section 14.6 for a fuller list of VBD discrimination tones.

14.5 Duration of Effects

The effect of a 2100Hz tone, and potentially other VBD discrimination tones, occurring on either direction of a call on an NGN or NGA, will effect jitter buffers, ECs and NLPs with differing durations as follows:

	Jitter Buffers	**Linear Echo Canceller**	**Non Linear Processor (low level echo suppressor)**
No tone	Adaptive	Enabled	Enabled
Plain 2100Hz tone (and some other VBD discrimination tones)	Fixed at 42ms for duration of call (current behaviour – see Section 14.4)	Enabled	Optionally disabled until next bidirectional silent period of 100ms to 400ms (ie may re-enable at 100ms-400ms; will re-enable at 400ms or more)
2100Hz tone with phase reversals (and some other VBD discrimination tones)	Fixed at 42ms for duration of call (current behaviour – see Section 14.4)	Disabled until next bidirectional silent period of 100ms to 400ms (ie may re-enable at 100ms-400ms; will re-enable at 400ms or more)	Disabled until next bidirectional silent period of 100ms to 400ms (ie may re-enable at 100ms-400ms; will re-enable at 400ms or more)

14.6 List of VBD Discrimination Tones

Whilst 2100Hz tone, with or without phase reversals, is the most commonly encountered VBD CPE discrimination tone, there are many others which may be detected by NGN/NGA edge equipment (MSANs, ATAs and Media Gateways) and trigger the echo canceller, NLP and jitter buffer behaviour described above. A fuller set of VBD discrimination tones is listed in the table below, though this is not exhaustive.

It is important to note that whether a particular VBD tone is detected by a particular manufacturer's MSAN, ATA or Media Gateway is not standardised, and is determined by the choice of the manufacturer or by the network operator's requirement specification. This also applies to the resulting behaviour of the echo canceller, NLP and jitter buffer behaviour, which is also not standardised. The only VBD discrimination tone for which behaviour is mostly standardised is 2100Hz tone.

Tone	Frequency	Echo Canceller	NLP	Jitter Buffer
ANS / CED tone; ANSam (V.8)	2100 Hz	Enabled*	Optionally Disabled*	Fixed*
ANS and ANSam tone with phase reversals	2100 Hz	Disabled*	Disabled*	Fixed*
T.30 CNG fax tone	1100 Hz	box dependent	box dependent	box dependent
V.18 Annex A (Baudot) text telephone tones	1400 Hz followed by 1800 Hz or vice versa	box dependent	box dependent	box dependent
V.21 T.30 fax pre-amble flag sequence	(0x7E where 0 = 1850 Hz and 1 = 1650 Hz).	box dependent	box dependent	box dependent
V.21 low channel 1 mark (response) tone	980Hz	box dependent	box dependent	box dependent
V.21 low channel 1 space tone	1180Hz	box dependent	box dependent	box dependent
V.21 high channel 2 mark tone	1650Hz	box dependent	box dependent	box dependent
V.22, V.22 bis and Bell 212	2250Hz	box dependent	box dependent	box dependent
V.23 FSK tones	Mark & Space frequency transitions	box dependent	box dependent	box dependent
V.23 forward channel mark tone, also non-speech calling tone	1300Hz	box dependent	box dependent	box dependent
V.23 backward channel mark tone	390Hz	box dependent	box dependent	box dependent
V.32	1200 Hz followed by 1800 Hz carrier	box dependent	box dependent	box dependent
V.8 bis initiation tones (ESi)	1375Hz / 2002Hz	box dependent	box dependent	box dependent
V.8 bis response tones (ESr)	1529Hz / 2225Hz	box dependent	box dependent	box dependent
Bell 103 answer tone	2225Hz	box dependent	box dependent	box dependent
Bell 103 originating tone	1270Hz	box dependent	box dependent	box dependent
Bell 202 mark tone	1200Hz	box dependent	box dependent	box dependent
Bell 202 space tone	2200Hz	box dependent	box dependent	box dependent
EDT (Same as V.21 low channel)	980 Hz and 1180 Hz	box dependent	box dependent	box dependent

* See table in Section 14.5 for exact fixing / disabling behaviour for 2100Hz tone.

