

All-IP Telephony Industry Guidance and Lessons Learnt

NICC Standards Limited

c/o TWP ACCOUNTING LLP,
The Old Rectory,
Church Street,
Weybridge,
Surrey KT13 8DE

Tel.: +44(0) 20 7036 3636

Registered in England and Wales under number 6613589

NOTICE OF COPYRIGHT AND LIABILITY

© 2020 NICC Standards Limited

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be that printing on NICC printers of the PDF version kept on a specific network drive within the NICC.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other NICC documents is available at:

<http://www.niccstandards.org.uk/publications/>

The NICC Standards Web site contains the definitive information on the [IPR Policy and Anti-trust Compliance Policy](#)

If you have any comments concerning the accuracy of the contents of this document, please write to:

The Technical Secretary,
NICC Standards Ltd,
secretary@niccstandards.org.uk

Copyright

All right, title where and interest in this document are owned by NICC Standards Limited ("NICC") and/or the contributors to the document (unless otherwise indicated that copyright is owned or shared with a third party). Such title and interest is protected by United Kingdom copyright laws and international treaty provisions.

The contents of the document are believed to be accurate at the time of publishing, but no representation or warranty is given as to their accuracy, completeness or correctness. You may freely download, copy, store or distribute this document provided it is not modified in any way and it includes this copyright and liability statement.

You may not modify the contents of this document. You may produce a derived copyright work based on this document provided that you clearly indicate that it was created by yourself and that it was derived from this document and provided further that you ensure that any risk of confusion with this document is avoided.

Liability

Whilst every care has been taken in the preparation and publication of this document, neither NICC, nor any working group, committee, member, director, officer, agent, consultant or adviser of or to, or any person acting on behalf of NICC, nor any member of any such working group or committee, nor the companies, entities or organisations they represent, nor any other person contributing to the contents of this document (together the "Generators") accepts liability for any loss or damage whatsoever which may arise from the use of or reliance on the information contained in this document or from any errors or omissions, typographical or otherwise in the contents.

Nothing in this document constitutes advice. Nor does the transmission, downloading or sending of this document create any contractual relationship. In particular no licence is granted under any intellectual property right (including trade and service mark rights) save for the above licence to download copy, store and distribute this document and to produce derived copyright works.

The liability and responsibility for implementations based on this document rests with the implementer, and not with any of the Generators. If you implement any of the contents of this document, you agree to indemnify and hold harmless each Generator in any jurisdiction against any claims and legal proceedings alleging that the use of the contents by you or on your behalf infringes any legal or other right of any of the Generators or any third party.

None of the Generators accepts any liability whatsoever for any direct, indirect or consequential loss or damage arising in any way from any use of or reliance on the contents of this document for any purpose.

Contents

All-IP Telephony	1
Industry Guidance and Lessons Learnt	1
Intellectual Property Rights	4
Foreword.....	4
Introduction	4
1 Scope	5
2 References	5
3 Abbreviations	6
4.1 Pre-test Requirements	7
5 Lessons learned	10
5.1 General Lessons	10
5.2 TeleCare supplier/service sector	10
5.3 Alarm & security suppliers/service sector	11
5.4 EPOS & ATM Suppliers/Service Providers	12
6 Case Studies.....	14
6.1 Case Study: V.22/V.22 bis Call Failures	14
6.2 Case Study: DC Phone Line Characteristics	16
6.3 Case Study: Payment services sector number block move	17
7 Industry Solutions.....	18
7.1 Routeing / Differentiating of VBD and Voice capable calls.....	18
8 Frequently Asked Questions.....	18
8.1 Consumer FAQ.....	18
8.2 Communications Provider FAQ	18
History	20

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to NICC.

Pursuant to the [NICC IPR Policy](#), no investigation, including IPR searches, has been carried out by NICC. No guarantee can be given as to the existence of other IPRs which are, or may be, or may become, essential to the present document.

Foreword

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

Introduction

This document provides information to CPs and the wider industry on the lessons learnt during testing of and migration to All-IPT network solutions.

1 Scope

This document provides CPs with key scenarios to test when implementing voice over IP networks. This document also provides a repository for industry lessons learnt with particular attention to the migration from traditional TDM based voice services to Voice provided over an IP bearer.

This document is anticipated to be frequently updated in light of experience gained from the deployment of voice over IP networks.

