Technical Document
PTC 228

Telecom User-Network Interface Specification: PBX SIP Trunking Service

DRAFT FOR PUBLIC COMMENT

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December 2009

TELECOM USER-NETWORK INTERFACE
PBX SIP TRUNKING
SERVICE

CONTENTS

FOREWORD

The purpose of this Specification is to supplement PTC 220 – “Requirements for Private Voice Networks connected to the PSTN/ISDN”, with requirements introduced when connecting private voice networks to the Telecom PBX SIP Trunking service.

The PBX SIP Trunking service User – Network interface includes the following components:

(a) VoIP (Voice over Internet Protocol) media path
(b) SIP (Session Initiation Protocol) signalling

This Specification covers requirements in these areas that are not specifically included in PTC 220. It describes the technical requirements for the control of voice services between a particular implementation of Telecom's network, known as “PBX SIP Trunking”, and a unit of user's equipment, ie an IP-capable PBX, connected via an IP bearer used to carry both VoIP speech and SIP signalling.

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Telecom shall not be liable for any loss (including consequential loss), damage or injury incurred by any person or organisation arising out of the sufficiency, accuracy, or utility of any such information or opinion.
The grant of a Telepermit for any item of terminal equipment indicates only that Telecom has accepted that the item complies with minimum conditions for connection to its network. It indicates no endorsement of the product by Telecom, nor does it provide any sort of warranty. Above all, it provides no assurance that any item will work correctly in all respects with another item of Telepermitted equipment of a different make or model, nor does it imply that any product is compatible with all of Telecom’s network services.

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1. Document Scope

This document covers the technical specification to be met by SIP-enabled IP-PBX Customer Premises Equipment in order to be approved for connection to Telecom's network, to enable direct IP peering between the private and public networks. This is necessary for a customer's IP-PBX to be connected to Telecom's network, so that the customer can subscribe to Telecom's PBX SIP Trunking service.

The document also covers the process to be followed to obtain approved for connection, together with details of tests required to be successfully conducted prior to such approval.

PBX SIP Trunking is a telephone access service for businesses. The calls are delivered using VoIP technology and use the SIP protocol (RFC 3261) for signalling between Telecom's network and customers' PBXs.
The PBX SIP Trunking service requires certain Telecom network equipment to be located in the customer's premises. This includes a Customer Located Network Equipment (CLNE) router, which is owned and managed by Telecom. The PBX SIP Trunking CLNE will be the service demarcation point and be connected on the network side to Telecom's Ethernet access network, and on the customer side will present a LAN interface used for connection to the customer's IP network (and thence to their IP-PBX).

The document defines the technical aspects of the user-network interface for the connection of IP-PBXs. It is intended for IP-PBX vendors, system integrators and users.

2. PBX SIP Trunking: PTC Process

2.1 Telepermit Principles

The same general principles will apply in obtaining approval for connection of CPE to the PBX SIP Trunking service as those outlined in PTC 220 sections 1 and 2, in particular:

(a) Telepermits (PTC 220 §1.2)
(b) Electromagnetic interference (PTC 220 §1.3)
(c) Legal requirements (PTC 220 §1.4)
(d) Mandatory requirements (PTC 220 §1.5)
(e) Exemptions from full compliance (PTC 220 §1.6)
(f) Warning to Suppliers (PTC 220 §1.7)
(g) Non-compliance aspects (PTC 220 §1.8)
(h) Warning notices (PTC 220 §1.9)
(i) Product and service compatibility (PTC 220 §1.10)
(j) Ongoing compliance (PTC 220 §1.11)

(k) Mode of presentation (PTC 220 §2.1)
(l) Marketing features (PTC 220 §2.4)
(m) Specialised services & features (PTC 220 §2.5)
(n) Variants of the same basic design (PTC 220 §2.6)
(o) Electrical safety (PTC 220 §2.7)
(p) Temperature (PTC 220 §2.8)

2.2 Interim Process

Due to the relative immaturity of standards used in the user-network interface for the PBX SIP Trunking service (eg compared with ISDN), there is a corresponding lack of standardised procedures and test facilities available yet for compliance testing. As a result it will be necessary to establish an interim process for PTC approval, pending such procedures and test facilities becoming available in future. The interim process is outlined in the following sections. (Note that this process may change without notice in future, as improvements are made and/or further test facilities become available.)