These VBD discrimination tones may also be recognised by NGNs and NGAs as valid triggers to upspeed the codec used to G.711 from a lower bit codec, if such lower bitrate codecs are used by default. Such codec upspeeding behaviour is common but is not standardised.

14.7 Testing NGNs & NGAs for VBD Discrimination Tone Detection

If the set of VBD discrimination tones recognised by a given NGN/NGA edge device, along with their subsequent triggered behaviours, is not defined with sufficient confidence, it may be necessary to test the relevant MSAN, ATA or Media Gateway in order to discover or confirm these behaviours. Any particular VBD discrimination tone may trigger one or more of:

- The echo canceller to disable (normally until the next period of bidirectional silence)
- The NLP to disable (normally until the next period of bidirectional silence)
- The jitter buffer to transition to fixed mode (for the duration of the call).

14.7.1 Testing for echo canceller disabling

Test equipment is available that can measure, single ended, the level of echo received on a call. Firstly, a measurement of the normal level of echo should be made that exists on a call to a particular destination, in the absence of any VBD discrimination tones. Next, the test VBD discrimination tone in question should be generated using a signal generator or a piece of test CPE of known behaviour. A measurement of the received echo should be made within 100ms of this VBD tone (see Section 14.5 for the timings of echo canceller re-enablement) and compared with the normal level of echo existing prior to the transmission of the VBD tone. A large increase in the received level of echo would indicate that the VBD tone used has triggered the echo canceller to disable. Note that if the echo canceller has been disabled, the NLP will be disabled as well.

14.7.2 Testing for NLP disabling

The same type of equipment and test setup should be used as described in Section 14.7.1. If on use of the VBD tone in question, the level of increase in received echo is small, then the VBD tone used in the test will be shown to have caused the NLP to disable, but not the echo canceller.

14.7.3 Testing for jitter buffer fixing

Test equipment is available that can measure, single ended, the round trip delay on a call, using correlated timing derived from the received far end echo. Firstly, a measurement of the normal round trip delay should be made that exists on a call to a particular destination, in the absence of any VBD discrimination tones. Next, the test VBD discrimination tone in question should be generated using a signal generator or a piece of test CPE of known behaviour. A second measurement of the round trip delay should be made and compared with the first A large increase in the measured round trip delay, of the order of several tens of milliseconds, would indicate that the VBD tone used has triggered the jitter buffer on one or both of the NGN/NGA edge devices to transition to fixed mode. For example, an increase in round trip delay of 74ms on an NGN to NGN call may indicate that two jitter buffers (one in each MSAN) have each increased by 37ms, i.e. from 5ms to 42ms. This calculation assumes that the jitter buffer behaviour in each case is ND1704 compliant (this requirement is currently under review; see Section 14.4), but in any case, any increase in round trip delay of this sort of magnitude is likely to indicate one or both jitter buffers transitioning to fixed mode. Note that on an NGN to legacy call, or an NGA to legacy call, there will still be two jitter buffers involved, -one at the NGN/NGA edge (MSAN or ATA) and one at the Media Gateway interconnect to legacy TDM network.

15 CPE Testing Guidance

This section describes the full set of test scenarios that can be undertaken for voice band data CPE and voice CPE.

Testing VBD CPE
The following list summarises the tests that should be undertaken for VBD CPE. Each test is described in more detail in Sections 15.1 to 15.3 below.

- Sensitivity to e2e delay
- Sensitivity to jitter buffer adaptations
- Functional test under all appropriate combinations of:
    - Call routes
    - Line losses
    - Communications direction
    - Supported functions

Voice CPE
The following list summarises the tests that should be undertaken for Voice CPE, described in more detail in Section 15.4.

