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NICC Document

Guidance on CPE Compatibility for All-IP Telephony Networks

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Contents

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Foreword

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

1 Scope

The present document contains guidance on the potential issues associated with the operation and compatibility of VBD (Voice Band Data) CPE (Customer Premises Equipment) (CPE) and voice CPE on All-IPT (All-IP Telephony) Networks.

The guidance contained in this document was originally published by NICC at the request of Ofcom and was intended for specific technical user groups with specific objectives in mind:

CPE Manufacturers and Service Providers

- To assist with 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.
- To assist in appropriate compatibility testing of such CPE against All-IPT Networks.
- To assist in the procurement of network access and services that meet the operational quality requirements of VBD systems.

All-IPT Network Designers and Operators

- To assist in the design and operation of All-IPT Core and Access Networks in order to provide transmission channels that minimise the disruption to the operation of VBD CPE and associated applications, and for the connection and use of voice CPE.

However, since the creation of the All-IP Task Group in 2016, NICC and Ofcom have recommended that CPE manufactures should cease development of VBD CPE. CPE manufactures should not rely on the use of ATAs (Analogue Telephone Adaptors) or ISDN SIP gateways (for example) to send VBD over IP telephony services. Instead, CPE manufactures should seek to develop native IP-based solutions using well-established technologies and protocols for data transmission such as Hyper-text Transfer Protocol (HTTP).

It has also been recommended that CPs and their customers seek to migrate away from VBD CPE to native IP implementations to minimise the risk of loss of service. This risk is expected to increase as more CPs implement voice services in accordance with ND1704 [1] that supports reliable transmission of human speech but minimal, unguaranteed support for Voice Band Data.

The ability for CPE Manufacturers and Service Providers to adequately test and validate the performance of VBD, or diagnose issues affecting the operation of VBD, is also expected to become increasingly difficult as the related test equipment is phased out or discontinued (see Section 16).

Despite the many complex technical issues to overcome, until VBD CPE has been fully phased out, some CPs may wish to develop, or continue to provide, networks that support VBD services. Alternatively, they may choose to implement measures which are conducive to more successful operation of VBD service while not fully supporting the performance and features necessary to guarantee reliable transmission in all cases.

For these reasons, ND1431 will remain in publication until VBD CPE is fully phased out, but it has been updated to make the position of NICC and Ofcom clear: CPs and their customers should migrate away from VBD CPE.

2 Informative References

[1] NICC ND1704: "End-to-End Network Performance Rules & Objectives for the Interconnection of NGNs"

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

[3] ETS 300 001: Attachments to Public Switched Telephone Network (PSTN); General technical requirements for equipment connected to an analogue subscriber interface in the PSTN Chapter 3: Ringing signal characteristics

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

[5] ITU-T Rec. Q.23: Technical features of push-button telephone sets

[6] ITU-T Rec. Q.24: Multifrequency push-button signal reception

[7] ITU-T Rec. V.21: 300 bits per second duplex modem standardized for use in the general switched telephone network

[8] ITU-T Rec. V.22: 1200 bits per second duplex modem standardized for use in the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits

[9] ITU-T Rec. V.22bis: 2400 bits per second duplex modem using the frequency division technique standardized for use on the general switched telephone network and on point-to-point 2-wire leased telephone-type circuits

[10] ITU-T Rec. V.23: 600/1200-baud modem standardized for use in the general switched telephone network

[11] ITU-T Rec. V.34: A modem operating at data signalling rates of up to 33 600 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits

[12] ITU-T Rec. V.90: A digital modem and analogue modem pair for use on the Public Switched Telephone Network (PSTN) at data signalling rates of up to 56 000 bit/s downstream and up to 33 600 bit/s upstream

[13] ITU-T Rec. V.92: Enhancements to Recommendation V.90

[14] British Standard BS8521: Specification for dual-tone multi-frequency (DTMF) signalling protocol for social alarm systems

[15] ITU-T Rec. G.161 Interaction aspects of signalling processing networks

[16] ITU-T Rec. G.711 Appendix II: A comfort noise payload definition for ITU-T G.711 use in packet-based multimedia communication systems

[17] ND1444: DTMF Best Practice Guide (BPG)

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

[19] ITU-T Rec. T.30: Procedures for document facsimile transmission in the general switched telephone network

[20] ITU-T Rec. V.150.0: Modem-over-IP networks: Foundation

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

[22] ITU-T Rec. V.151: Procedures for the end-to-end connection of analogue PSTN text telephones over an IP network utilizing text relay

[23] ITU-T Rec. V.152: Procedures for supporting voice-band data over IP networks

[24] ETSI ES 201 235: Specification of Dual Tone Multi-Frequency (DTMF)

[25] ITU-T Rec. V.32: A family of 2-wire, duplex modems operating at data signalling rates of up to 9600 bit/s for use on the general switched telephone network and on leased telephone-type circuits

[26] ITU-T Rec. V.32bis: A duplex modem operating at data signalling rates of up to 14 400 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits

[27] ITU-T V.8 : Procedures for starting sessions of data transmission over the public switched telephone network

[28] ETSI EN 300 659-1 Access and Terminals (AT); Analogue access to the Public Switched Telephone Network (PSTN); Subscriber line protocol over the local loop for display (and related) services; Part 1: On-hook data transmission

[29] IETF RFC 2833: RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals

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

[31] ND1016: Requirements on Communications Providers in relation to Customer Line Identification display services and other related services

3 Definitions, symbols and abbreviations

3.1 Definitions

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

All-IP Network Designers and Operators: Technical experts involved in the design and operation of All-IP Access and/or All-IP Core networks that support All-IPT Services.

All-IPT Services: All Internet Protocol Telephony Services are delivered over All-IP Core Networks and All-IP Access Networks.

All-IP Core Networks: A carrier class network used in place of a traditional circuit switched network, typically carrying multiple services over a converged IP based architecture.

All-IP Access Networks: 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, street-based cabinets or local exchange.

Analogue Telephony Adaptor: An electronic device that connects standard analogue telephones to VoIP (Voice over Internet Protocol) systems.

bis and ter: bis means second and ter means third (Latin). With reference to the ITU-T V-Series modem modulations, these are second and third revisions (respectively).

CPE Manufacturers: Technical experts involved in the design and manufacturer of voice and/or Voice Band Data CPE that are intended to operate over All-IPT Services.

Handshaking: Refers to the transmission that takes place between VBD CPE at the start of a call before any actual data can be exchanged. This may include negotiation of error control, connection speed and evaluation of analogue line characteristics in order to optimise transmission performance.

Media Gateway: A network device that translates media streams from one transmission technology to another (such as TDM to IP and vice-versa), converts media streams from one audio format to another (such as decoding and encoding from one codec family to another) or performs other interworking functions that involve changes to media streams.

Multi Service Access Node (MSAN): A generic edge of network device. This may be exchange or cabinet based, depending on network topology.

Non Linear Processor: The residual echo suppressor part of an ITU-T Rec. G.168 [2] compliant echo canceller.

Primary Edge (PE) Routers: A network router typically located at the edge (or boundary) between internal and external networks with respect to the network owner (for example a router proving connectivity between one service provider's networks and another).

REN (Ringer Equivalence Number): Represents the electrical loading effect of a telephone ringer on a telephone line. In the UK, a maximum REN of 4 is allowed in accordance with ETS 300 001, Chapter 3 [3]

Service Providers: Technical experts involved in the provision of All-IPT.

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

3.2 Abbreviations

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

AJB	Adaptive Jitter Buffer
ATA	Analogue Telephony Adapter
ATM	Automatic Teller Machine
CLI	Calling Line Identity
CPE	Customer Premises Equipment
CPS	Carrier Pre-Select
DECT	Digital Enhanced Cordless Telecommunications
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
HTTP	Hypertext Transfer Protocol
Hz	Hertz
IP	Internet Protocol
IPT	Internet Protocol Telephony
ISDN	Integrated Services Digital Network
ITU	International Telecommunications Union
JB	Jitter Buffer
JBA	Jitter Buffer Adaptation
MDCT	Modified Discrete Cosine Transform
ms	milliseconds
MSAN	Multi Service Access Node
NGA	Next Generation Access
NGN	Next Generation Network
NLP	Non Linear Processor
PCM	Pulse Code Modulation
PDV	Packet Delay Variation
PLC	Packet Loss Concealment
PPM	Parts Per Million
PSTN	Public Switched Telephone Network
REN	Ringer Equivalence Number
RTD	Round Trip Delay
RTP	Real-time Transport Protocol
SBC	Session Border Controller
SIP	Session Initiation Protocol
SMS	Short Messaging Service
SLIC	Subscriber Line Interface Circuit
VBD	Voice Band Data

4 Introduction to All-IP Telephony Services

All-IP Telephony (All-IPT) Services are delivered over All-IP Core Networks and All-IP Access Networks. All-IP Core Networks replace the traditional circuit switched network. All-IP Access Networks replace part or all of the copper/aluminium infrastructure in the access network. This includes (but is not limited to) FTTP (Fibre To The Premises), FTTC (Fibre To The Cabinet) and HFC (Hybrid Fibre Access) technologies.

All-IP Access Networks will typically be used to provide derived voice services alongside Customer Premises Equipment (CPE) such as a home router with an integrated Analogue Telephone Adapter (ATA) and/or support for IP phones and/or Digital Enhanced Cordless Telecommunications (DECT) phones.

Most of the qualitative descriptions in this document relate to voice channel behaviour and apply equally to both All-IP Core and All-IP Access Networks. However, it is more difficult to provide quantitative descriptions of All-IP Access Networks that incorporate twisted pair copper and/or aluminium due to variation of the electrical characteristics. With both core and access networks, the quantitative behaviour of the various potential voice channel impairments can vary significantly from one design and implementation to another: not all core and access networks are necessarily created equal with respect to compatibility with VBD CPE and voice CPE.

Traditional Public Switched Telephone Networks (PSTNs) are circuit switched networks. When a call is made, a dedicated data path is established 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 ITU-T Rec. G.711 [4] codec, which generates a continuous 64 kbit/s data stream. It is this 64 kbit/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 remains constant except when a transmission fault causes an in-call re-route of the transmission path.

All-IPT Services are carried over packet-based networks. There is no continuous point-to-point physical connection set up when a call is made. Instead, packets containing multiple audio samples (typically, the equivalent of 20 ms of audio) are sent from the IP source system to the IP destination system by the use of an appropriate address attached to each packet. A delay is incurred in the packetisation process whilst the send buffer (for example 20 ms) is filled with multiple audio samples (whereas in switched circuit networks every individual audio sample is transmitted as soon as it is converted, which incurs sub-ms delay).

Packets during a particular All-IPT voice call 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. In a well-managed IP network, the delay variation should be relatively small, but it may not be possible to guarantee this at all times. In an All-IPT call, the audio codec used to convert from analogue audio signal to audio data, and vice versa, may or may not be the same ITU-T Rec. G.711 [4] codec typically used in traditional circuit switched networks. The audio data stream produced by whichever codec is used, is packetised, sent across the IP network, reassembled at the far-end IP system 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.