2.2.1 Self Evaluation

IP-PBX vendors applying for PTC approval shall firstly complete a suitability evaluation for any new IP-PBX that they wish to be connected to the PBX SIP Trunking service. This would be a “desk-based” check performed by the IP-PBX vendor or system integrator, and based on technical requirements described in this document and the relevant parts of PTC 220.

2.2.2 Test Facility

If the IP-PBX vendor/system integrator then believes that their IP-PBX is compatible, they would next approach Telecom for agreement to temporarily install their IP-PBX and conduct a PTC test in the Telecom NIL laboratory, in order to obtain Telecom NZ approval to use their IP-PBX with the PBX SIP Trunking service. The Telecom NIL laboratory is located in Tory St, Wellington, and contains an off-line implementation of the PBX SIP Trunking service suitable for such test purposes.

2.2.3 Conducting Tests

The compliance tests will be conducted by Telecom (or its nominated agent). Assistance will be sought from the IP-PBX vendor/system integrator in setting up the IP-PBX ready for testing, and as required during the testing. The IP-PBX vendor/system integrator shall document the IP-PBX configuration requirements expected to be needed for the compliance tests in the Telecom NZ NIL environment (these details shall be same as those intended to be later supplied to the end customer or installer). Except for normal customer provisioning and possible minor configuration changes on the PBX SIP Trunking CLNE, no Telecom network changes to the standard PBX SIP Trunking service configuration will be permitted. For the avoidance of doubt, the resolution of any interoperability issues will be the responsibility of the IP-PBX vendor/system integrator (ie not the responsibility of Telecom nor its nominated agent).

2.2.4 Certification

Following completion of compliance tests the test results will be reviewed by Telecom, and if satisfactory the IP-PBX would be granted a Telepermit in eth PTC228 series The agreed IP-PBX configuration used for the compliance tests, together with any PBX SIP Trunking CLNE minor configuration changes, would be documented in a “PTC 228/YY/XXX PBX SIP Trunking Interoperability Specification” for that particular IP-PBX (where "PTC 228/YY/XXX” would refer to the specific IP-PBX). This Inter operability Specification would form part of the PBX SIP Trunking certification and be re-used by all end customers using the same type of IP-PBX.

2.3 Compliance Testing

2.3.1 Test Procedures

(1) The test procedures required for granting a Telepermit for IP-PBXs covered by this Specification are detailed in this section and Appendices 1 and 2.

(2) The selection of tests applicable to the stated features of the product is the responsibility of Telecom (or its nominated agent). For this reason it is important that details of all available features be supplied by the applicant. If Telecom is not advised of all features and additional ones are discovered after a Telepermit has been granted, then it may be necessary for the product to be retested. Where there is any doubt, the matter should be discussed with Access Standards, preferably prior to commencement of testing.

(3) For a relatively complex service such as PBX SIP Trunking, difficulties may arise in performing the necessary tests. In such cases, the Applicant shall provide the necessary liaison to ensure that Telecom (or its nominated agent) has sufficient information to readily perform the tests. It is recommended that specific complexities be discussed with Access Standards in the first instance, particularly if the device incorporates any process such as adaptive self-adjustment of its parameters.

(4) Telecom (or its nominated agent), in conjunction with the Applicant, may be able to modify the IP-PBX or its configuration temporarily to meet the Telepermit requirements, but is under no obligation to do so. Full details of such modifications shall be included with the test report and furnished as part of the application.

(5) Tests and measurements shall be carried out with the equipment functioning normally, and the relevant requirements of this Specification shall be complied with under normal working conditions. Where appropriate, the following should also be made available:-

(a) Information on the most suitable means of disabling any automatic facilities during the test programme.
(b) Any equipment or software necessary for the initialisation of the device under test.

2.3.2 Test Plan

Tests shall be conducted in accordance with the test plan in Appendix 1 of this document. The test plan includes the following test types:

(a) Functional tests
(b) Non-Functional tests
(c) Protocol tests

The main emphasis is on Functional tests, in order to verify that the IP-PBX can handle the main types of calls expected to be encountered when used with the PBX SIP Trunking service (ie the aim is to test the user – network interface, rather than the IP-PBX itself). Some Non-Functional tests are included to cover areas such as Electrical Safety, Transmission performance, etc. Protocol tests are not covered in detail and are generally only expected to be needed when trouble-shooting problems with other tests.

2.3.3 Test Results

(1) Full test results in the form of a Test Report issued by Telecom (or its nominated agent), shall be provided in support of all Telepermit Applications.