- Dialling out
- Ringing and call answer
- Voice quality
- CLI
- SMS
- Other functions

15.1 Testing VBD CPE for Sensitivity to End to End Delay

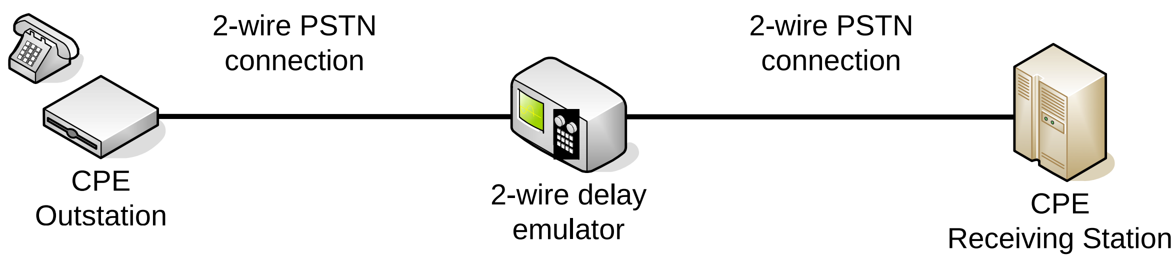
Information about e2e delay on NGNs and NGAs, and its potential impact on VBD CPE is contained in Section 7.

The test for sensitivity to end to end delay does not require a NGN/NGA line, and can be executed independently from any NGN/NGA platform. This is because the effect of end to end delay is the same, whatever the underlying platform, and this means that TDM based technologies can, and should be used. The use of TDM based test equipment is strongly preferable, because no other detrimental effects are likely to be introduced to any test calls, which would potentially confuse the cause of any failures.

Test equipment is available that can be used to introduce increased end to end delay in test calls. This type of test can be set up in a number of ways, three of which are described here. CPE operation should at least be tested against the 300ms round trip delay stated in ND1701, but operation should preferably be additionally tested to the point of failure, with the value of RTD at the point of failure recorded.

The impact of line loss on the sensitivity of VBD CPE to e2e delay can normally be disregarded. Testing should however include relevant combinations of event/transaction reporting/polling, and other supported CPE functions, as described in Section 15.3.

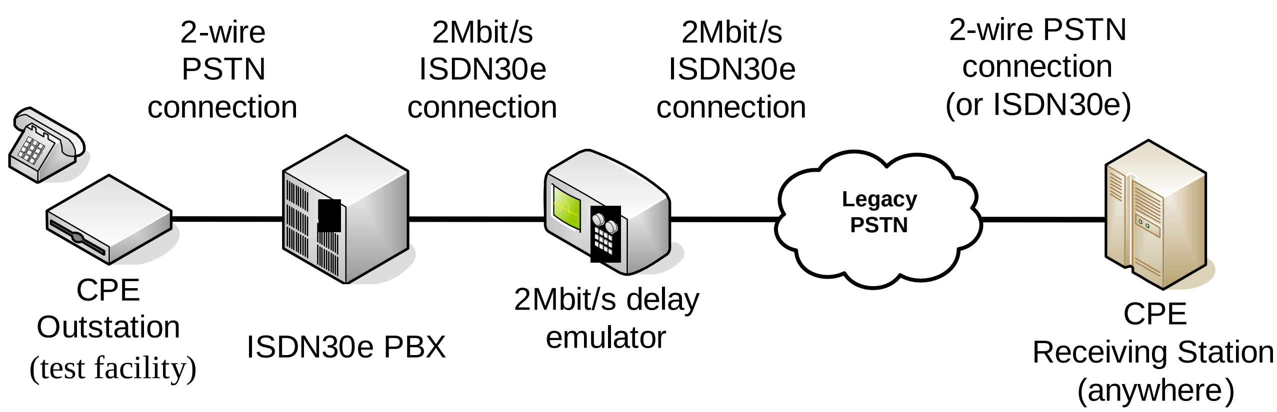
15.1.1 Stand Alone Delay Emulator



This method can be used when both ends of the VBD CPE application is available at the same location. The CPE outstation is connected to one side of the delay emulator, and dials through to the CPE central station connected to the far side of the delay emulator.