2 References

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

2.1 Normative references

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

- [1] ND1704 – End to End Performance Rules
- [2] ND1704 – Performance Calculator
- [3] ND1444 – DTMF Best Practice Guide
- [4] ND1443 – Security Best Practice Guide
- [5] ND1431 – Guidance on CPE Compatibility

3 Abbreviations

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

ALL-IPT	-	All IP Telephony
ARC	-	Alarm Receiving Centre
ATA	-	Analogue Terminal Adapter
ATM	-	Automatic Teller Machine
BB	-	BroadBand
BS	-	British Standard
BSIA	-	British Security Industry Association
CP	-	Communications Provider
CPE	-	Customer Premises Equipment
DCMS	-	Department for Digital, Culture, Media and Sport
DJB	-	De Jitter Buffer
DTMF	-	Dual Tone Multi Frequency
E2E	-	End to End
ENISA	-	European Network and Information Security Association
EPOS	-	Electronic Point Of Sale
FTTC	-	Fibre To The Cabinet
FTTP	-	Fibre To The Premises
IOT	-	Internet of Things
IP	-	Internet Protocol
IPX	-	IP Exchange
ITU	-	International Telecommunications Union
NGA	-	Next Generation Access
NGN	-	Next Generation Network
NLP	-	Non Linear Processor
SIA	-	Security Industry Association signalling format protocol
SIN	-	Supplier Information Note
SIP	-	Session Initiation Protocol
SLIC	-	Subscriber's Line Interface Circuit
SOGEA	-	Signal Order Generic Ethernet Access
SOGFAST	-	Single Order GFAST
TDM	-	Time Division Multiplexing
TSA	-	TEC(Technology Enabled Care) Services Association
VBD	-	Voice Band Data
VoBB	-	Voice over BroadBand
xDSL	-	A variant of DSL (Digital Subscriber Line)

4 Test Guidance

This section documents some of the customer scenarios and test cases that should be considered when determining the suitability of a Voice over BB implementation.

4.1 Pre-test Requirements

The following aspects need to be addressed before testing can be started:–

- Understanding supplier and users' requirements from testing
- Identifying groups of devices with common characteristics and having a suitable subset available for test
- Expected outcomes defined and agreed
- Existing standards documentation should be referred to and their relevance understood, such as:-
 - International, e.g. ITU specification V.22bis
 - National standards, e.g. ND1431
 - CP specific documentation, e.g. BT SInS

4.2 Customer Scenarios and Test cases

A number of CPs offer free of charge test facilities for customers / vendors to test their equipment in an All-IPT network, including Virgin Media, Openreach and BT.

The following are the suggested customer scenarios and test cases from the BT's Test Lab facility that are currently available to industry.

Test	Customer Scenario	Test Case
0a	Benchmark supplier equipment validation test Supplier equipment connected to Narrowband PSTN connection on a standard xDSL Broadband enabled line. <i>(Repeat for various equipment types)</i>	Outbound call / connection to receiving centre over PSTN
		Inbound call / connection from receiving centre over PSTN
0b	Benchmark line validation demo - SOGEA BB performance and IP Voice service working <ul style="list-style-type: none"> • DSLab infinity lines speeds up to 80M/10M 	Register devices Speed test / downloads Basic voice calls and other tests
1a	Simulated conversion from PSTN to Consumer IP Voice service over SOGEA <ul style="list-style-type: none"> • 'Off-the-shelf' service at Aadastral Park • New Openreach faceplate • No PSTN 	Equipment response on withdrawal of dial tone / voltage on line
1b	Initial tests of equipment on IP Voice service <ul style="list-style-type: none"> • Using the ATA port on the Smart Hub 	Outbound call / connection to receiving centre over IP Voice
		Inbound call / connection from receiving centre over IP Voice
2	General use of supplier equipment connected to the ATA port using IP Voice service	Service response under normal Hub and normal line conditions
3a	Loaded Hub (busy home) simulation <ul style="list-style-type: none"> • Simulation of busy home (Hub CPU load), internet & Wi-Fi • Various conditions (L/M/H) 	Service response under loaded Hub conditions
3b	Loaded BB line (traffic) simulation <ul style="list-style-type: none"> • Simulation of loaded line (BB traffic load) • Various conditions (L/M/H) 	Service response under loaded BB line conditions