(2) Such Test Reports shall be in a format as close as practicable to that given in the Test Plan shown in Appendix 1 of this Specification. All tests shall be addressed, stating "NA" where a test is not applicable.

(3) Photocopies of original Test Reports shall be accompanied by either colour photographs or colour photocopies of the product.

(4) All Test Results shall be relevant to the product in the form in which it will be offered for connection to the Telecom Network. It is not permitted to use different software and hardware settings simply to meet different clause requirements.

(5) In cases where equipment has been previously tested on behalf of another Telecommunications Authority, the resultant test results may be submitted to Telecom as additional support for a Telepermit application. Such test results must be relevant to the appropriate PTC Specification requirements for them to

be considered for the grant of a Telepermit. In general, other Authorities' compliance certificates alone are not sufficient for acceptance.

3. Definitions

3.1 General

(1) The basic elements of a private voice network are shown in Figure 3.1, which has been updated from the similar diagram in PTC 220 to include the FXD (VoIP) Digital Gateway trunk.

(2) For the avoidance of confusion over terms used in this Specification, they are defined as follows:-

AC: **A**rea **C**ode (1-digit)

CLIP: **C**alling **L**ine **I**dentity **P**resentation

CLIR: **C**alling **L**ine **I**dentity **R**estriction

CLNE: **C**ustomer **L**ocated **N**etwork **E**quipment (refer also section 4.3)

DN: **D**irectory **N**umber (7-digit)

FXD (VoIP): **F**oreign e**X**change central office **D**igital interface (**VoIP**). This equipment interfaces an IP transmission link to a digital VoIP port on a Telecom switch. (In PBX terminology this functionality would be contained in a VoIP trunk module.)

FXS: **F**oreign e**X**change **S**ubscriber interface . This equipment interfaces a transmission link to standard PTC 200 compliant CPE.

MOS: **M**ean **O**pinion **S**core

PBX: Private Branch Exchange (refer also section 4.3)

QoS: **Q**uality **o**f **S**ervice

RFC: **R**equest **F**or **C**omment

RTCP: **RT**P **C**ontrol **P**rotocol

RTP: **R**eal-time **T**ransport **P**rotocol

SDP: **(1)** **S**ervice **D**elivery **P**oint, defined as the Ethernet interface on the customer “side” of the PBX SIP Trunking CLNE.

(**2)** **S**ession **D**escription **P**rotocol, defined in RFC 2327.

SIP: **S**ession **I**nitiation **P**rotocol

TRP: **T**ransmission **R**eference **P**oint, also known as the 0 dBr point. Point within network where losses are measured from and to (ie similar usage to that in PTC 220).