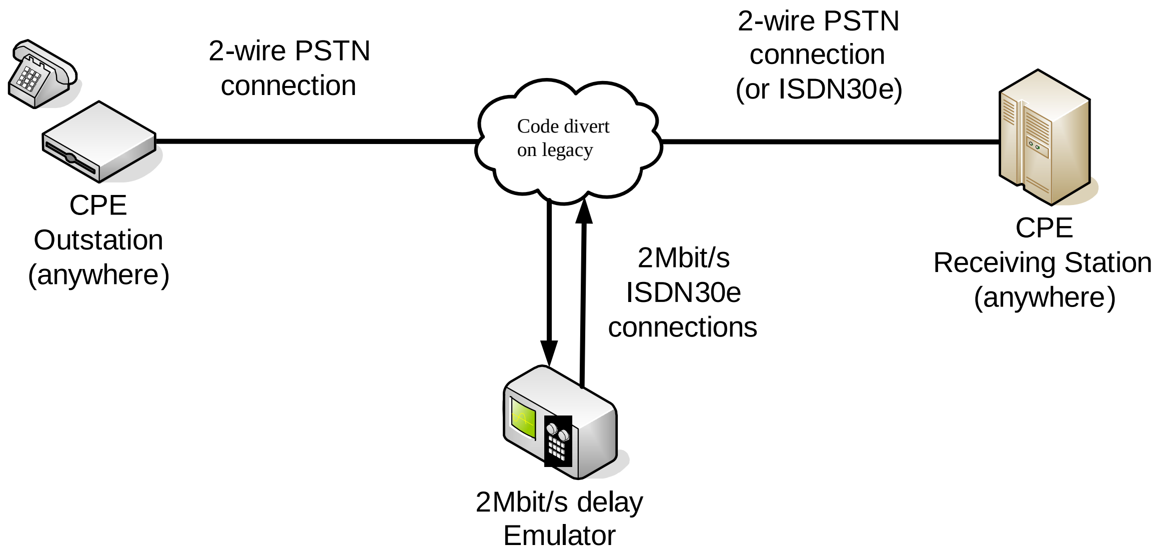
15.1.2 Dial-Through Delay Emulator

This method can be used where the CPE outstation (eg card payment terminal etc) is available at the test facility location, but the CPE central station (eg the card payment server into which the card payment terminal dials) is not portable, and the live CPE central station must be used. The CPE outstation is connected to the dial-through delay facility, and dials through to the live CPE central station located anywhere on the live legacy PSTN. This method of introducing additional delay will provide a high and guaranteed minimum RTD of a certain value, rather than a precise figure, unless the RTD is actually measured, as described in Section 15.1.4, on a test call using the test rig.



15.1.3 Remote Dial-up Delay Code

This method can be used where neither the CPE outstation (eg card payment terminal etc) nor the CPE central station (eg the card payment server into which the card payment terminal dials) are available at the test facility location. An ‘Indirect Access’ type network prefix code can be set up by a network operator to provide a remote dial-up facility. This code is added to the normal number that the CPE outstation dials, and the call is routed through a delay emulator. Both the CPE outstation and the CPE central station can therefore be located independently anywhere on the live legacy PSTN network, assuming both ends are on lines that support the prefix code. This method of introducing additional delay will provide a high and guaranteed minimum RTD of a certain value, rather than a precise figure, unless the RTD is actually measured, as described in Section 15.1.4, on a test call using the code.



15.1.4 Measuring e2e Delay

Test equipment is also available that can actually measure end to end delay on a given call, which can be useful in confirming the level of expected end to end delay on an individual test call. This typically relies on the technique of measuring the round trip delay of far end echo on a call. This technique normally requires that all echo cancellers on a call route are switched off, which is achieved by the use of an initial 2100Hz tone, which will cause all G.168 compliant echo cancellers to disable, immediately followed by an audio burst for which the echo can be detected and correlated in time. Note that this technique cannot unambiguously measure the round trip delay on any call routes that contains any IP network hops, as the 2100Hz tone used by the test equipment will normally cause jitter buffers to transition to fixed mode, substantially increasing the overall round trip delay for that call.

15.2 Testing VBD CPE for Sensitivity to Jitter Buffer Adaptations

Information about jitter buffer adaptations on NGNs and NGAs, and its potential impact on VBD CPE is contained in Section 8.