Most All-IPT calls are likely to encounter:

- Increased end-to-end delay compared to traditional circuit switched network calls. See Section 7 for more detail on end-to-end delay.
- Audio channel discontinuities caused by Jitter Buffer Adaptations, which do not occur on traditional circuit 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 circuit switched network calls. See Section 9 for more detail on echo cancellation.
- Packet loss due to contention with other traffic, de-jitter buffer discards, transmission channel impairments and clock timing drift where network components and end systems are not fully and accurately synchronised.

Some All-IPT calls may also encounter:

- Use of a voice codec other than the ITU-T Rec. G.711 [4] codec typically used on traditional circuit switched networks. Such codecs may have lower bitrate or different compression characteristics than ITU-T Rec. G.711 [4] 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 circuit 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, All-IPT CPE may encounter:

- Reduced (albeit in-spec) maximum off-hook loop current compared with the PSTN. See Section 14.1 for more details on reduced loop current.
- Balanced ringing rather than unbalanced ringing. See Section 14.2 for more details on balanced ringing.

5 Introduction to Voice Band Data CPE

5.1 Types of VBD CPE

CPE that makes a call and then transfers any sort of data to any sort of receiving CPE is a VBD application. It is difficult to accurately keep track of volumes of different VBD CPE, but decades of development has led to a large legacy install base and, despite the advice of the NICC and Ofcom to cease development, there are cases of providers continuing to sell 'new' VBD equipment together with long-term service contracts.

The following table contains an indicative list of CPE that use VBD transmission along with typical data transmission methods used in the UK:

Table 1: VBD CPE and transmission methods

CPE/Application	Transmission method (also see Table 2)
Satellite TV STBs (set-top boxes)	V.34, V.90
Fax machines	V.17, V27ter, V.29, V.34
Dial-up Internet access	V.34, V.90 and V.92
Security alarm panels	DTMF, V.34
Telecare social alarms	DTMF
Lift Emergency Phones	DTMF, V.34
EPOS (Electronic Point of Sale) card sale terminals	V.22bis, V.34
Pre-payment card top-up machines	V.22bis, V.34
Fire alarms	DTMF, V.34
Automated Teller (cash) Machines	V.22bis, V.34
Telemetry remote monitoring	V.22, V.22bis, V.23, V.34
SMS (Short Message Service) phones	V.23
Text Phones	V.21, Baudot
Telehealth (eHealth) remote monitoring	V.34, V.90, V.92
Payphone remote management	V.34
PBX (Private Branch Exchange) remote management	V.34
Custodial tagging devices	V.34
CLI (Calling Line Identification) phones, display boxes	V.23

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.

Until quite recently, VBD CPE outstations have typically been connected to one or more individual analogue voice lines (generally on the legacy PSTN) or via one or more ISDN connections. However, individual voice lines are increasingly being migrated to All-IPT, while primary rate ISDN connections are increasingly being migrated to IP-based SIP trunking products.

There is considerable inertia in some of the relevant industry segments to moving from VBD implementations to native IP implementations, despite there being many well-established technologies and protocols for data transmission (such as HTTP). CPE manufactures, as well as CPs, must encourage the migration to these alternatives and away from VBD CPE. Progress in the migration from VBD CPE to IP based CPE is seen as critical to maintaining the functionality currently relied upon by customers of both CPE manufactures and serving CPs.

5.2 Voice Band Data transmission method characteristics

Table 2 below contains further information about the VBD transmission methods identified in Section 5.1.

Table 2: VBD transmission methods

Transmission method	**Standards reference**	**Description**
DTMF	ITU-T Rec. Q.23 [5], Q.24 [6]	Used by security alarm panels, fire alarms, telecare and 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 and Social alarms protocols include BS8521 [14]. DTMF based protocols are generally half duplex.
V.21	ITU-T Rec. V.21 [7]	Used by most text relay phones. 300 bit/s. Full duplex.
V.22	ITU-T Rec. V.22 [8]	Used by some telemetry equipment. 1200 bit/s or 600 bit/s Full duplex.
V.22bis	ITU-T Rec. V.22bis [9]	Used by all EPOS terminals, all pre-payment card top-up machines, all ATM machines, and some telemetry equipment. 2400 bit/s. Full duplex.
V23	ITU-T Rec. V.23 [10]	Used by most telemetry equipment. Also used by CLI phones, display boxes and other CLI CPE, and by fixed line phones with SMS. 1200 bit/s Half duplex.
V.34	ITU-T Rec. V.34 [11]	Used by some dial up internet modems, some fax machines, first generation satellite TV boxes, telehealth equipment, and management functions in a wide range of CPE. 33.6 kbit/s. Full duplex.
V.90, V.92	ITU-T Rec. V.90 [12], V.92 [13]	Used by most dial up internet modems, second generation satellite TV boxes and telehealth equipment. 56 kbit/s downstream and 33.6 kbit/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 All-IPT core and access edge devices as VBD discrimination tones. Of these, 2100 Hz, with and without phase reversals, is the most common, but there are many more. The effect of most of these VBD tones on jitter buffers and echo cancellers in All-IPT networks 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 15.

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
- Hotline Autodiallers
- Call Barring Devices
- Distinctive Ringing Devices
- REN (Ringer Equivalence Number) Boosters
- Extension Ringers
- 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).
- Native IP phones and terminals (including voice-enabled IoT, smart speakers, software clients on PCs etc.)
- Loop Disconnect (i.e. ‘rotary dial’) phones – (not typically supported) by ATAs)

The scale of compatibility issues faced by voice CPE when connected to All-IPT networks is likely to be relatively small provided that best practice guidelines are followed. Most of the potential compatibility issues described in this document relate to Voice Band Data CPE. Nevertheless, there are some potential compatibility issues with All-IPT networks that do relate to voice CPE, and these are described in Section 14.

7 End-to-end delay

7.1 End-to-end delay on All-IPT networks

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 circuit switched network, for normal geographic number routeing, is typically between a minimum of 4 ms for calls originating and terminating on the same DLE, and a maximum of 50 ms for a worst-case routeing.

On All-IPT networks the RTD will normally be higher than on circuit switched networks for the same type of call.

For example, the RTD on calls over a single UK wide All-IPT core network, for normal geographic number routeing, will typically be between a minimum of 30 ms for calls originating and terminating on access equipment located on the same Physical Edge (PE), and a maximum of 75 ms for a worst-case routeing. The RTD on similar calls between a circuit switched network and an All-IPT core network from the same network operator, will typically be between a minimum of 40 ms and a maximum of 90 ms. However, the RTDs encountered on an All-IPT core network will depend on the specific architecture and design used and may exceed the values given here especially if the originating or terminating end points utilise the mobile access network.

The RTD on calls carried over an All-IP access network will depend on how the call is carried beyond the boundaries of the access network. For calls carried across an All-IP access - All-IP core - All-IP access route, the RTD values encountered are likely to be greater than the values described above for All-IPT core networks.

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 All-IPT network, it is important to understand the minimum and maximum RTD values that can reasonably be expected to occur on the relevant core or access network 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 access, core and, if applicable, TDM network routes.
- This bottom-up analysis to also be done for all relevant call route scenarios involving call routeing 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 PEs (Primary Edge routers) SBCs (Session Border Controllers), MGWs (Media Gateways), MSANs (Multi Service Access Nodes), ATAs (Analogue Termination Adaptors) or other IP end system.
- A top-down measurement of real end-to-end delays on all relevant call route scenarios in order to ensure compliance with the limits 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. These include the use of:

- Non-geographic numbering
- Number portability
- Carrier pre-select
- Indirect access codes
- VBD discrimination tones (see Section 15)
- Call routeing across multiple operators' networks

ND1704 [1] contains guidance on the calculation of Round Trip Delay that calls on fixed networks in the UK should encounter, along with ND1704 [1] Annex IV which provides a calculator for specifying performance criteria for relevant call scenarios.

7.2 Minimising end-to-end delay on All-IPT networks

End-to-end delay on All-IPT networks can be minimised by:

- Using adaptive jitter buffers (see Section 8), as required by ND1704 [1] for All-IP core networks.
- Meeting the end-to-end delay limits for IP connected All-IP core networks defined in ND1704 [1].
- Providing decoding of non-geographic numbers to avoid tromboning calls out to other networks for such decoding.
- Ensuring endpoints (e.g. ATAs, IP phones etc.) have nominally zero source jitter.
- Choosing a suitable packetisation time such as 10 ms (see also Section 10).

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 often have time-outs built into them, i.e. an amount of time in which the CPE expects to see a response to a given transmission. If, or when, the time-out is exceeded, the CPE may attempt to 'retry' the transmission, and it may do this a number of times before assuming there is an issue and hanging up.

Clearly, the time-out must take into account both the network RTD and the acceptable, or expected, response time of the receiving centre equipment or receiving CPE. It has often been in the interests of customers and/or providers to keep this time-out to a minimum as it helps to keep call duration (and hence cost) to a minimum whilst optimising performance of the service. This was relatively straightforward to achieve on traditional circuit switched networks because of the relatively predictable (and low) RTD.

For these reasons, the protocol timeout for VBD CPE is often close to the RTD encountered on traditional circuit switched networks. When such CPE is migrated to an All-IPT voice line, it is at high risk of failing. The degree of risk of failure is dependent on:

- Network Round Trip Delay
- Far-end CPE response time
- The maximum response time allowed by the protocol (i.e. the protocol time-out)

The increased RTD encountered in All-IPT networks compared to traditional circuit switched networks is described in Section 7.1, and protocol time-outs clearly need to allow for this RTD. Protocol time-out must 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. Therefore a security alarm panel may work successfully on a given network RTD with the equipment of one receiving centre, but fail with the slower responding equipment of another receiving centre. The same can be true of other categories of VBD CPE.

There are still risks to consider even when the relevant protocol time-outs lead to successful operation over a given network RTD and with a given far-end CPE. For example, if there is little safety margin, then there will be increased risk of failure if any aspect of the call routeing changes. Changes that may increase the RTD on calls made, include:

- Instances where the VBD user and/or Service Provider were using the same All-IPT Network Operator but move to different All-IPT Network Operators, thus incurring additional network hops for interworking.
- The introduction of Carrier Pre-Select (CPS), causing calls to be carried by another operator’s network.
- The introduction of Indirect Access codes, causing calls 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.
- Number Translation Services (NTS) which are provided by another network node.

Additional network RTD may be incurred where jitter buffer changes are invoked on All-IPT access and core networks. The resulting increase in the RTD on a call can be very significant. 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 the time of publication, the maximum network RTD expected on calls made over the UK network is subject to the access type and technologies used. ND1704 [1] provides guidance on acceptable delays to achieve a good quality voice service however, this does not guarantee successful transport of legacy VBD signalling.

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 or may not be tightly defined and paired with the transmitting CPE. Also, it may or may not be owned and operated by the same organisation. If not tightly paired with each other, the operators of each end may need to co-operate to ensure that CPE response times are understood and accounted for in the relevant protocol time-outs. Industry level co-operation, co-ordination and standardisation may also be required.