VoIP: **V**oice **o**ver **I**nternet **P**rotocol

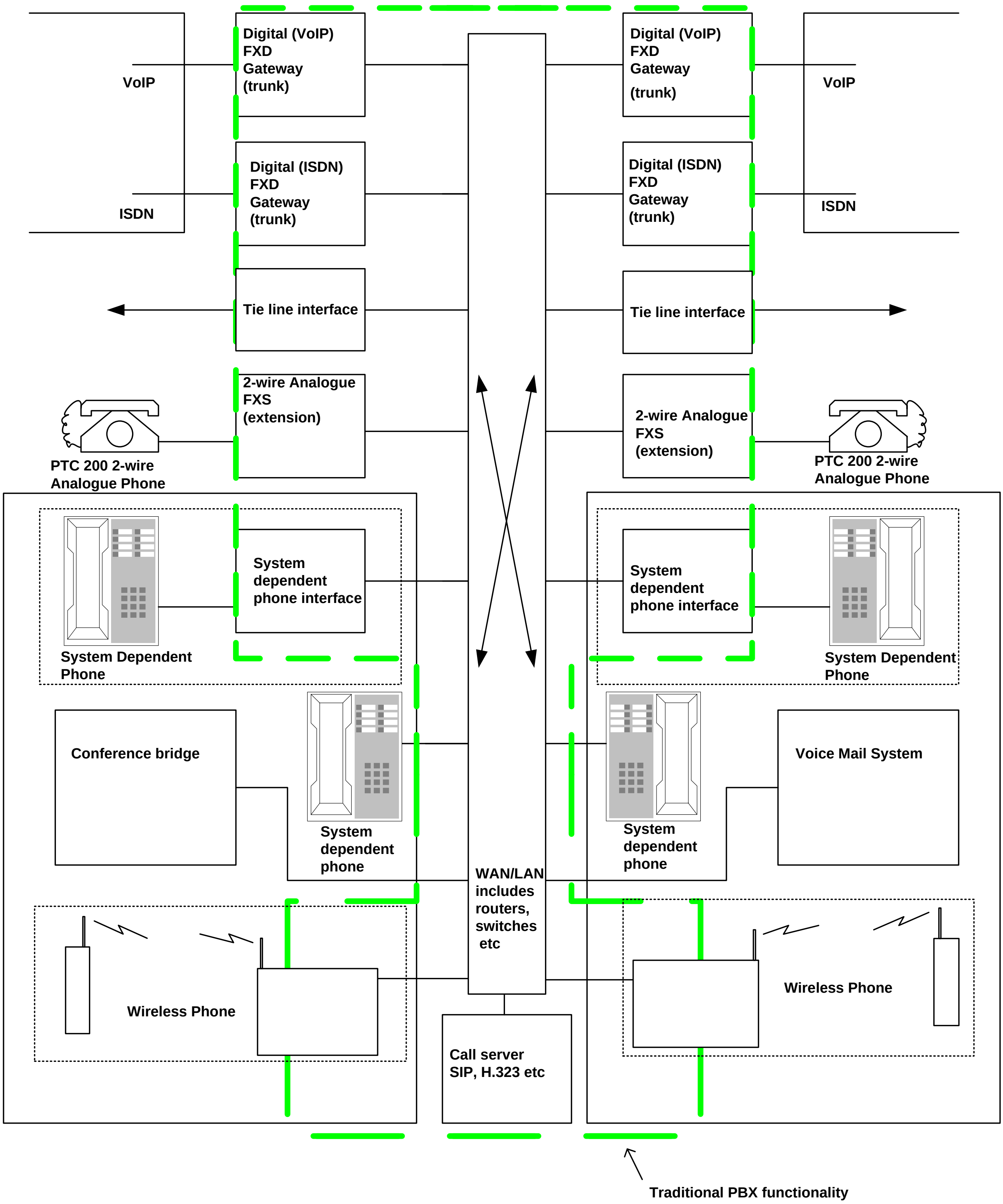


Fig 3.1 Elements of a private voice network

4. Technical Requirements

4.1 Introduction

This section outlines an interface specification that enables direct IP peering between Telecom's network and SIP-enabled IP PBX systems, for the purpose of originating and/or terminating audio (voice) and clear channel fax calls.

It specifies the minimum set of standards that must be supported, provides precise guidance in the areas where standards leave multiple implementation options, and specifies a minimum set of capabilities that must be supported by a SIP-enabled IP PBX system.

The specific areas where this section provides implementation guidance include:

(a) Specification of the minimum set of protocols that must be supported by the PBX systems
(b) Specification of the standards associated with these protocols that must or should be supported by the PBX systems
(c) Specification of formulation of protocol messages
(d) Specification of minimum QoS requirements
(e) Specification of minimum requirements for codec support, packetization intervals, and capability negotiation
(f) Specification of the method for handling echo cancellation
(g) Specification of the method for transporting DTMF tones
(h) Specification of the method for handling fax transmissions

The following key assumptions have been made with regards to this interface specification:

(i) The only service to be delivered over this interface is audio-based call origination and/or termination. The delivery of any other service (e.g. video-based services, instant messaging, etc) is out of scope.
(j) SIP (RFC 3261) is the only call control protocol supported between the PBX and Telecom's network (for the PBX SIP Trunking service).
(k) Media traffic may pass directly between the media endpoints (e.g. IP phones, ATAs) and Telecom's network, and not traverse the PBX. In this case, the same overall PBX requirements shall apply to the endpoints (except for signalling). It is the responsibility of the Telepermit holder to ensure that these endpoints have been thoroughly checked against the requirements of this specification.
(l) Signalling considerations between the PBX and endpoints within the customer network are outside the scope of this document.
(m) Signalling considerations within Telecom's network are outside the scope of this document.
(n) Modem support is outside the scope of this document (other than those used in fax transport).
(o) Layer 3 network design is outside of the scope of this document.
(p) Element management, network management, network security, and OSS considerations are outside the scope of this document.

4.2 Conventions and Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119.

4.3

4.4 Network Architecture

The following general network architecture applies to the PBX SIP Trunking service.