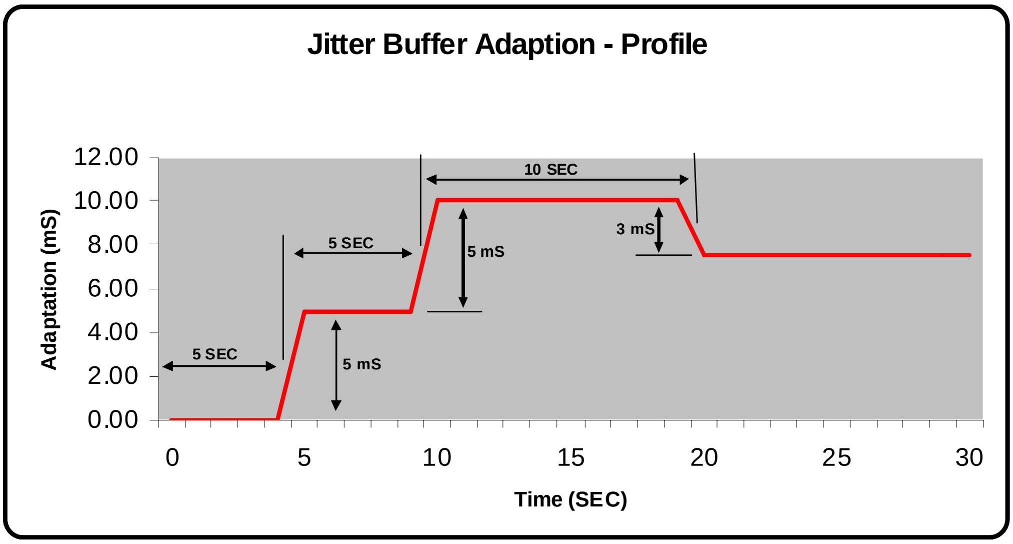
It might be thought that the most realistic method of testing the impact of jitter buffer adaptations on various types of CPE would be to use an IP network emulator to inject artificially increased levels of jitter into the backhaul of an MSAN or ATA. The jitter levels used could then be varied to mimic the expected real world jitter profile, or a speeded up version of it, and to trigger the jitter buffers at each end point to adapt accordingly. However this approach is very problematic due to:

- the relatively long periodicity of even a speeded up jitter profile compared to the very short duration nature of many of the test calls required.

- the difficulty of ensuring that a provoked jitter buffer adaptation occurs at the precise moment required, i.e. at a particular point in a call lasting only a few seconds.
- interpreting test results arising from this artificially provoked testing being difficult because it will always be difficult to know exactly how many adaptations have been triggered, and at what size, and at what points in time.

An alternative and preferred test methodology is to emulate the effect of jitter buffer adaptations on the transmission channel using a TDM based delay emulator. When a jitter buffer adapts up or down, it produces a step change in end to end delay, an effect which can be exactly reproduced by changing the e2e delay applied using the delay emulator. This can be done at any point in a call, at a highly controllable moment in time. This method may be somewhat less benign to CPE test calls than some jitter buffers might be, as some jitter buffers will, where possible, wait for silent periods before they adapt. However it is better for a test to be slightly more conservative (i.e. more aggressive) than it is for it to be insufficiently testing.

The preferred method of testing CPE for sensitivity to jitter buffer adaptations therefore involves using a TDM based delay emulator to apply a sequence of upward and downward adaptations which represents a ‘reasonable worst case’ scenario for calls likely to be encountered. Such a ‘reasonable worst case’ jitter buffer adaptation test profile should take into account the total number of network hops that may reasonably be encountered, the resulting maximum likely total level of jitter to be encountered by each of the two network endpoints, and the behaviour of the jitter buffer control algorithms at each end point (which may be different). An illustrative example of such a jitter buffer adaptation test profile is shown below, but it should be stressed that the actual profile used should be the result of considering the above factors for the actual networks involved.