8 Adaptive Jitter Buffers

8.1 Adaptive Jitter Buffers on All-IPT networks

In an All-IPT network, any deviations from the regular arrival times of packets are referred to as packet jitter, or Packet Delay Variation (PDV). These 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.

Jitter buffers in All-IPT networks 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 end-to-end delay, but may suffer overflow or underflow when maximum levels of jitter are encountered. To meet these two potentially conflicting objectives, jitter buffers in All-IPT networks 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 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 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 Implementing Adaptive Jitter Buffers on All-IPT networks

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.

8.2.1 Mid-call adaptations

Jitter buffer adaptations can be avoided on calls made over any one IPT network if the minimum and default jitter buffer length are configured sufficiently higher than the maximum level of jitter occurring on that All-IPT core/access network. 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 All-IPT network 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 All-IPT network, 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 All-IPT core/access network.

It is unlikely, however, that the minimum and default jitter buffer lengths can be configured sufficiently high enough to avoid adaptations when encountering levels of jitter potentially occurring on calls routed over multiple operators' All-IPT core networks. The maximum level of incoming jitter to any All-IPT core network is, in principle, limited by ND1704 [1]. However, 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 All-IPT core network.

8.2.2 Start of call adaptations

The behaviour of jitter buffers when encountering fax or dial-up modems is mandated in ND1704 [1] Rule PD12, which states that fixed de-jitter buffers shall be used.

8.2.3 Synchronisation

The loss of synchronisation of any network component in the All-IPT network 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 applicable stand-alone clock. Similarly, any All-IPT termination equipment (such as an 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 applicable device (e.g. 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 50 ppm, will produce a regular 5 ms Jitter Buffer Adaptation every 100 seconds.

For maximum compatibility with VBD CPE, All-IPT ATAs should be synchronised to the head end of the All-IPT access network when the service is declared a VBD service as per the ND1704 [1] Definitions.

When the service is declared a 'Voice service' as per the ND1704 [1] Definitions, the decision to implement synchronisation (or deploy devices with more stable and accurate clocks) will come down to a balance of technical and economic feasibility, subjective impact on voice quality and whether a provider wishes to implement measures to protect, albeit not guarantee, successful VBD operation.

8.3 Impact of Jitter Buffer Adaptations on VBD CPE

8.3.1 Effect of Jitter Buffer Adaptations

As is described in more detail in Section 8.1, when a jitter buffer adapts upwards in length, CPE will receive an unexpected period of silence in the reconstructed audio stream. While if that jitter buffer adapts downwards in length, the CPE will receive an unexpected period of missing audio data (or cuts) in the reconstructed 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 will still lead to VBD signals being shifted forwards (where the jitter buffer adapts upwards) or backwards (where the jitter buffer adapts downwards) relative to any preceding transmission. If such timing is important to a specific VBD applications, the effect of the 'retiming' may cause transmission issues.

8.3.2 Frequency and magnitude of Jitter Buffer Adaptations

The maximum frequency of Jitter Buffer Adaptations on the transmission channel of calls carried across All-IPT core/access networks is difficult to predict. It 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 All-IPT termination equipment on an All-IPT access network is unsynchronised to the head end of the All-IPT access network, proportional to the magnitude of the difference in clock speed between the incoming packet streams and the standalone clock.

The size of upward and downward adaptations of an adaptive jitter buffer on an All-IPT core/access network is dependent on the exact jitter buffer design but will, typically, be in the order of tens of milliseconds.

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.

8.3.3 When Jitter Buffer Adaptations occur

CPE may encounter audio channel discontinuities caused by Jitter Buffer Adaptations in the following circumstances:

Start of call Adaptations

- If a VBD CPE call is recognised as such by the network, then depending on the network implementation, a relatively large inserted silence period may be encountered by CPE. This may occur either at the start of the call, or shortly after the detection of a VBD discrimination tone. It is caused by jitter buffers at each end of the call adapting upwards as they transition into fixed mode (as described in Section 8.2.2).

  Any upward adaptations, and the resulting silence periods, are likely to occur at approximately the same time, but will not be precisely synchronised. This is because each jitter buffer, and its VBD discrimination tone detection system, is autonomous.

  It is also possible that one detection system detects a VBD discrimination tone, but the other does not. Or that one jitter buffer is configured to transition to fixed mode on VBD detection, but the other is not. In these cases a large inserted silence period might occur at one end of the call only, and so in one direction of transmission only. The use and length of jitter buffers in interconnected All-IPT core/access networks in fixed mode is mandated in ND1704 [1] for VBD services.

- If the level of jitter at the start of a call is higher than the initial length of the de-jitter buffer, then one or more upwards adaptations will occur. The jitter buffer control algorithm will increase the buffer size until there is enough 'headroom' (or safety margin) between the adapted jitter buffer length and the level of jitter.

Mid-call adaptations

- If the level of jitter increases significantly at any point in a call, then one or more upwards adaptations may occur. The jitter buffer control algorithm will increase the buffer size until there is enough 'headroom', or safety margin, between the adapted jitter buffer length and the increased level of jitter.
- If the level of jitter decreases significantly at any point in a call, then one or more downwards adaptations may occur. The jitter buffer control algorithm will decrease the buffer size until the 'headroom', or safety margin, between the adapted jitter buffer length and the decreased level of jitter, is not excessive.

8.4 Minimising Sensitivity of VBD CPE to Jitter Buffer Adaptions

As indicated in Section 8.3.2, 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 in Section 8.2.3, a 5 ms upward adaptation caused by a loss of network equipment synchronisation (or lack of network equipment synchronisation by design) is a very long period compared to many commonly encountered symbol rates, equating to 100% of a 200 baud symbol length, or 20% of a 40 baud 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.
- Early transmission of 2100 Hz 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 applications. 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 in Section 16 of this document.

However, as stated in Section 1, the overriding recommendation is to avoid such issues in the first place; VBD CPE should be replaced as soon as practical with a suitable native IP-based solutions using well-established technologies and protocols for data transmission (such as HTTP).

9 Echo cancellation

9.1 Echo cancellation on All-IPT networks

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

Echo cancellers (ECs) used on both traditional circuit switched networks and All-IPT core/access networks will typically be built to meet the requirements of ITU-T Rec. G.168 [2]. Such ECs comprise two separate parts:

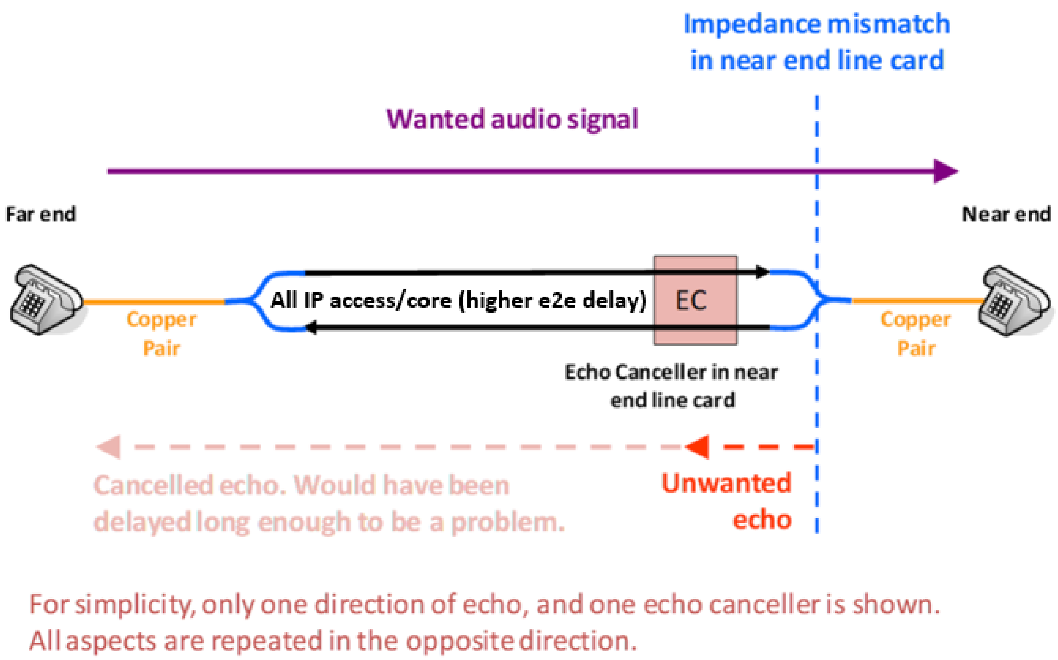
- Linear echo canceller, which cancels most of the echo.
- Non Linear Processor (NLP), which suppresses the remaining low level residual echo.

ITU-T Rec. G.168 [2] 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 many years on traditional circuit switched networks for international calls and some national calls. There exists the potential for the signal clipping described above to occur on any call made over a circuit switched network that encounters echo cancellation.

Echo cancellation will typically be used by default on all voice calls on All-IPT networks due to the increased end-to-end delay encountered by such calls (see Figure 1 below). There will therefore typically be an increased likelihood for the signal clipping described above to occur on calls made over All-IPT networks

Figure 1: Simple echo path with increased end-to-end delay



It is important to note that echo cancellers are not only contained in the All-IPT core/access edge device (for example, serving node or ATA) but also in Media Gateways that interconnect to circuit switched networks. The design and operation of these echo cancellers may well be different.

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

9.2 Optimising echo canceller implementation on All-IPT networks

The implementation of echo cancellers in digital networks is described in ITU-T Rec. G.168 [2], which should be adhered to when implementing any echo canceller in All-IPT core/access networks. One piece of text contained in Appendix I of ITU-T Rec. G.168 (‘Guidance for Application of Echo Cancellers’) is particularly relevant to All-IPT networks that explicitly support VBD services per ND1704 [1] and is therefore quoted here:

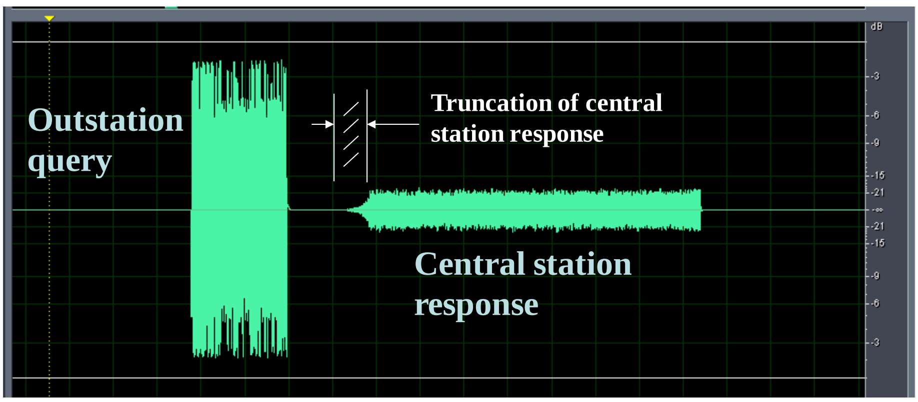
‘PSTN network planners were expected to continue to evolve the network in such a way that it would not knowingly prevent the continued carriage of a permissive voice band data/facsimile service.’