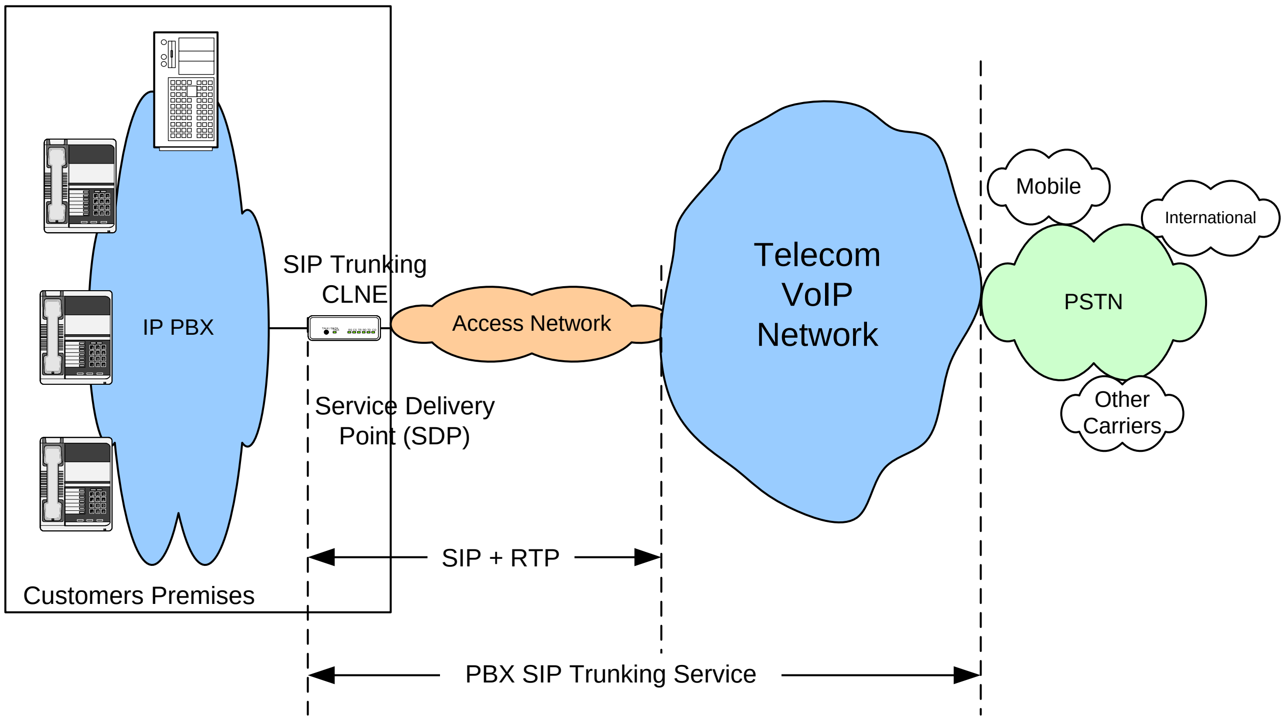


Figure 4.1 Network Architecture

IP PBX (PBX): The IP PBX constitutes a customer's collection of network elements that provides packetized voice call origination and termination services using SIP for signalling and RTP for media traffic.

PBX Extensions: IP/analogue phones connected to the PBX (refer section 4.4). On a call from a PBX extension without a dedicated telephone number (i.e. DDI number) the PBX may choose to assert its "main" identity (i.e. the pilot number), while a call from an extension with a dedicated DDI number could use the identity of that user's specific telephone number.

SIP Trunking CLNE (Customer Located Network Equipment): The CLNE is a device located at the customer's premises that provides the demarcation point between the Customer network and the Telecom network. It provides protocol-aware NAT at the Telecom and Customer network edges. It is owned and managed by Telecom NZ.

Telecom's VoIP Network: Telecom's VoIP network provides PSTN-like services to customers via IP infrastructure. It consists of IMS-based call control functions that provide SIP message routing; application servers that provide call origination/termination services to customers using SIP; signalling and trunking gateways that interface with the PSTN. It will be referred to as "the Telecom network" throughout this document.

4.5 Connection Models

The PBX SIP Trunking service will be delivered over a physical 100 BASE-T (IEEE 802.3) interface on the CLNE, allowing flexibility with the type of connectivity, e.g. to a separate network or the PBX directly.

4.5.1 CLNE directly connected to the Customer PBX

The PBX on the customer premises can be connected to the CLNE physically with the use of a Category 5e crossover cable.

In this scenario all signalling (SIP) and media (RTP) traffic passes between the CLNE and the customer PBX.