4	<p>Degraded BB line simulation</p> <ul style="list-style-type: none"> • Introduce packet loss • Introduce E2E delay • Various conditions (L/M/H) 	<p>Service response under degraded line conditions (Impairments and combinations of impairments to be agreed)</p>
5	<p>Use of the Voice Reinjection Solution</p> <ul style="list-style-type: none"> • extension wiring integration • Openreach faceplate and cable to ATA on hub 	<p>Service response and various calls / connections from/to equipment on an extension socket</p>
6	<p>Opportunity to consider supplier 'Native IP' equipment solutions over IP Voice or standard FTTC/Infinity lines</p>	

5 Lessons learned

As well as general lessons learnt there are currently three service sectors that test data is available for. The following sections contain key points currently observed through the BT Test Labs. In addition to the data from the BT Labs, links to relevant trade associations' testing guidance and lessons learnt are included at the end of the relevant sub sections, where available.

5.1 General Lessons

Outside of the service sectors mentioned in later sections, there are CP regulatory requirements such as the Text Relay service. The legacy Text Relay service implementations utilise VBD. It has been found during testing that legacy Text Relay services behave as most other VBD services and will generally work. However, as with other VBD services, this cannot be guaranteed.

It should be noted that VBD cannot be supported on 4G and 5G mobile networks, as the codecs in use on these networks cannot support VBD. It is also worth noting that although codecs with 2G and 3G mobile networks claim to support VBD, in practical implementations these networks have been found to provide a very poor customer experience.

IMPORTANT NOTES

VBD service users should take care to check that as networks evolve to IP Networks that they test their equipment in network providers' labs to ensure the service still works.

To reduce the instances of service failure, interconnects as well as network resources should be dimensioned for extraordinary load.

Consideration must be given to the traffic profile assumptions made by manufactures and the actual traffic profile experienced on the network end to end for each call scenario. Instances have been noted when these profiles vary significantly and packet loss has been encountered.

5.2 TeleCare supplier/service sector

The following summarises the key learnt observations that have been seen through the current industry testing conducted at the BT ALL-IPT Test facility following 48 external organisations conducting testing:-

- Some suppliers already have Native IP based products which cope better with IP packet loss
- Power resilience is seen as an issue as existing end devices normally have back-up batteries that last in excess of 24 hours, whereas the requirement on CPs is to ensure at least 1 hour of battery backup for the network.
- DTMF based protocols are generally quite robust to impairment
- Most kit & protocols are reasonably robust to low/medium levels of packet loss
- Most kit & protocols are robust to significant packet delay:
 - Visits confirmed BS8521 protocol less resilient to larger delays (300/550/1100ms) and packet loss levels.
 - CPC protocol was seen to cope well with even larger delays

- The use of higher impairment levels to demonstrate that all devices *can* fail is proving useful to suppliers to understand at which level equipment will cease to function correctly
- Most suppliers are moving towards either native IP based products or full mobile SIM solutions
- Most ARCs are TDM based at the moment
- Protocol choice is most significant factor when comparing different Hardware implementations
- Telecare OK button press and call-centre call if no button press features
 - It was found that the technical solution for these features was very resilient to packet loss and delay
- Supplier testing with 3 h/w devices, 3 protocols and 3 different ARCs
 - ARC supplier and communications path (e.g. multiple CPs / non-geo / other factors) can have significant impact on performance under impairment conditions
 - CPC protocol performed well with extreme delays and very high packet loss!
- Amplified Phones basically work (they have no 'special service' protocols)
 - Amplified Phones Testing Day with Ofcom and NADP held
- Some current PSTN/network issues already being seen
 - It is not unusual to experience repeat calls being required often due to exchange updates or other Service Providers

The TSA (TEC.Technology, enabled care, Services Association) is the recognised trade body for the TeleCare sector with over 350 organisation members and have been active in the space of voice and VBD testing, producing a test specification for their market sector, and is available from the following location:- <https://www.tsa-voice.org.uk/>

5.3 Alarm & security suppliers/service sector

The following summaries the key learnt observations that have been seen through the current industry testing conducted at the BT ALL-IPT Test facility following 48 external organisations conducting testing:-