Annex B of ITU-T Rec. G.168 [2] 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, which has a recommended maximum of 5 ms. This parameter describes how quickly the NLP should go from active (i.e. cancelling an unwanted residual echo) to inactive (i.e. allowing a wanted signal through) after the onset of a wanted signal at the input of the echo canceller. An NLP that has an operate time for this transition that is significantly greater than the recommended maximum is likely to have a significantly increased potential to clip data signals.

9.3 Impact of echo cancellation on VBD CPE

ITU-T Rec. G.168 [2] 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.

Figure 2: VBD transmission truncated by relatively slow NLP

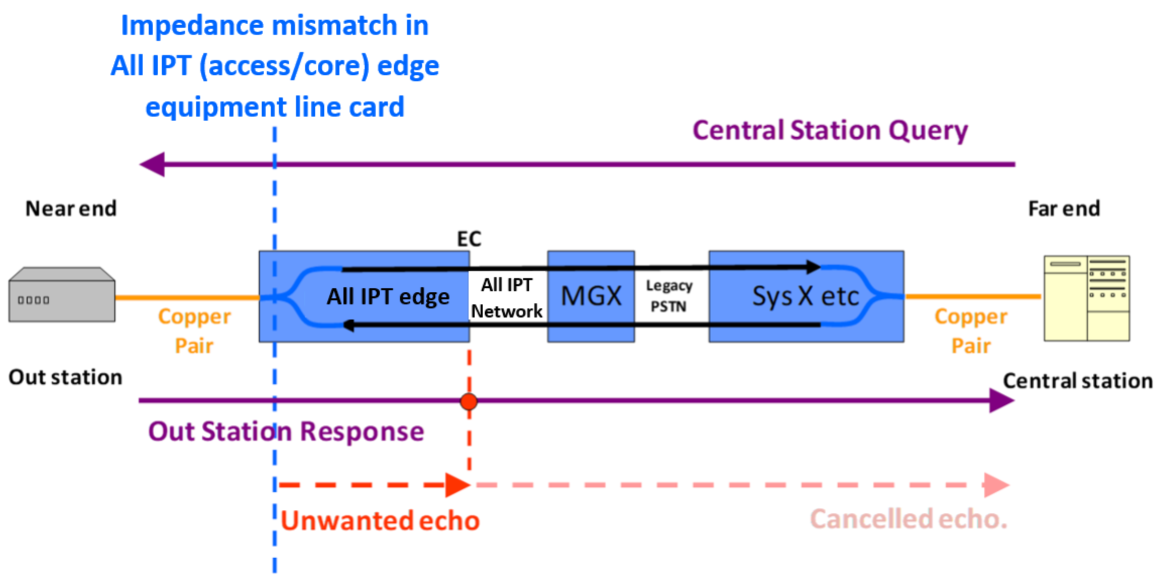


Clipping can in principle occur in two directions, as described in the next two sections.

9.3.1 Potential impact of serving node/ATA echo canceller operation

The echo canceller contained in the All-IPT edge equipment can potentially truncate a very fast response from CPE connected to the edge equipment. For example, a fast responding EPOS terminal or telemetry out station connected to an All-IPT core/access network, could have its response truncated by the NLP on that edge device (the central station may be on either traditional PSTN or an All-IPT core/access network). This is shown in the following diagram.

Figure 3: NLP clipping at edge equipment due to fast responding outstation

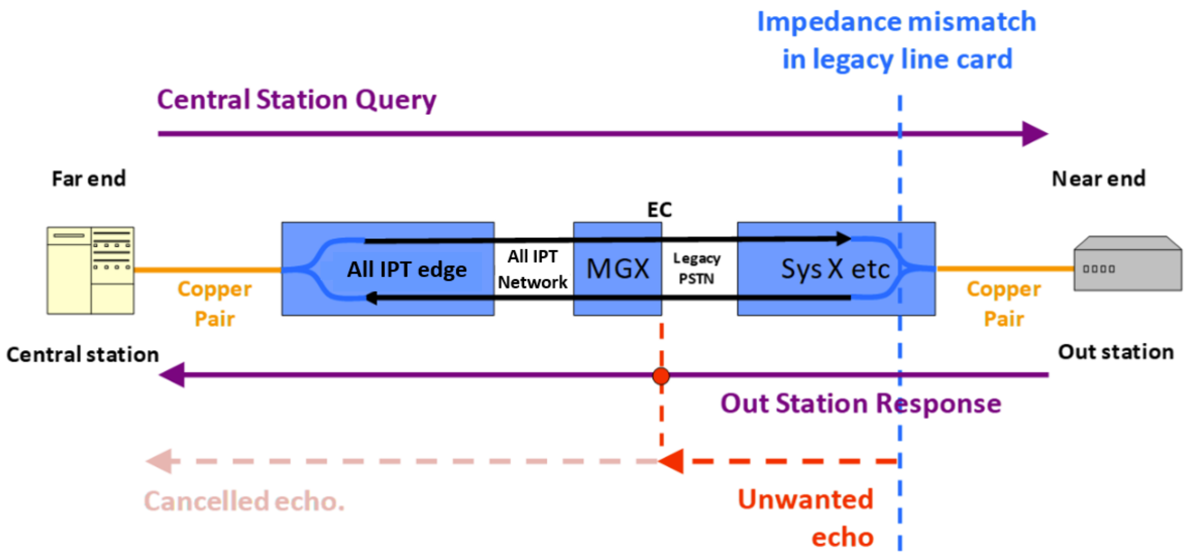


In the above diagram, the unwanted echo from the All-IPT core/access network edge equipment is cancelled by the echo canceller in that edge device. A fast responding out station CPE at the All-IPT core/access network end may have its response clipped by the NLP operation in the serving node/ATA echo 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 traditional PSTN line card. For example, a fast responding EPOS terminal or telemetry out station connected to a traditional PSTN line card, has its response truncated by the NLP on the relevant Media Gateway trunk card. (The central station is on an All-IPT network) This is shown in the following diagram.

Figure 4: NLP clipping at MGW due to fast responding outstation



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 traditional 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 canceller operation

If a failure of VBD CPE occurs on an All-IPT core/access line in either of the above two configurations, the capture of audio traces at the far end and near end two-wire points will help to confirm whether the failure mechanism is sensitive to echo canceller operation. Furthermore, if the echo canceller can be configured to be permanently disabled at either the All-IPT core/access network edge equipment, or the Media Gateway, then this will help confirm the cause of failure.

VBD CPE needs to be tested for sensitivity to NLP clipping or any other aspect of echo canceller operation by making test calls on real-world All-IPT core/access lines, as this is the only way that the effect of the actual All-IPT core/access echo canceller will be seen. See Section 16 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 ITU-T Rec. G.168 [2]. 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 below.

‘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.’

ITU-T Rec. G.168 [2] makes repeated reference to ITU-T Rec. G.161 [15] ‘Interaction Aspects of Signal Processing Network Equipment’, and Section I.4 mentioned above specifically references ITU-T Rec. G.161 [15]. 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 below.

‘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.’

It may be possible to protect the operation of VBD CPE against NLP clipping by one or more of the following:

- The use of 2100 Hz 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 15. Note however, that NLP disabling on detection of 2100 Hz tone is optional in the relevant standard (ITU-T Rec. G.168 [2]) and will depend on the exact configuration of the relevant All-IPT network.
- Use of a Full Duplex transmission modulation because once disabled the NLP and/or echo canceller will remain disabled. This is in contrast to half duplex, which is likely to allow the NLP and/or echo canceller to re-enable following any period of 100 ms – 400 ms of bidirectional silence (see Section 15 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 All-IPT networks

On any digital telephone network, an audio codec is used 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 circuit switched networks, the codec used is typically ITU-T Rec. G.711 [4], which produces a 64 kbit/s stream of audio data. There are three options in ITU-T Rec. G.711 [4] which are of relevance:

- Whether the ‘A-law’ or ‘μ-law’ method is used to improve the handling of the wide dynamic range found in speech. The μ-law method is used in North America and Japan, while A-law is used in Europe and the rest of the world.

  This is applicable to both traditional circuit switched networks and to All-IPT networks.

- Whether a packet loss concealment technique is used (such as ITU-T Rec. G.711 [4] Appendix I). 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 circuit switched networks, which do not use packet transmission.

- Whether silence suppression is used (such as ITU-T Rec. G.711 Appendix II [16]). 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 circuit switched networks, which do not use packet transmission.

ND1704 [1] stipulates that for VBD services, the ITU-T Rec. G.711 [4] A-law codec shall be used without silence suppression (see section 10.2 for further information). There are no specific rules regarding Packet Loss Concealment (PLC). PLC will not help to improve VBD services (since it is designed to ‘fill in the gaps’ of missing speech signals as opposed to operating as any kind of data correction method that would help VBD signals in any way). However, PLC in itself should not adversely impact VBD as it should only operate when there are missing packets and under these circumstances VBD operation can become unpredictable anyway.

ND1704 [1] stipulates that for Voice services calls are not limited to ITU-T Rec. G.711 [4]. It encourages the use of modern codecs that provide increased audio bandwidth, low bit rates and (to an extent) resilience to errors and packet loss, with the aim of improving perceived call quality over and above traditional circuit switched telephony.

10.2 Impact of voice codecs on VBD CPE

The types of audio channel distortions caused by the use of different types of audio codecs available is a large and complex subject.

If the same ITU-T Rec. G.711 [4] audio codec is used on an All-IPT network as is used on traditional circuit switched networks, then VBD CPE designed for use on traditional circuit switched networks should encounter no difference in the quality of the audio channel with regard to audio coding (echo canceller operation, Jitter Buffer Adaptations and network performance variation aside).

If codecs other than ITU-T Rec. G.711 [4] are used, then problems are likely to occur with VBD CPE. The severity of errors will ultimately depend on the specific codec. However, broadly speaking, basic codecs that are variants of PCM (Pulse Code Modulation) will probably be less problematic than modern codecs based on LPC (Linear Predictive Coding) and MDCT (Modified Discrete Cosine Transform). This is because the latter use knowledge of psychoacoustics to significantly lower the required bit rate in a way that preserves intelligibility of human speech, but significantly degrades actual signal integrity.

As a simple example, it may not be necessary to transmit all frequencies in a speech signal because when heard by a human those frequencies are masked. Whereas those signal components could be critical to a VBD modem transmission.

Similarly, some audio codecs are unable to support the transmission of DTMF sufficiently for accurate digit recognition and therefore relay mechanisms must be used (see Section 11 for further information regarding DTMF).

10.3 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 (e.g. those listed in Section 8.4) will not necessarily give any protection against the use of codecs other than ITU-T Rec. G.711 [4]. Such codecs cause audio distortions that are constantly present in the audio channel.