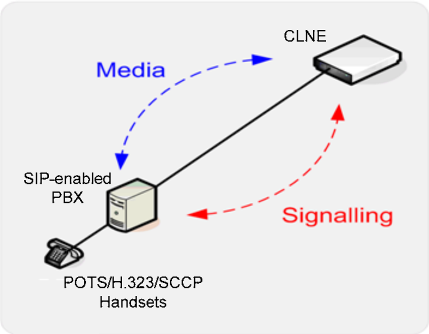


Figure 4.2 Directly connected CLNE

4.5.2 CLNE connected to Customer LAN/WAN

The PBX can be connected to the CLNE through the customer LAN/WAN (this is a more likely scenario).

Signalling (SIP) traffic always passes between the CLNE and the customer PBX.

Depending on the architecture of the PBX/customer network design:

(a) Media (RTP) traffic (and RTCP where it exists) may pass directly from the media endpoints (e.g. IP phones, ATAs and transcoding point) to the CLNE.
(b) The PBX may remain in the RTP (and RTCP where it exists) bearer path (e.g. where transcoding by PBX is required or a centralised deployment model is in place).

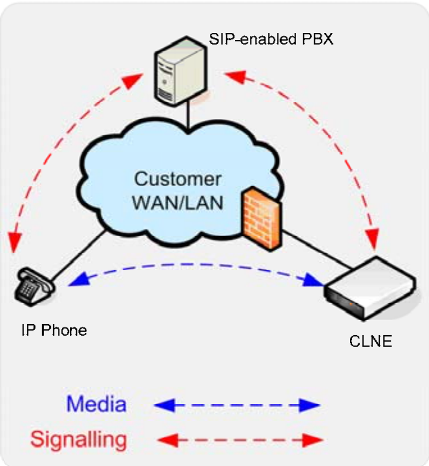


Figure 4.3 LAN/WAN connected CLNE – option a

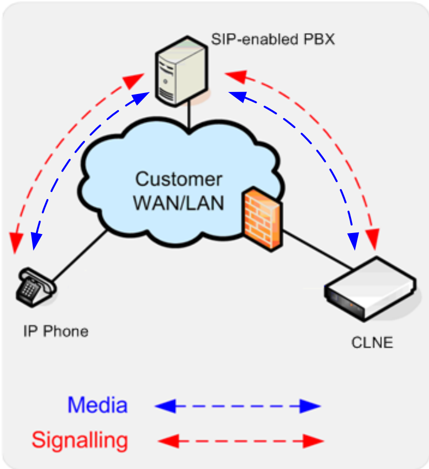


Figure 4.4 LAN/WAN connected CLNE – option b

4.6 Standards Support

The protocols used at the network interface shall conform to the following specifications.
(Comments on Telecom's specific implementation of these protocols are shown where applicable.)

Standard	Description	
RFC 791	Internet Protocol (IPv4) Note: IPv6 support will be required in future.	Mandatory
RFC 2327	SDP: Session Description Protocol	Mandatory
RFC 2833	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals - RFC 2833 is Telecom's preferred method for transporting DTMF tones. - The customer endpoints should be able to handle in-band if RFC 2833 is not supported (by the PBX and/or the far end).	Recommended
RFC 3261	SIP: Session Initiation Protocol - The transport methods supported are UDP (RFC 768) and TCP (RFC 793). - UDP is Telecom's preferred transport method. - The PBX-specific agreed transport method (ie UDP <u>or</u> TCP) shall apply in <u>both</u> directions between the PBX and the CLNE.	Mandatory
RFC 3264	An Offer/Answer Model with Session Description Protocol (SDP)	Mandatory
RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method	Recommended (at minimum to receive)
RFC 3550	RTP: A Transport Protocol for Real-Time Applications	Mandatory

4.7 Transmission

Transmission requirements for PBX SIP Trunking are generally similar to those outlined in PTC 220 sections 4 and 7.3, notwithstanding the change to VoIP transmission.

4.7.1 Loss Plan

(a) Loss from TRP to Network IP Trunk Interface: 0.5 dB to -0.5 dB,
Objective: 0 dB

(b) Loss from Network IP Trunk Interface: 0.5 dB to -0.5 dB
Objective: 0 dB

4.7.2 Attenuation Frequency Distortion

The loss distortion with frequency between the Network IP Trunk Interface and the TRP, and the TRP and the Network IP Trunk Interface shall be within the following limits, using an input level of -10 dBm0.