- Most suppliers already have Native IP based products
- DTMF based protocols are generally quite robust to packet loss
 - Better than SIA
- V.22 bis modem protocol is not very resilient to packet loss
- Most hardware & protocols are robust to significant packet delay
 - DTMF not so good at very high delay levels, as timings are tighter; SIA is better
 - There is variance between manufacturers hardware
- The use of higher impairment levels to demonstrate that all devices *can* fail. This is proving useful to suppliers to understand at which level equipment will cease to function correctly
- Majority of suppliers are moving towards Native IP based products

- Some ARCs are both TDM and IP capable at the moment
- Some current PSTN/network issues already being seen
 - It is not unusual to experience repeat calls being required often due to exchange updates or IP Service Providers being present in the chain of delivery
- Local PSTN long lines issue seen in the BT Digital Services Lab
 - PSTN long lines have been seen to affect analogue PSTN benchmarking tests
- Concerns about battery back-up under power cut conditions
 - Alarm providers generally use either 12 or 24 hour back up for their alarms depending on whether it is a fire or security alarm
- Concerns about local line disconnections on migration
 - Security/fire system methods of connectivity may differ by location/premises and may be left disconnected dependent on the Hub installed and/or connectivity via existing extension wiring
 - Alarm installer visit is likely to be required following migration

The BSIA (British Security Industry Association) is the recognised trade association for the alarm and security sector and have been very active in the space of voice and VBD testing producing two test specification for their market sector. The test specifications have been published and are available from the following location:-

<https://www.bsia.co.uk/bsia-front/pdfs/bsia-test-plans1.zip>

5.4 EPOS & ATM Suppliers/Service Providers

The following summaries the key learnt observations that have been seen through the current industry testing conducted at the BT ALL-IPT Test facility.

- Older EPOS terminals can already be configured to PSTN or native IP as primary. If moving to All-IPT, they would benefit from being configured to use a native IP solution and avoid using the TDM voice network.
- EPOS suppliers have tested with BT Consumer's Digital Voice Service and the BT Enterprise Business ATA
 - Testing with BT Digital Voice lines, EPOS devices "generally work" but are not very resilient under impairment conditions
 - Typically, older modem protocol (e.g. V32) are not resilient to packet loss
 - Longer term downloads only successful on Clean IP Voice lines
- Latest EPOS models can switch between GPRS/Wi-fi/Bluetooth/PSTN
 - Hence EPOS industry may migrate to native IP connectivity to avoid potential problems with ALL-IPT
- ATM supplier tested primarily with Business ATA/IP Voice lines
 - Testing with Business IP Voice lines, ATM devices "generally work" but are not very resilient under impairment conditions
 - Typically, older modem protocol (e.g. V22) not resilient to any impairments
- Newer ATM machines work over IP

- Concern that several thousand machines exist and are in use, which do not support native IP or mobile connectivity.
- PCI Compliance
 - PCI compliance is a matter for the Service Provider (for example, the Acquirer / payments provider, i.e. Barclaycard) and the Customer (Merchant) buying the payment product set
 - Therefore, a PoS terminal Service is between the Merchant of the shop and the provider of the Point of Sale (PoS) service. If the Merchant then connects their PoS terminal to a router/hub and it supports no other service there is no issue with PCI compliance as it is dedicated for that service. If the hub supports multiple services it effectively becomes a 'shared' device and the Merchant would have to take steps to segregate the PoS terminal
 - In the instance that there is one PoS terminal for a corner shop, the onus will be on the Merchant to ensure / self-certify compliance. As networks and the number of terminals becomes more complex (the service taken by the Merchant) there will be varying degrees of compliance which the Service Provider will advise on given the product

6 Case Studies

In addition to the case studies summarised below, users and operators are encouraged to read ND1431 Guidance on CPE Compatibility on NGNs and NGAs [5] as it includes valuable detail regarding the sensitivity of CPE (in particular Voice Band Data CPE) to NGAs/NGNs. The case studies below are likely to serve as a practical illustration of the types of issues raised in ND1431.

6.1 Case Study: V.22/V.22 bis Call Failures

Description:

A leading provider of end-user devices obtained evidence of an increase in the number of Voice Band modem transaction failures, as calls were routed via the BT IPX platform (and as such, subject to TDM/IP conversion). Such issues were not evident on calls routed only via TDM infrastructure.

It is understood that end-user devices were connected to PSTN lines, with TDM to IP and IP to TDM conversions occurring on IPX media gateways.