As noted above in Section 10.2, the severity of such distortions will depend on the specific codec used. Noted below are some mitigating actions that might work for codecs based on variants of PCM (Pulse Code Modulation), but are unlikely to be effective against modern codecs utilising psychoacoustic speech compression techniques:

- 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 2100 Hz VBD discrimination tone in order to renegotiate the codec choice to ITU-T Rec. G.711 [4].

11 DTMF relay, fax relay and modem relay

11.1 DTMF relay, fax relay and modem relay on All-IPT networks

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. The Voice Band Data signal is then 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. Keypad presses are transmitted by the handset across the mobile network, and not as in-band DTMF signals, but as ‘User pressed key 1’ and ‘User pressed key 2’ messages etc. For a mobile call to the traditional PSTN the in-band DTMF is 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. It will 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 All-IPT networks

If the design aim of an All-IPT core or access network is to offer a quality and transparency of transmission channel as close as possible to a traditional circuit switched network – as if often the case for VBD services - then the use of any relay technique needs to be avoided.

As such, ND1704 [1] stipulates that for VBD services, DTMF, fax and data should be carried in-band using ITU-T Rec. G.711 [4]. Specifically, this is try and ensure reliable handling of time sensitive, machine-machine transmission of DTMF, fax and data.

However, for ‘Voice services’ (as defined in ND1704 [1]), the use of modern codecs that provide increased audio bandwidth, low bit rates and (to an extent) resilience to errors and packet loss is encouraged to improve perceived call quality (see also Section 10.1). Therefore, ND1704 [1] stipulates that for Voice services, DTMF relay (using IETF RFC 2833 [29] and 4733 [30] RTP Telephony Events) should be used to carry DTMF.

ND1444 [17] contains more information covering the best practice for carrying DTMF across All-IPT networks.

11.2.1 Use of DTMF Relay

As previously noted, ND1704 [1] stipulates that for VBD services, DTMF should be carried in-band using ITU-T Rec. G.711 [4]. DTMF relay should not be used for VBD.

For ‘Voice services’ (as defined in ND1704 [1]) DTMF relay (using IETF RFC4733/2833 RTP Telephony Events [5]) should be used to carry DTMF.

ND1444 [17] contains more information covering the best practice for carrying DTMF across All-IPT networks.

11.2.2 Use of Fax relay

Fax relay is commonly implemented according to ITU-T Rec. T.38 [18], which standardises the carriage of standard ITU-T Rec. T.30 [19] fax across an IP network.

Taking the example of an ITU-T Rec. T.30 [19] compliant fax machine connected to an All-IPT network equipped with ITU-T Rec. T.38 [18]:

- The ITU-T Rec. T.30 [19] fax machine dials the far end fax number as normal
- The call is terminated on the near edge of the network, and the 14,400 kbit/s (or lower rate) fax data is extracted and transmitted across the network in IP packets according to ITU-T Rec. T.38 [18]
- At the far edge, the full ITU-T Rec. T.30 [19] fax signals are regenerated and transmitted to the far end fax machine.

As far as UK interoperability standards are concerned, ND1704 [1] stipulates that for VBD services, fax should be carried in-band using ITU-T Rec. G.711 [4] (i.e. not using fax relay)

11.2.3 Use of modem relay

Modem relay is implemented in a similar manner to fax relay, and is standardised by ITU-T Rec. V.150.0 [20], V.150.1 [21], V.151 [22] and V.152 [23].

The most likely supported modulation for modem relay is ITU-T Rec. V.34 [11]. This may also support an ITU-T Rec. V.90 [12] and V.92 [13] modem that trains down to the lower ITU-T Rec. V.34 [11] rate.

Support on modem relay gateways for older, lower rate modem standards (such as ITU-T Rec. V.22 [8], V.22bis [9] and V.23 [10]), 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, and switches to the use of ITU-T Rec. G.711 [4] to transmit the modem tones in-band.

As far as UK interoperability standards are concerned, ND1704 [1] stipulates that for VBD services (modem) data should be carried in-band using ITU-T Rec. G.711 [4] (i.e. not using modem relay).

11.3 Impact of relay techniques on VBD CPE

11.3.1 Impact of DTMF relay

As stated in section 11.2, ND1704 [1] stipulates that for VBD services DTMF should be carried in-band using ITU-T Rec. G.711 [4] (i.e. not using DTMF relay).

ND1444 [17] contains more information covering the best practice for carrying DTMF across All-IPT networks, but the following extract from Section 5 is included below as it is most relevant to DTMF-based VBD and why this rule applies:

'A number of implementations use machine generated DTMF sequences as opposed to tones generated directly as a result of human input (e.g. responding to Interactive Voice Response prompts by selecting digits on a keypad). This includes the following implementations and respective tone duration specifications:

- *ITU-T Rec. V.18 Annexe B The DCE (Data Circuit-terminating Equipment) shall detect characters at least 40 ms in length with silent intervals of at least 40 ms. The DCE shall transmit DTMF characters at least 70 ms in length with silent intervals of at least 50 ms.*

- *BSI Group BS8521 Telecare protocol: On duration = 80 ms (+/-5 ms). Tone sequences are sent with a (80 ± 5) ms gap between tones.*

- *BSIA Publication 255 Fast Format Alarm protocol: The duration of each DTMF digit is nominally 60 ms followed by an inter-digit pause of 60 ms silence. The minimum is 50 ms and the maximum should be 100 ms, in each case.*

In order to minimise message transmission times, these sequences may use very short pulse lengths and gaps. As such, they may be susceptible to pulse length distortion caused by tandem encoding/decoding from inband=>TE=>inband to the extent that pulse or inter-digit gap lengths are below minimums guaranteed to ensure correct digit detection.

Any terminal should not distort either pulse lengths or inter-digit gap lengths by more than +/- 10 ms when various lengths of pulse & gap (at least as great as the minimum defined above) are played through a back-to-back encoder/decoder.

Conversely, there are some applications that use quite long digit durations with tight tolerances, e.g. BSI Group BS8521 telecare protocol, which uses 1000 ms (+/-5 ms) digits.

However, adhering to Rule VBD1 in ND1704 'DTMF, fax and data should be carried in-band to reliably handle time sensitive, machine-machine transmission of DTMF, fax and data.' avoids such issues.'

11.3.2 Impact of fax relay

ITU-T Rec. T.30 [19] fax and T.38 [18] fax relay are both complex standards, and problems with faxes not communicating successfully with ITU-T Rec. T.38 [18] 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.

As previously noted, ND1704 [1] stipulates that for VBD services, fax should be carried in-band using ITU-T Rec. G.711 [4] (i.e. not using fax relay). This would avoid such issues.

11.3.3 Impact of modem relay

Implementation of modem relay techniques is again a complex area, and there is significant potential for incompatibilities with modem CPE.

If an All-IPT network employs lower bit rate codecs, then support for standard ITU-T Rec. V.34 [11] (and trained down ITU-T Rec. V.90 [12] and V.92 [13] modems), is most likely to be available.

Support for other modem types is less likely and therefore ‘modem pass-through’ (as described in Section 11.2.3) would probably be used instead. 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.

As previously noted, ND1704 [1] stipulates that for VBD services, modem data should be carried in-band using ITU-T Rec. G.711 [4] (i.e. not using modem relay). This would avoid such issues.

11.4 Maximising compatibility of VBD with relay techniques

Notwithstanding the stipulation in ND1704 [1] that for VBD services DTMF, fax and data should be carried in-band using ITU-T Rec. G.711 [4], the compatibility of VBD CPE with relay techniques can be maximised by the use of:

- Fully compliant DTMF tone generation within DTMF CPE according to ETSI ES 201 235 [24] and ITU-T Q.23 [5] (also see ND1444 [17])
- Fully complaint ITU-T Rec. T.30 [19] fax devices.
- The latest modem standards, such as ITU-T Rec. V.34 [11], ITU-T Rec. V.90 [12] and V.92 [13].

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 All-IPT networks

12.1.1 Delay to dial tone on All-IPT networks

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 circuit switched networks.

On All-IPT networks, this delay may be affected by the operation and loading of the call server that is handling the call signalling for serving nodes.

In All-IPT access networks, dial tone will typically be applied locally by SIP based ATAs.

12.1.2 Post dial delay on All-IPT networks

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 circuit switched networks, depending on the number of network hops and digit maps involved, and this will be largely unchanged in ALL-IP networks.

12.1.3 Post answer delay on All-IPT networks

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 in 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 serving node or ATA.

This delay typically totals fractions of a second on traditional circuit switched networks, and this will be largely unchanged in ALL-IP networks.

12.2 Minimising post dial delay and post answer delay on All-IPT networks

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

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

- The performance of the telephony interface (e.g. serving node, MSAN, ATA) needs to match those of line cards used in traditional circuit switched networks.
- The design and dimensioning of the relevant All-IPT call server needs particularly careful consideration to avoid undesirable increase in one or more of these delays.

12.3 Impact of 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 All-IPT network as described in Section 16.3.

Testing for sensitivity to any potential increases in these delays that may occur on a heavily loaded All-IPT network 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 exists. 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 All-IPT 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 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 and 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 CPE send and receive levels

13.1 General issues

Transmission levels impact a number of performance areas, for example:

- End-to-end transmission loss as seen by customers e.g. the loudness of telephony connections.
- Level of talker echo on telephony connections.
- Performance of echo suppression devices (which need stable and accurate transmission levels to reliably detect periods of speech and silence).

13.2 Telephony connections

Traditional end-to-end transmission losses for telephony connections have been specified in terms of Overall Loudness Rating (OLR), which takes into account performance of telephone instruments and hence represent the acoustic-to-acoustic performance as seen by customers. ITU-T recommendations in the G-series include maximum long term and short-term objectives for values of OLR.

International network planning has required an apportionment of the end-to-end OLR values into allowances for national extensions and the international transit network. The international apportionment principles can be applied to All-IPT networks. As the inter-exchange networks of UK operators use digital transmission (thus adding zero loss to the overall connection), the following simplified model can be used for transmission loss planning. OLR can be sub-divided into Send Loudness Rating (SLR) and Receive Loudness Rating (RLR) where:

OLR = SLR + RLR dB

SLR and RLR represent the transmission losses between the acoustic input/output points and the digital bit stream of the two directions of transmission. Recommended values for SLR and RLR are also included in the G-series Recommendations as shown below.