Frequency (Hz)	**Loss relative to the loss at 1000Hz (dB)**
300 - 400	+1, -0.3
400 - 600	+0.75, -0.3
600 - 2000	+0.35, -0.3
2000 - 2400	+0.45, -0.3
2400 - 3000	+0.7, -0.3
3000 - 3400	+1.7, -0.3

- *Reference ITU Recommendation Q.552*

4.7.3 Variation of Gain with Input Level

With a 1000 Hz sinewave signal at a level between -55 dBm0 and +3 dBm0, the gain of that signal between the Network IP Trunk Interface and the TRP, and between the TRP and Network IP Trunk Interface, relative to the gain of a signal at an input level of -10 dBm0, shall be within the following limits:

Input level (dBm0)	**Gain Variation (dB)**
-55 to -50	+/- 1.6
-50 to -40	+/- 0.6
-40 to +3	+/- 0.3

- *ITU-T Recommendation Q.552 clause 3.1.1.4*

4.7.4 Analogue FXS, VoIP/Digital FXD

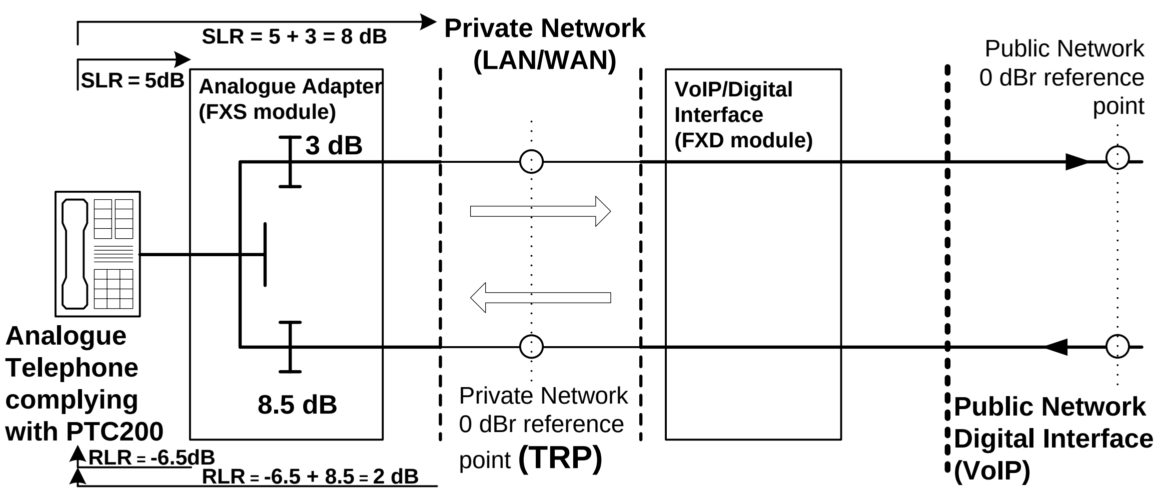


Figure 4.6.4 Standard PTC200 phone port to digital (VoIP) trunk

4.7.4.1 FXS Loss Plan

(a) Loss from FXS Analogue port to TRP: 2.5 to 3.5 dB, **Objective: 3 dB**
(b) Loss from TRP to FXS Analogue port: 8 to 9 dB, **Objective: 8.5 dB**

4.7.4.2 FXD Loss Plan

(a) Loss from TRP to FXD Digital (VoIP) trunk: 0.5 dB to -0.5 dB,
Objective: 0 dB
(b) Loss from FXD Digital (VoIP) trunk to TRP: 0.5 dB to -0.5 dB,
Objective 0 dB

- *The losses/gains shall be measured at 1000Hz*
- *The 3dB and 8.5dB losses shown in the FXS module each include 2.5 dB loss which represents the 2-wire analogue loss in the cable between the FXS module and the analogue phone. If this line is likely to be long in practice, the FXS PAD values may be reduced to 0.5dB and 6dB respectively.*
- *Note: -ve losses are gains*