Status:

Investigation showed that the provider had taken steps to adapt communication between modem and terminal. This included reducing the duration of ITU-T Rec. V.25 2100 Hz ANS tone to ~100 ms to minimise call duration, which is below the recommended value of 3.3 seconds (+/- 700 ms) according to ITU-T Rec. V.25, section 4.3. Once this was increased in line with the ITU-T Rec. V.25 recommendation, errors and call failures dropped to an acceptable level.

Although the fault was cleared, users and operators should be aware of the technical detail below.

Explanation:

Upon detection of ITU-T Rec. V.25 ANS tone, it is common for TDM/IP Media Gateways (MGWs) to transition to a state referred to as VBD (Voice Band Data) mode to minimise potential harm to data carried in audio signals. This includes some, or all, of the following:

- Renegotiation of codec (on the IP 'leg') to G.711 if the current session is not using G.711 (including appropriate choice of A-Law or μ -law with Voice Activity Detection disabled): G.711 is normally required for reliable conveyance of VBD (as specified in ND1704 for VBD Capable services).

Note, TDM/IP gateways may also switch from DTMF relay to in-band (although this is not applicable to V.22bis). It is also assumed VBD relay was not in use in this case.

- Disablement of the NLP (Non-linear Processor) component of the MGW echo canceller (if used) to avoid signal clipping at the onset of VBD signals. ITU-T Rec. G.168 compliant echo cancellers are designed to ensure such clipping does not adversely impact voice quality, but it is acknowledged that 'NLPs may affect the transmission of data through an enabled echo canceller'.

It is assumed that in this case ITU-T Rec. V.25 ANS tone without phase reversals was used where the NLP component is normally disabled until there is 100-400 ms of bidirectional silence.

High speed modems (e.g. V.32bis and above) that have their own echo cancellers use ITU-T Rec. V.25 ANS tone with periodic phase reversals to disable both linear echo cancellation and the NLP network components. This is not applicable in this case.

Note: The behaviour described above applies to network echo cancellers on traditional switched networks as well as IP/TDM MGW echo cancellers. However, the removal of the NLP component on detection of 2100 Hz tone is optional in G.168 and therefore may not be implemented even if G.168 compliance is stipulated.

- Transition of the IP packet de-jitter buffer (DJB) from an adaptive state to a fixed state (adaptations during a data call are likely to corrupt the data stream, whereas for voice signals they can normally take place in periods of silence without impacting user experience and at the same time allowing end-to-end delay to be minimised).

Depending on the configuration of the DJB prior to transitioning to a fixed state and what the fixed setting is, this will result in a gap (for an 'upward' transition to a longer buffer) or loss of signal (for a 'downward' transition to a shorter buffer).

This is required by ND1704 for VBD Capable services.

One explanation for the fault is that 100 ms ITU-T Rec. V.25 ANS tone duration is not long enough for any G.168 echo canceller to disable its NLP component. G.168 specifies a 1 second operation time to distinguish between ANS tone with and without phase reversals (to facilitate correct linear echo canceller disablement as discussed above) and to avoid accidental removal during speech or music signals that may contain 2100 Hz (or similar) frequency components.

This means that NLP was almost certainly not disabled. However, the same can be said for calls routed only via TDM, therefore this alone probably does not sufficiently explain the increase in failed calls over IPX (although it does mean the subsequent solution may have had an additional benefit in reducing call failures in TDM routing scenarios as well).

A more likely explanation is the behaviour of the IP de-jitter buffers and/or codec negotiation.

There is no formal specification for the required ANS tone duration before an IP/TDM gateway transitions to a fixed DJB and/or carries out any necessary codec renegotiation. It does not need to distinguish between ITU-T Rec. V.25 ANS tone with or without phase reversals, therefore the main technical requirements are to wait for a sufficiently long tone such that false triggering on speech or music is avoided, whilst ensuring it happens quickly enough to avoid corrupting critical handshaking or data transmission. It is possible 100 ms was too short, the de-jitter buffer was not fixed and/or codec was not renegotiated, with calls subsequently failing because of de-jitter buffer adaptations and/or codec distortion.

One further explanation is that the gateway did transition to a fixed DJB and did carry out any necessary codec renegotiation, however, because of the short ANS tone duration, this did not occur/complete before the transmission of critical data (e.g. during the transmission of unscrambled binary 1s). When the ANS tone duration was increased, any corruption due to the time taken to transition to a fixed buffer or to G.711 coding occurred whilst ANS tone was still being transmitted and subsequent data transmission was unaffected.