Figure 5: ITU-T Recommended SLR and RLR values

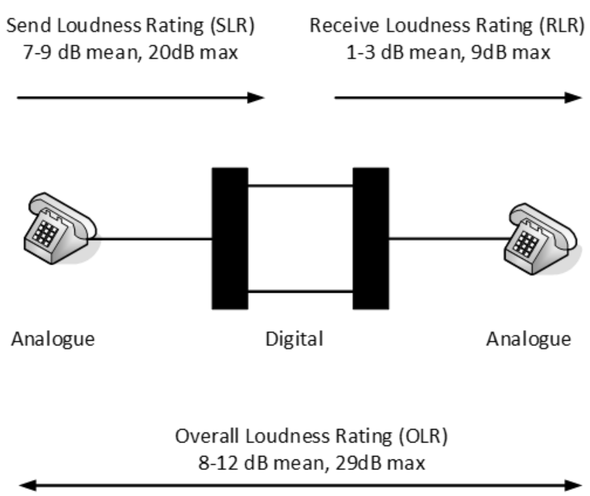
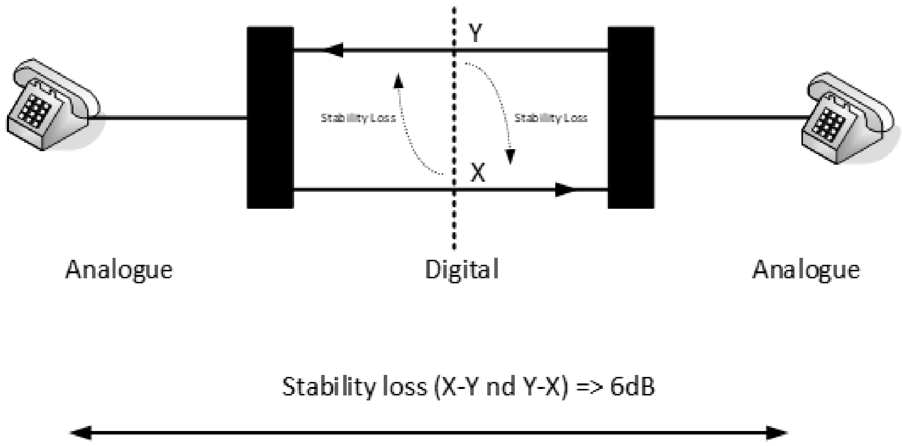


Figure 6: Stability loss calculation



Loudness rating values (OLR, SLR, RLR)

Achieving the OLR objectives within the UK requires agreement of SLR and RLR objectives.

It is recommended that All-IPT networks within the UK should meet ITU-T recommended values for SLR and RLR as shown above. Meeting these objectives will also ensure that UK international access connections are consistent with the international transmission loss plan.

13.3 Voice Band Data connections

Compliance with the recommendations for telephony (as specified in Section 13.2) is likely to result in acceptable values of transmission loss for most VBD services.

14 Other potential compatibility issues

This section describes a number of additional potential compatibility issues that may affect both VBD CPE and voice CPE.

14.1 Reduced maximum loop current

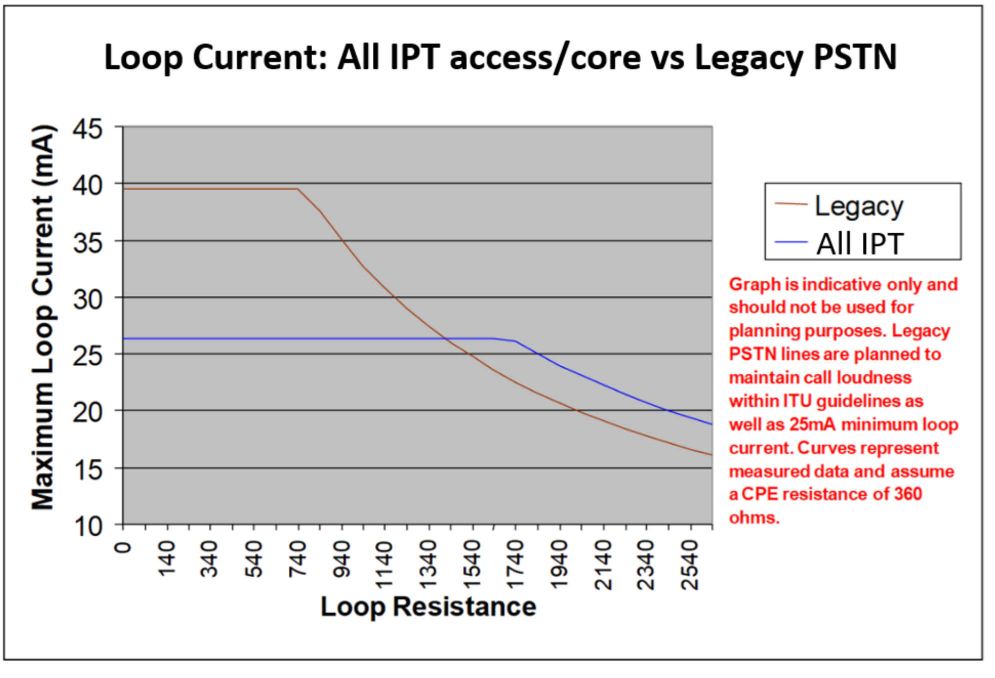
Loop current is the current that flows when CPE goes off hook, and is determined by the maximum voltage available from the line interface circuit, the current control implemented on the line interface circuit, the loop resistance of the copper line and the DC characteristics of the CPE when off hook.

UK traditional PSTN, System X and AXE 10, exchange line cards have constant current controls implemented which aim to deliver approximately 40 mA 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 40 mA 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 25 mA loop current.

The maximum loop current available on All-IPT services may replicate the characteristics of the UK traditional PSTN, but could also be limited to a lower figure of 25 mA, which is consistent with network design trends in Europe and elsewhere. An example of this is shown in the following graph. It should be noted that the change from 40 mA to 25 mA does not affect longer lines with higher loop resistance, but affects shorter lines with lower loop resistance.

Figure 7: Loop current comparison of 40 mA and 25 mA



Any CPE unable to operate correctly with 25 mA 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 All-IPT core/access network line itself, or on longer legacy voice lines of sufficiently high loop resistance such that only 25 mA loop current is available, which can be easily checked with a simple multimeter.

It is worth noting that where the use of an ATA and IP backhaul are deployed it is worth ensuring that the ATA supports the minimum 25 mA loop current. Higher current up to 40 mA may be considered in order to maximise CPE compatibility.

14.2 Balanced ringing

Most of the existing UK traditional 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 All-IPT networks 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 balanced ringing is, in fact, already used in parts of the UK traditional 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 All-IPT network 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 All-IPT network line.

14.3 On-hook voltage

The on-hook voltage of voice lines on the UK traditional PSTN network is approximately 50V, although a maximum of 70V is specified. CPE can incorporate over voltage detectors that trigger substantially below 70V. Any All-IPT network line that provides an on-hook voltage of significantly greater than 50V may trigger the over voltage alarm of such CPE.

14.4 CLI delivery performance

The successful operation of Calling Line Identity (CLI) on voice lines is dependent on both the network implementation of CLI being correct, and also of the relevant CPE being fully conformant with the relevant terminal equipment specifications (for example, ND1016 [31]).

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 All-IPT Access Network equipment.

15 2100 Hz and other VBD discrimination tones

15.1 The effect of VBD discrimination tones on All-IPT networks

Various VBD CPE discrimination tones may be recognised by All-IPT networks 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 15.2.
- 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 (ITU-T Rec. V.32 [25] and above) which contain their own built-in echo cancellers which might conflict with any network echo cancellers left enabled. See Section 15.3.
- Triggering adaptive jitter buffers to transition to fixed mode. See Section 14.4.
- Up-speeding the codec used to ITU-T Rec. G.711 [4] 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 15.6 for a comprehensive list of such tones that may be recognised by All-IPT networks.

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

15.2 The effect of plain 2100 Hz (and other VBD discrimination tones) on NLPs

Plain 2100 Hz tone was originally used by CPE to disable network echo suppression devices. Such echo suppression devices have mostly been replaced by ITU-T Rec. G.168 [2] 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 ITU-T Rec. G.168 [2] echo cancellers is optionally disabled by plain 2100 Hz tone but, if so, will re-enable after 100 ms to 400 ms of bidirectional silence (i.e. may re-enable at 100 ms - 400 ms; will re-enable at 400 ms or more). This is significant for ITU-T Rec. V.23 [10] modem applications, which although sometimes issue 2100 Hz, are half duplex and therefore allow the NLP to re-enable at the end of the first data burst.

ND1704 [1] mandates the use of echo cancellers that are compliant to ITU-T Rec. G.168 [2]. Note that removal of the NLP on detection of 2100 Hz tone is optional in ITU-T Rec. G.168 [2] and therefore may not be implemented in every network.

Other VBD discrimination tones may have the same affect on NLPs as plain 2100 Hz, though this behaviour is not standardised. See Section 15.6 for a more complete list of VBD discrimination tones.

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

2100 Hz tone with phase reversals is used by CPE to fully disable network ITU-T Rec. G.168 [2] echo cancellers. This is used by any modem which includes its own echo canceller function, specifically ITU-T Rec. V.32 [25], V.32bis [26], V.34 [11], V.90 [12] and V.92 [13] 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 ND1704 [1] by reference to ITU-T Rec. G.168 [2].

Other VBD discrimination tones may have the same affect on NLPs and ECs as 2100 Hz with phase reversals, though this behaviour is not standardised. See Section 15.6 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 Rec. V.8 [27].

15.4 The effect of 2100 Hz (and other VBD discrimination tones) on adaptive jitter buffers

The behaviour of jitter buffers when encountering fax or dial-up modems is mandated in ND1704 [1], which states that fixed de-jitter buffers shall be used.

Therefore, with ND1704 [1] compliant All-IPT networks, when 2100 Hz tone (with or without phase reversals) is recognised, the jitter buffers on a call will transition to be fixed. This will increase the RTD by the difference between the initial length of the jitter buffer on the call and the value chosen for the fixed mode of the buffer, as well as the number of jitter buffers on a call.

The number of jitter buffers on a call carried over a single All-IPT network, or any All-IPT network interconnected via IP, will be one in each direction of the call (one at each edge of the All-IPT networks domain), i.e. two in the round trip. However, it is possible for calls to be routed via IPT core-TDM-IPT core or IPT access-IPT core-TDM-IPT core-IPT access, 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 All-IPT network case when all the jitter buffers become fixed.

Other VBD discrimination tones may have the same effect on jitter buffers as 2100 Hz, though this behaviour is not standardised. See Section 15.6 for a fuller list of VBD discrimination tones.

15.5 The effect of duration

2100 Hz tone, and potentially other VBD discrimination tones, occurring in either direction of a call on an All-IPT Core or Access network, will affect jitter buffers, ECs and NLPs with differing durations. These are summarised in Table 3 below.

Table 3: The effect of duration

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

15.6 List of VBD discrimination tones

Whilst 2100 Hz tone, with or without phase reversals, is the most commonly encountered VBD CPE discrimination tone, there are many others which may be detected by All-IPT edge equipment (serving node, MSANs, ATAs and Media Gateways) and trigger the echo canceller, NLP and jitter buffer behaviour described in sections 15.1 to 15.5. A more complete set of VBD discrimination tones is listed in Table 4 below, though this is not exhaustive.

It is important to note that whether a particular VBD tone is detected by a particular manufacturer’s serving node, 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 2100 Hz tone.