The impact of line loss on the sensitivity of VBD CPE to jitter buffer adaptations can normally be disregarded. Testing should however include relevant combinations of event/transaction reporting/polling, and other supported CPE functions, as described in Section 15.3.

15.3 Functional Testing of VBD CPE

Functional testing of the VBD CPE should be undertaken on actual NGN/NGA voice lines in order to test for sensitivity to echo canceller operation, balanced ringing, reduced loop current, delay to dial tone, post dialling delay, post answer delay, voice codec, on-hook voltage, and any modem, fax or DTMF relay technique that may be implemented. Further information on these effects can be found in previous sections of this document.

Testing should be undertaken using all appropriate combinations of:

- Call routes. Testing should be undertaken under all call route combinations:
  - legacy PSTN to legacy PSTN (baseline check)
  - legacy PSTN to NGN/NGA
  - NGN/NGA to legacy PSTN
  - NGN/NGA to NGN/NGA
- Line losses. Testing should be undertaken under all appropriate line losses:
  - NGN MSAN voice lines: the UK access network exchange to customer range of 0dB to 15dB.
  - See Section 15.5 for more information on the distribution of line losses in the UK access network.
  - Street based NGA ATA voice lines: the UK access network cabinet to customer range of 0dB to perhaps 5dB [Note: this figure is a suggested estimated only]
  - Customer based NGA ATA voice lines: customer premises wiring range of 0dB to perhaps 1dB [Note: this figure is a suggested estimated only]
- Communications direction. Testing should be undertaken for both directions of CPE communications:
  - Outstation CPE calling central station CPE (reporting)
  - Central station CPE calling outstation CPE (polling)
- Supported functions. Testing should be undertaken for all appropriate functions of the VBD CPE, for example:
  - Event / transaction reporting and polling calls
  - CPE management or configuration calls
  - Firmware or software update calls
- Additionally, a line migration & regression test may be appropriate, which tests the effect of the line disconnection occurring during any migration from legacy PSTN line to an NGN/NGA voice line, and regression back in the reverse direction. This test is particularly relevant to telecare / social alarm CPE, to determine whether the line disconnection that occurs during migration or regression causes the CPE to locally alarm and/or require a reset by the end customer.

Depending on the level of confidence required, and the criticality of the VBD CPE application, it may be considered sufficient to use a selected subset of the significant number of possible combinations of the variables listed above.

15.4 Testing Voice CPE

Functional testing of Voice CPE should be undertaken on an actual NGN/NGA voice line in order to confirm successful operation of the CPE on the NGN MSAN line card, or the NGA ATA interface. This will check operation against balanced ringing, reduced loop current, on-hook voltage and any other characteristic of the telephony interface.

Testing should be undertaken with the test CPE item connected to the NGN MSAN line card or NGA ATA interface, calling to and called from a far end proven reference voice CPE item. Testing should be undertaken against:

- all appropriate line losses as described in Section 15.3
- all basic CPE functions, including:
    - Dialling out

      DTMF and loop-disconnect
    - Ringing

      Normal and any alternative distinctive ringing cadences
    - Voice quality

      Normally only a basic subjective check of voice quality is required. Note: full objective voice quality testing of the NGN/NGA network itself is a substantial and separate activity.
- other relevant supported CPE functions, such as:
    - CLI performance

      Testing CLI performance properly requires a high number of call / CLI transmission attempts. Success rates need be determined to the nearest percentage point if small difference in performance are to be determined, which requires call attempt volumes of the order of 100 attempts per CPE item. This is made easier with an automatic call generator
    - SMS