Note: performance related issues have been discounted in this case on the basis that the issue was resolved by extending the ITU-T Rec. V.25 ANS tone duration, which suggests most calls are free from packet loss and unaffected by any increase in end-to-end delay as a result of traversing an IP network(s).

Lessons Learnt

Overall, this case should serve to illustrate the complexity and scope for functional issues in supporting VBD over IP. It also exposes how a historical lack of standardisation leaves implementations open to interpretation; therefore, it is often hard to identify the exact root cause of an issue even when it has been satisfactorily resolved (in fact, there may be multiple causes).

It should also be noted that in this case there was a feasible solution (at least from a technical standpoint) since extending the ITU-T Rec. V.25 ANS tone duration gave a reasonable amount of

time for the associated network components to adapt to a state with the best possible chance of reliably conveying VBD. However, the only VBD discrimination tone for which behaviour is 'mostly' standardised is 2100 Hz tone. Some manufacturers may implement additional tone detectors, but there will inevitably be VBD calls for which IP/TDM gateways do not or cannot reliably transition to VBD mode before critical data transmission occurs. Unless the signal transmitted by the related CPE can be changed, there will be no technical solution for this.

6.2 Case Study: DC Phone Line Characteristics

Description:

For PSTN TDM based voice services, the Subscriber's Line Interface Circuit (SLIC) is located in a central site within part of the CPs network, providing the d.c. power for the CPE (telephony equipment) to operate. For ALL-IPT services the SLIC and associated power feed will be contained within the ATA at the customer's home.

Status:

Some ATAs are known to operate SLIC devices at the lower end of the d.c. line characteristics defined in the ETSI EN standards. i.e. they operate with a reduced On-Hook line voltage and/or reduced Off-Hook constant current.

Lessons Learnt - Voltage

- A number of CPEs make use of the On-hook -50Vd.c line voltage to trickle charge an internal battery (or capacitor). With a reduced On-Hook line voltage the trickle charge current will be reduced and thus limit the charging capability of the CPE.
 - Some CPEs store address book (and/or configuration) data in volatile memory which is permanently powered from the small internal rechargeable battery (or capacitor). Failure of the battery (or capacitor) to charge will result in address book (and/or configuration) data to be lost.
 - Some wholly line powered CPE use an internal battery to fully operate in the Off-Hook state. If this battery fails to charge during On-Hook periods, the CPE will fail to operate fully when Off-Hook.
- Upon receipt of an incoming call, the FSK modem signalling is transmitted to the CPE to convey the Caller Display information. This occurs during short periods whilst different balance impedances are applied to the line by the SLIC and CPE. Observations have shown that some CPE and ATAs/SLICs become incompatible, particularly when lower On-Hook line voltages are used, preventing the calling number to be displayed.
- Some CPE have an On-Hook line voltage detection circuit. This is used to draw attention to the owner that the CPE has been unplugged, or that there is a fault with the telephone line. For some CPE with a fixed voltage detection threshold, a lower voltage from an ATA/SLIC, may constantly trigger the warning from the CPE.

Lessons Learnt – Constant Line Current

- From experience CPs have observed that some CPE does not operate as expected with lower line currents, i.e. lower than 30mA. It is highly recommended that ATAs should be selected or configured to operate with constant line currents, i.e. >40mA, similar to those used on PSTN TDM networks

Note: operating at lower constant line currents has the effect of dissipating more power into the SLIC rather than into the CPE. As a result, the CPE has less available power to function. Operating at higher line currents has the opposite effect, that of lowering the power dissipated in the SLIC, and making more power available for the CPE.

- Some examples of issues observed when operating with lower constant line currents:

- Lower speech quality. The lower line current mimics the CPE working on a long line. It in turn increases its internal dynamic gains associated with the Send and Receive speech levels. Where these are not fully matched to the dynamic gains of the ATA/SLIC, speech quality can be compromised. This problem can also produce unexpected changes to the Sidetone levels, which again manifests as poor speech quality.
- Answering machines. Answering machines use a number of techniques to detect a call is in progress. One being to detect if line current is above a predetermined level. At the end of a message recording, failure to detect a reduction in line current can cause an answering machine to continue recording for an extended period.