Table 4: VBD Discrimination Tones

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	device dependent	device dependent	device dependent
V.18 Annex A (Baudot) text telephone tones	1400 Hz followed by 1800 Hz or vice versa	device dependent	device dependent	device dependent
V.21 T.30 fax pre-amble flag sequence	(0x7E where 0 = 1850 Hz and 1 = 1650 Hz).	device dependent	device dependent	device dependent
V.21 low channel 1 mark (response) tone	980 Hz	device dependent	device dependent	device dependent
V.21 low channel 1 space tone	1180 Hz	device dependent	device dependent	device dependent
V.21 high channel 2 mark tone	1650 Hz	device dependent	device dependent	device dependent
V.22, V.22 bis and Bell 212	2250 Hz	device dependent	device dependent	device dependent
V.23 FSK tones	Mark and Space frequency transitions	device dependent	device dependent	device dependent
V.23 forward channel mark tone, also non-speech calling tone	1300 Hz	device dependent	device dependent	device dependent
V.23 backward channel mark tone	390 Hz	device dependent	device dependent	device dependent
V.32	1200 Hz followed by 1800 Hz carrier	device dependent	device dependent	device dependent
V.8 bis initiation tones (ESi)	1375 Hz / 2002 Hz	device dependent	device dependent	device dependent
V.8 bis response tones (ESr)	1529 Hz / 2225 Hz	device dependent	device dependent	device dependent
Bell 103 answer tone	2225 Hz	device dependent	device dependent	device dependent
Bell 103 originating tone	1270 Hz	device dependent	device dependent	device dependent
Bell 202 mark tone	1200 Hz	device dependent	device dependent	device dependent
Bell 202 space tone	2200 Hz	device dependent	device dependent	device dependent
EDT (Same as V.21 low channel)	980 Hz and 1180 Hz	device dependent	device dependent	device dependent

* See table in Section 15.5 for exact behaviour for 2100 Hz tone.

VBD discrimination tones may also be recognised by All-IPT networks as valid triggers to negotiate the use of ITU-T Rec. G.711 [4] if other codecs are used by default. Such codec negotiation is common but is not standardised.

15.7 Testing All-IPT networks for VBD discrimination tone detection

If the set of VBD discrimination tones recognised by a given All-IPT edge device, along with their subsequent triggered behaviours, is not clearly specified, it may be necessary to test the relevant Serving Node, 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).

15.7.1 Testing for echo canceller disabling

Test equipment is available that can measure the level of single ended 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 100 ms of this VBD tone (see Section 15.5 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.

15.7.2 Testing for NLP disabling

For NLP testing, the same type of equipment and test setup should be used as described in Section 15.7.1.

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.

15.7.3 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 All-IPT edge devices to transition to fixed mode. Note that on an All-IPT to traditional

PSTN call there will still be two jitter buffers involved - one at the All-IPT edge (e.g. MSAN or ATA) and one at the Media Gateway interconnect to the TDM-based network.

16 CPE testing guidance

This section describes the full set of test scenarios that can be undertaken for VBD 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 16.1 to 16.3 below.

- Sensitivity to end-to-end delay
- Sensitivity to Jitter Buffer Adaptations
- Functional test under all appropriate combinations of:
  - Call routes
  - Line losses (where applicable)
  - Communications direction
  - Supported functions

It should be noted that many of the tests rely on TDM-based test equipment (which is expected to be phased out or discontinued) and access to traditional PSTN lines. Therefore, the ability for CPE Manufacturers and Service Providers to adequately test and validate the performance of VBD, or diagnose issues affecting the operation of VBD, is expected to get increasingly difficult. Providers should instead seek to develop native IP-based solutions using well-established technologies and protocols for data transmission (such as client-server HTTPS).

Voice CPE
The following list summarises the tests that should be undertaken for Voice CPE, described in more detail in Section 16.4.

- Dialling out
- Ringing and call answer
- Voice quality
- CLI
- SMS (where applicable)
- Other applicable functions

16.1 Testing VBD CPE for sensitivity to end-to-end delay

Information about end-to-end delay on All-IPT networks, and its potential impact on VBD CPE is contained in Section 7.

The test for sensitivity to end-to-end delay does not require an All-IPT network. This is because the effect of end-to-end delay is the same, whatever the underlying platform. TDM-based test equipment is strongly advised in order to avoid introducing IP degradations that would potentially confuse the cause of any failures.

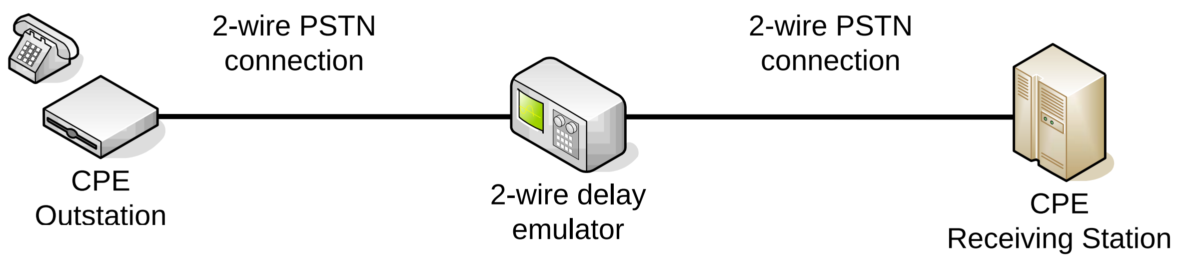
Legacy test equipment is therefore required to increase end-to-end delay in test calls. This type of test can be set up in a number of ways, three of which are described in Sections 16.1.1 to 16.1.3 (all require access to a traditional PSTN line to avoid introducing IP network degradations that could affect the outcome).

The impact of line loss on the sensitivity of VBD CPE to end-to-end 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 16.3.

16.1.1 Stand alone delay emulator

This method can be used when both ends of the VBD CPE application are 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.

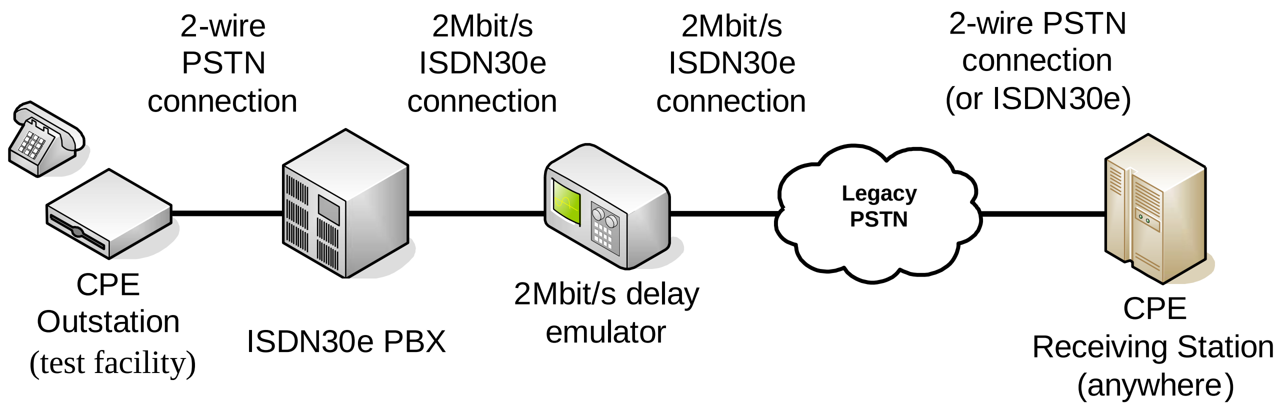
Figure 8: Stand alone delay emulator



16.1.2 Dial-through delay emulator

This method can be used where the CPE outstation (e.g. card payment terminal etc.) is available at the test facility location, but the CPE central station (e.g. 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 traditional 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 16.1.4, on a test call using the test rig.

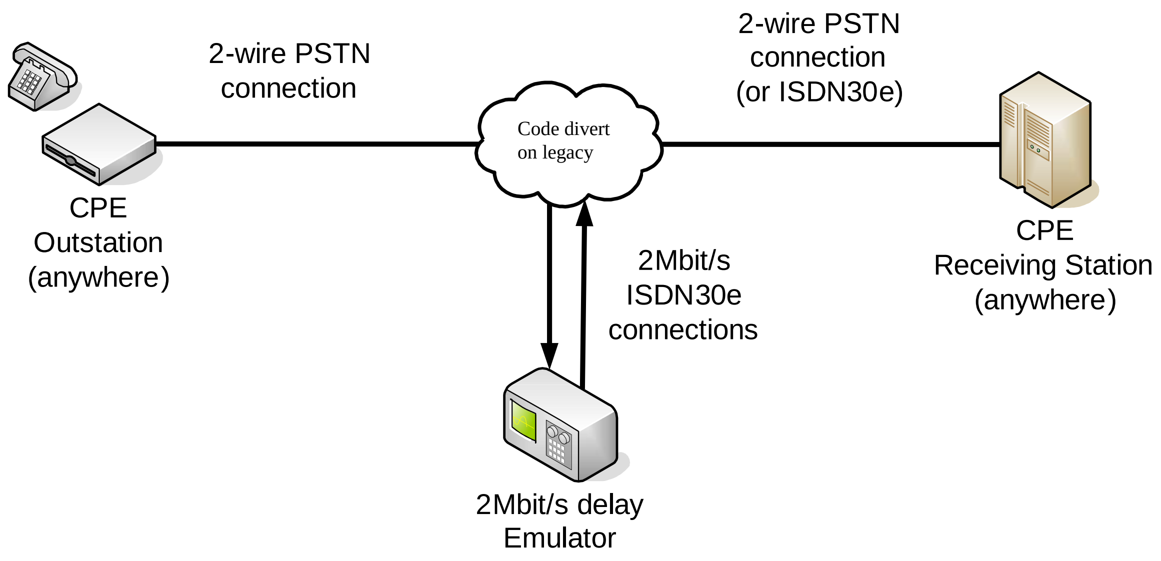
Figure 9: Dial-through delay emulator



16.1.3 Remote dial-up delay code

This method can be used where neither the CPE outstation (e.g. card payment terminal etc.) nor the CPE central station (e.g. the card payment server into which the card payment terminal dials) are available at the test facility location. An ‘Indirect Access’ 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 traditional 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 16.1.4, on a test call using the code.

Figure 10: Remote dial-up delay code



16.1.4 Measuring end-to-end 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 2100 Hz tone, which will cause all ITU-T Rec. G.168 [2] 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 2100 Hz 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.

16.2 Testing VBD CPE for sensitivity to Jitter Buffer Adaptations

Information about Jitter Buffer Adaptations on All-IPT networks, and the potential impact on VBD CPE, is contained in Section 8.