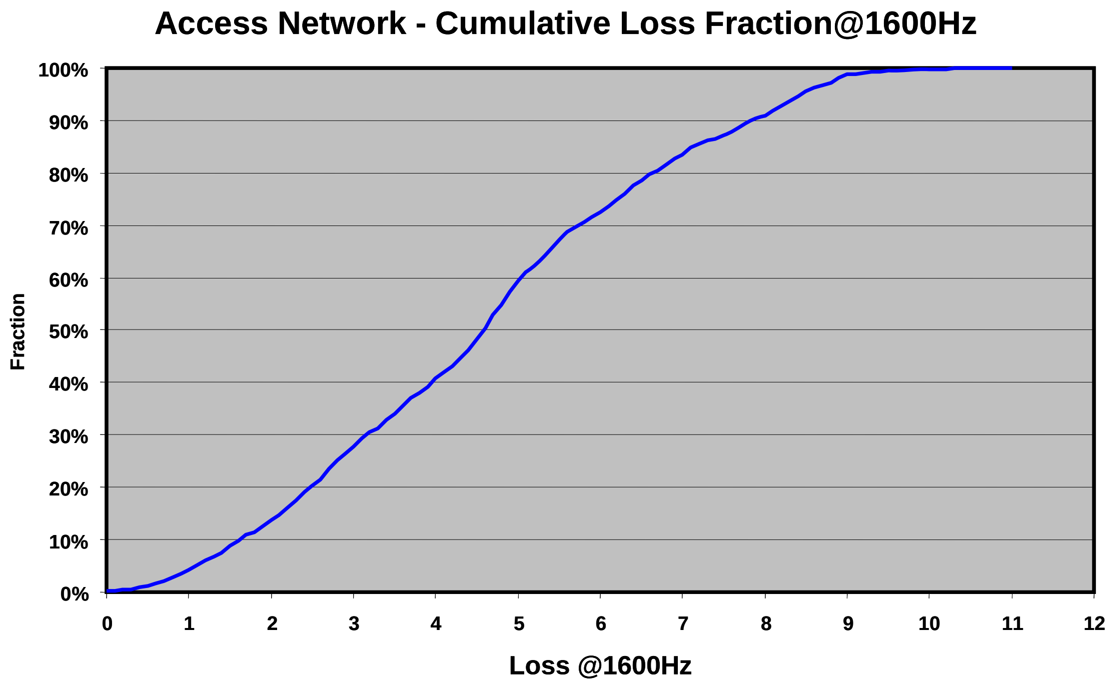
      Similar considerations to the testing of CLI apply.
    - Answering machine ringing trip
    - Answering machine line release
    - Answering machine record/playback
    - Memory
    - Display operation
    - Loudspeaking operation
    - Payphone call length timing start

15.5 Line Loss in the UK Access Network

Testing against different line losses should use fixed or variable line loss attenuators designed to emulate lines in the UK access network. These are sometimes calibrated in km of 0.5mm copper, or but the actual line loss / attenuation in dB is the most meaningful parameter to use, as line length can equate to different line losses depending on the thickness of copper pair assumed.