Although the CPE is not in an Off-Hook state during the caller display FSK transmission period, there is a d.c. load resistance applied to the line, known as the wetting current. With lower line voltages, it has been observed that lower constant current feeds can also have an impact on the reception of caller display data.

6.3 Case Study: Payment services sector number block move

A CP moved an 0800 1k number block from terminating CP A to CP B at specific date and time to ensure rollback was available in case of traffic failure. The traffic consisted of high priority, short duration calls.

6.3.1 Change:

To re-route all CP B on-net originated 0800 X directly to the terminating point therefore not breaking out across the interconnect to CP A via TDM Interconnect. This change routed traffic directly across various direct SIP trunks into the terminating service provider. Therefore, taking both a TDM interconnection and BT transit (overflow) out of the routing.

6.3.2 Issue:

One originating customer, whose calls originated on an IP network transited CP B via SIP interconnect and routed out via their direct TDM interconnect had a 100% success rate before the change. After the change, the site had 100% failure rate.

6.3.3 Cause:

At that time, the originating CP configuration only supported G711u and G729. Prior to CP B building data to catch 08000706xxx to be captured on-net, this traffic was routed via the TDM interconnect – this call path negotiated G711u with the originating CP and the switch then carried out the SIP>ISUP interworking, resulting in an uncompressed audio path. Once CP B had implemented the routing to keep traffic on-net and point it directly at the terminating SIP trunks the problem emerged with the terminating Service Provider trunks did not support G711u – this resulted in the network choosing the common codec between ingress/egress which is G729 which in turn causes the final termination to fail due to latency. This was solved by the originating CP adding G711a into their IPIC configuration, allowing calls to set up in G711a and therefore an uncompressed audio path.

7 Industry Solutions

7.1 Routeing / Differentiating of VBD and Voice capable calls.

While TDM networks are in existence, there are interim solutions that CP's may employ to overcome the issue of preserving a TDM path for the successful transmission of legacy VBD. However, as TDM networks are retired in favour of IP networks, these solutions become impractical. Typical solutions may include:

- Routeing on number differentiation
- CPs maintaining a database of endpoints and routeing calls accordingly
- Using a prefix code from the CPE.

8 Frequently Asked Questions

8.1 Consumer FAQ

- Does this mean my phone connection will be over fibre?
 - Not necessarily. Your broadband may still be over copper but your phone **will** use internet protocols to make and receive phone calls over your broadband connection.
- Will my phone still work during a power outage?
 - IP Telephony requires a local power source to power service provider electronics. The facilities available to customers to maintain this power source in the event of a power cut will be CP dependent.
- Phone connections for critical services have an SLA of "fix within 4 hours". Will this still apply?
 - It is expected that critical broadband connections will be available from some (but not all) operators with a similar SLA but this will be CP dependent.
- How do you get a simple phone to make calls?
 - A number of CPs will be offering voice services over IP including the support of legacy CPE
- When will ISDN30 primary interconnects be withdrawn and what will replace them?
 - Most ISDN30 services are hosted from TDM exchanges which are planned for withdraw by 2025. ISDN30 can also be hosted from IP voice platforms and presented to the customer via IP/ISDN30 gateways (or ATAs). It maybe that some CPs may offer this as services beyond 2025.

8.2 Communications Provider FAQ

- What will happen if I am served by a poor performing fixed broadband connection.

- CP's may wish to offer a mobile alternative where a fixed line solution is no longer practical.
- What will be in place to assist customers that wish to continue to use existing legacy (Non IP Based) equipment?
 - This will be CP dependent; some may choose to offer a degree of legacy support through the use of ATA technology it is suggested that in the first instance that an approach to the relevant CP is made.
- Will the IP Voice calls over fibre be encrypted by default from day one?
 - For most CPs their All-IPT voice services will operate using private IP networks. Where private IP networks are used, voice service will be inaccessible from the 'public broadband internet'.
 - The requirement for encryption, although highly recommended in NICC standards and guidelines, will be a CP dependent feature. Following the NICC standards and guidelines, it is expected that all CPs will encrypt and / or segregate voice services to provide secure services

History

Document History		
Version	Date	Milestone
1.1.1	21 st November 2019	Initial publication
2.1.1	21 st December 2020	Second publication to include further information and uses cases, changes to the FAQ section, including new case study from the Payments Services sector, and various minor editorial changes.