One approach to testing the impact of Jitter Buffer Adaptations on various types of CPE is to use an IP network emulator to apply increased levels of jitter to the backhaul of a serving node, MSAN or ATA. The jitter levels used could then be varied to mimic the expected real world jitter profile in order to trigger the jitter buffers at each end point to adapt accordingly. However, this approach is very problematic due to:

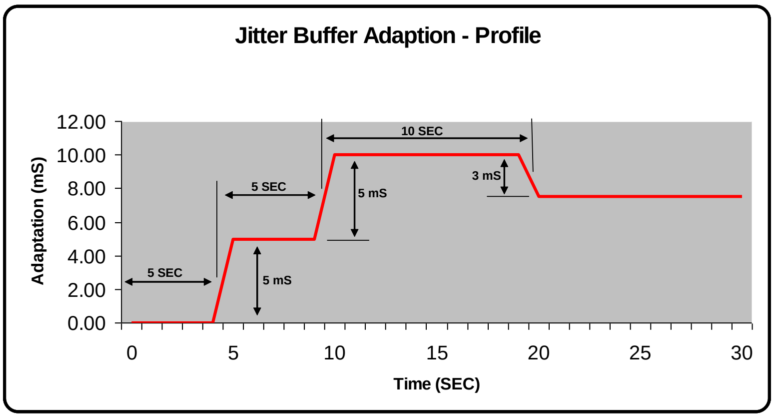
- The relatively long periodicity of even a sped up jitter profile compared to the very short duration nature of many of the test calls required.
- The difficulty of ensuring that a provoked Jitter Buffer Adaptation occurs at the precise moment required, i.e. at a particular point in a call lasting only a few seconds.
- The difficulty interpreting test results 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.

A better method is to emulate the effect of Jitter Buffer Adaptations on the transmission channel using a legacy TDM-based delay emulator. When a jitter buffer adapts up or down, it produces a step change in end-to-end delay, behaviour that can be accurately reproduced by changing the end-to-end delay applied using the delay emulator. This can be done at any point in a call, at any point in time.

This method requires a legacy TDM-based delay emulator to apply a sequence of upward and downward adaptations to represent a ‘reasonable worst-case’. 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 total level of jitter likely 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 in practice, the profile used should be the result of considering the above factors for the specific networks under test.

Figure 11: Jitter buffer adaptation profile



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 16.3.

16.3 Functional testing of VBD CPE

Functional testing of the VBD CPE should be undertaken on the All-IPT service under test in order to assess 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:
    - Traditional PSTN to traditional PSTN (baseline check)
    - Traditional PSTN to All-IPT
    - All-IPT Core/Access to traditional PSTN
    - All-IPT Core/Access to All-IPT Core/Access
- Line losses. Testing should be undertaken under all appropriate line losses:
    - Local exchange based voice lines: the UK access network exchange to customer range of 0 dB to 15 dB (see Section 16.5 for information on the distribution of line losses in the UK access network).
    - Street cabinet based voice lines: the UK access network cabinet to customer range of 0 dB to approximately 5 dB (note, this is a suggested estimate and could vary between operators).
    - Customer based voice lines: customer premises wiring range of 0 dB to perhaps 1 dB (note, this is a suggested estimate 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 and regression test may be appropriate, which tests the effect of the line disconnection occurring during any migration from legacy PSTN line to an All-IPT Core/Access service, 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.

16.4 Testing voice CPE

Functional testing of Voice CPE should be undertaken on the All-IPT service under test in order to confirm successful operation of 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 All-IPT Access Network 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 16.3
- ND1704 [1] voice quality metrics covering packet loss, end-to-end delay and variable delay metrics as defined in the ND1704 [1] Performance Calculator.
- All basic CPE functions, including:
    - Dialling out

      DTMF and loop-disconnect
    - Ringing

      Normal and any alternative distinctive ringing cadences
    - Voice quality
- Other relevant supported CPE functions, such as:
    - CLI

      Fully testing CLI requires a high number of call / CLI transmission attempts. Success rates need be determined to the nearest percentage point if small difference in performance is 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.

      In the UK both caller display protocols defined in ETSI EN 300 659-1 [28] are used between the analogue SLIC (Subscriber Line Interface Circuit) and the user's equipment.

      In an All-IP solution, the protocol to be used is a function of the ATA (likely selectable in the configuration). Implementors should ensure their ATAs are configured to the desired protocol.

      In the UK, most implementations use 'Data transmission prior to ringing'. Although this method is less widely used internationally, it does provide a better user experience than the alternative 'Data Transmission during ringing'.
    - SMS

      Similar considerations to the testing of CLI apply to the testing of SMS.
    - Answering machine ringing trip

- Answering machine line release
- Answering machine record/playback
- Memory
- Display operation
- Loudspeaking operation
- Payphone call length timing start

16.5 Line loss

This section only applies to All-IPT services that use the UK's copper access network to provide an analogue connection between the customer premises and local (such as network topologies with an MSAN edge device).

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, but the actual line loss / attenuation in dB is the most meaningful parameter to use (line length can equate to different line losses depending on the thickness of the copper pair).

16.5.1 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 15 dB at 1600 Hz. However, the large majority of lines are between 0 dB and 10 dB line loss at 1600 Hz. The following graph shows the distribution of calculated dB loss at 1600 Hz 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.

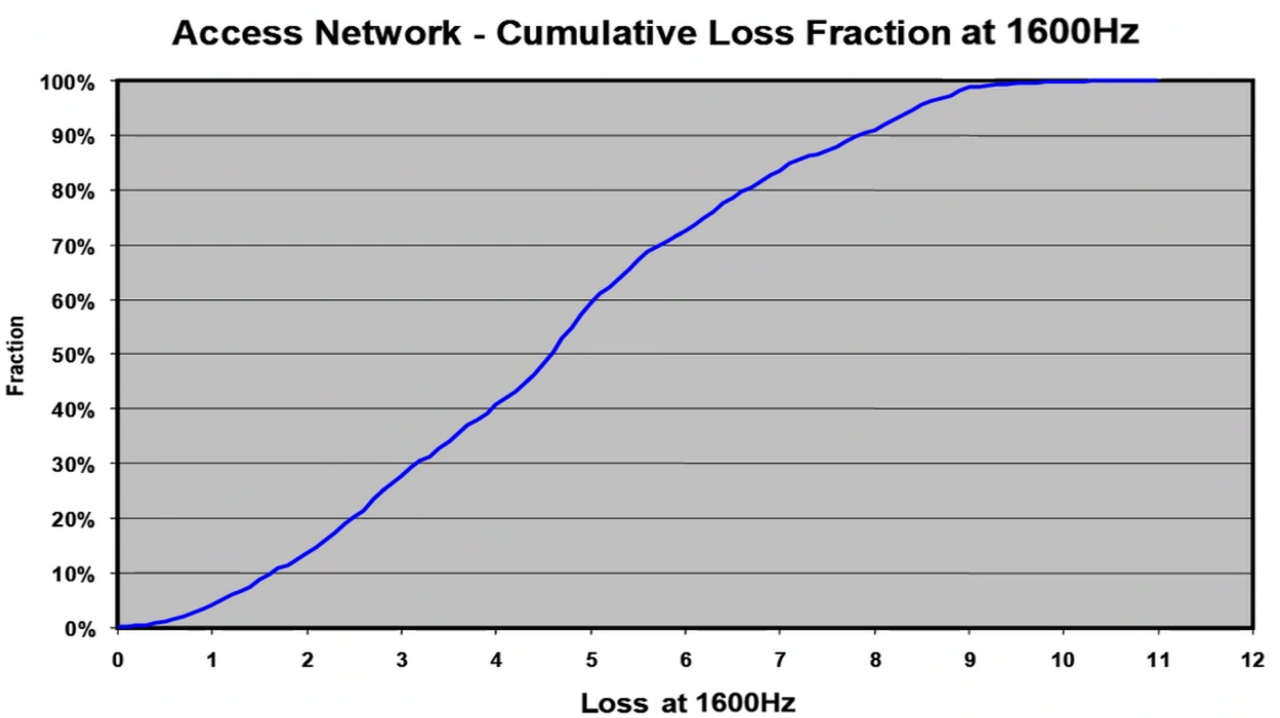


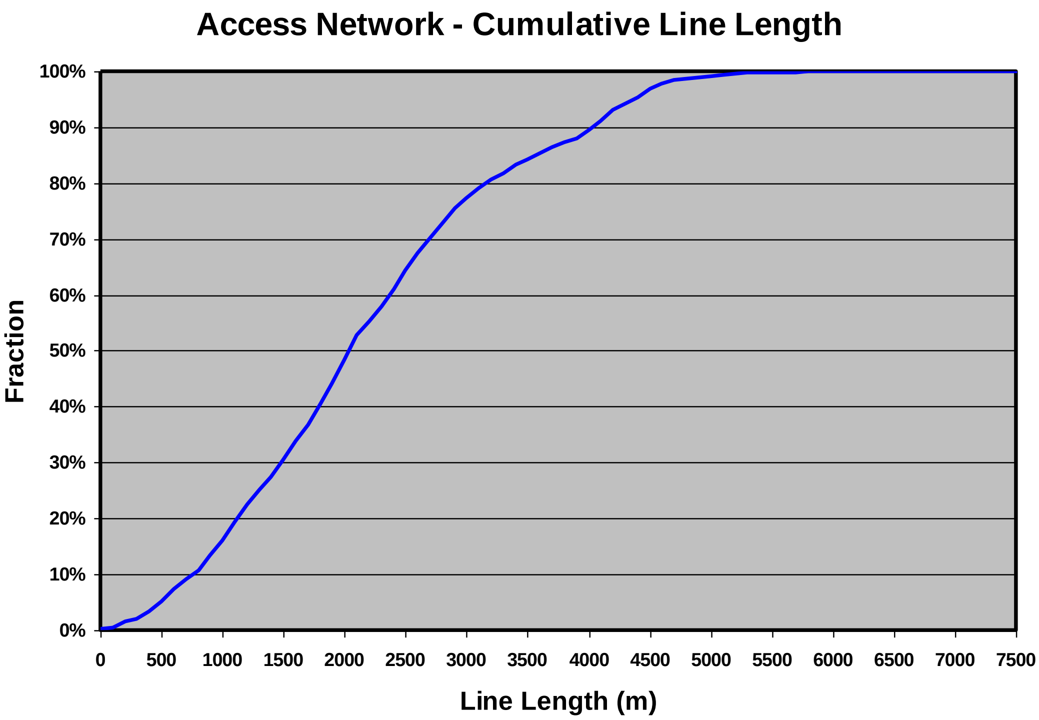
Figure 12: Access network cumulative loss at 1600 Hz

In this sample of lines, no line exceeds 15 dB at 1600 Hz line loss, and 99.8% of lines are actually within 10 dB at 1600 Hz 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.

16.5.2 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.

Figure 13: Access network cumulative line length



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 at 1600 Hz) nor DC loop resistance (ohms) can be simply calculated from line length. This is because 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.

16.5.3 Line loss, line card gain and end-to-end attenuation

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

The total end-to-end 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 end-to-end network attenuation. For the purposes of the figures presented here, the end-to-end internal network attenuation is effectively 12 dB. Hence the total end-to-end attenuation on a traditional PSTN connection should lie between:

- A minimum of (0 dB near end line loss – 0 dB near end line card gain + 12 dB internal network attenuation – 0 dB far end line card gain + 0 dB far end line loss) = 12 dB end-to-end attenuation

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

History

Document history		
Version	Date	Status
1.1.1	March 2011	Initial publication
2.1.1	May 2023	Second publication