15.5.1 Distribution of Line Loss in the UK Access Network

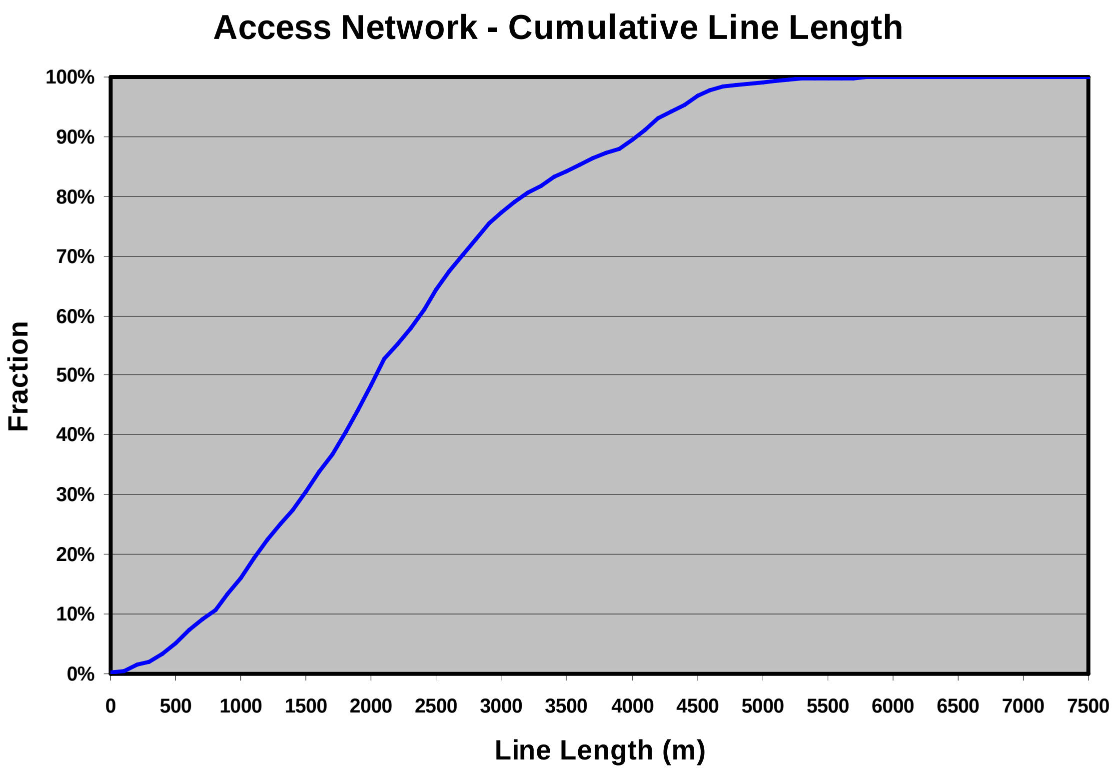
The planning limit for lines (copper pairs) used to provide PSTN in the UK access network is 15dB @ 1600Hz. However, the large majority of lines are between 0dB and 10dB line loss @ 1600Hz. The following graph shows the distribution of calculated dB loss @ 1600Hz for a sample of around 4,000 lines in the UK access network. The loss is calculated from lengths and gauges of cable segments taken from cable records for each line.



In this sample of lines, no line exceeds 15dB @ 1600Hz line loss, and 99.8% of lines are actually within 10dB @ 1600Hz line loss. However the number of lines at the extreme end of the sample is low, and these percentages cannot be guaranteed to be completely representative of the UK access network as a whole.

15.5.2 Distribution of Line Length in the UK Access Network

The following graph shows the distribution of line lengths for a sample of around 4,000 lines, taken from cable records for each line.



In this sample of lines, no line exceeds 6km line length, and over 99% of lines are within 5km line length. However the number of lines at the extreme end of the sample is low, and these percentages cannot be guaranteed to be completely representative of the UK access network as a whole.
It should be noted that neither line loss (in dB @ 1600Hz) nor DC loop resistance (ohms) can be simply calculated from line length, as any given line will typically be made up of a number of cable segments, each of differing pair gauge (thickness), and therefore of differing line loss per km and different loop resistance per km.

15.5.3 Line Loss, Line Card Gain & e2e Attenuation

On the UK legacy PSTN, line card gain is used to compensate for access network line loss. An AGC gain setting is typically used on lines with loss between 0dB and 10dB, with the AGC having a total range of 2.5dB adjustment. There are two additional fixed gain settings available, typically used on lines up to 12.5dB and 15dB respectively, with each setting introducing an additional 2.5dB. Thus there is a total range of 0dB to 7.5dB gain available on legacy PSTN line cards.

When testing CPE against a particular line loss, introduced using a line loss emulator, it is important to ensure that the line card gain setting is configured correctly for the line loss being used.

Though it should not normally be necessary to do so, the total e2e attenuation on a connection can be calculated by adding up the line loss at each end, the effective line card gain at each end, and the effective e2e network attenuation. For the purposes of the figures presented here, the e2e internal network attenuation is effectively 12dB. Hence the total e2e attenuation on a legacy PSTN connection should lie between:

- A minimum of (0dB near end line loss - 0dB near end line card gain + 12dB internal network attenuation – 0dB far end line card gain + 0dB far end line loss) = 12dB e2e attenuation
- A maximum of (15dB near end line loss – 7.5dB near end line card gain + 12dB internal network attenuation – 7.5dB far end line card gain + 15dB far end line loss) = 27dB e2e attenuation

16 Moving to IP based CPE

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.

History

Document history		
1.1.1	March 2011	Initial publication