4.7.5 Digital (or SIP/IP) Phone to digital (VoIP) trunk (FXD)

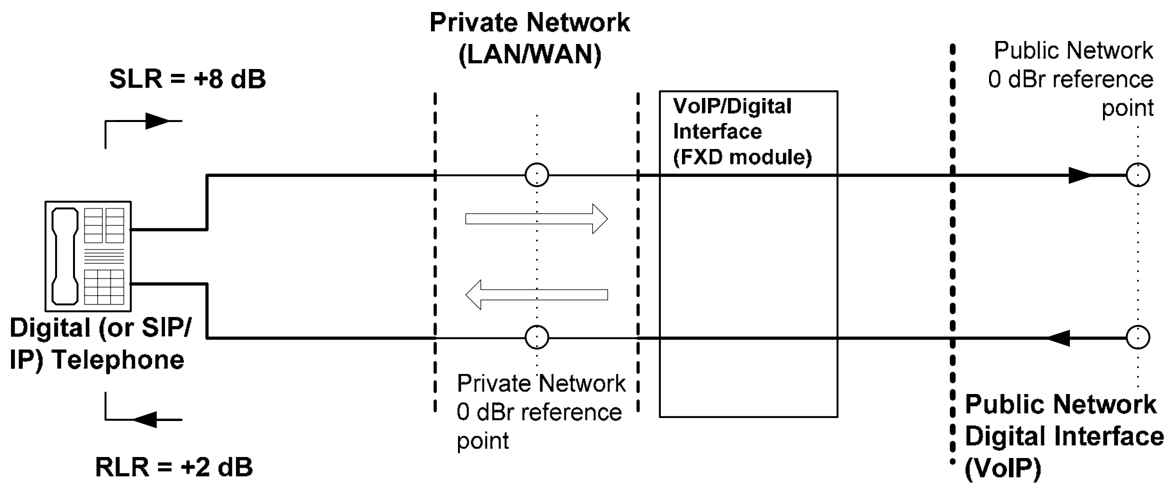


Figure 4.6.5 Digital (or SIP/IP) phone connected via digital (VoIP) FXD

4.7.5.1 Digital Phone Loudness Ratings

(a) Send Loudness Rating (SLR): +10 dB to +6 dB, **Objective: +8 dB**
(b) Receive Loudness Rating (RLR): +4 dB to 0 dB (-8 dB with volume control), **Objective: +2 dB**

4.7.5.2 FXD (VoIP)

(a) Loss from TRP to FXD Digital (VoIP) trunk: 0.5 dB to -0.5 dB,
Objective: 0 dB
(b) Loss from FXD Digital (VoIP) trunk to TRP: 0.5 dB to -0.5 dB,
Objective: 0 dB

- *Digital phone in this context is any phone other than a Telepermitted analogue phone complying with PTC200 which would connect to a Telepermitted FXS module.*
- *From a transmission point of view the digital phone may in fact be a system dependent analogue phone AND its associated FXS function. That is, the various gains/losses may be distributed between the phone and the FXS however the designer chooses, provided the SLR and RLR at the 0 dBr reference point are +8 dB and +2 dB respectively. It is assumed that the private network 0 dBr point is digital, although it could be analogue. However, it is unlikely that this would be the case.*

4.7.6 Private Voice Mail/ IVR System to Digital (VoIP) Network Interface

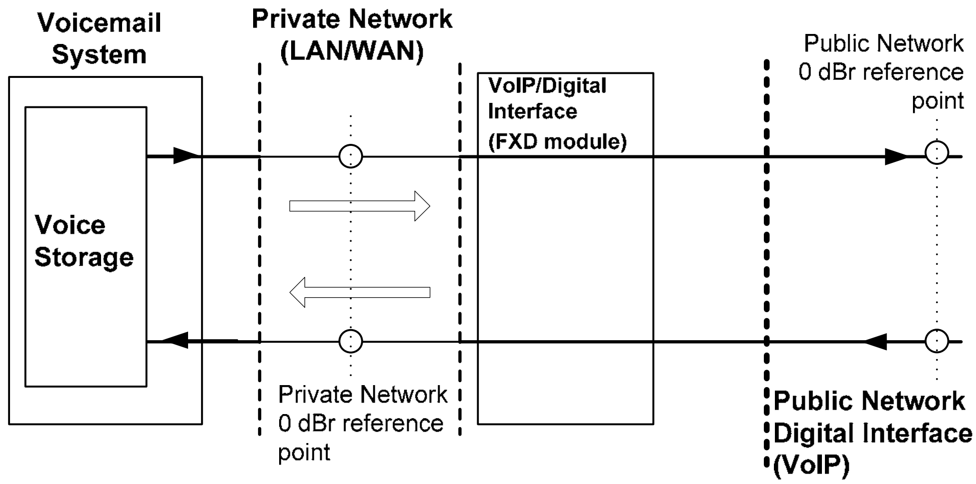


Figure 4.6.6 Private Voice mail system with digital (VoIP) network interface

Voice mail systems connected to private networks shall retransmit stored messages at the same level they were received at.

4.7.7 Tie Lines

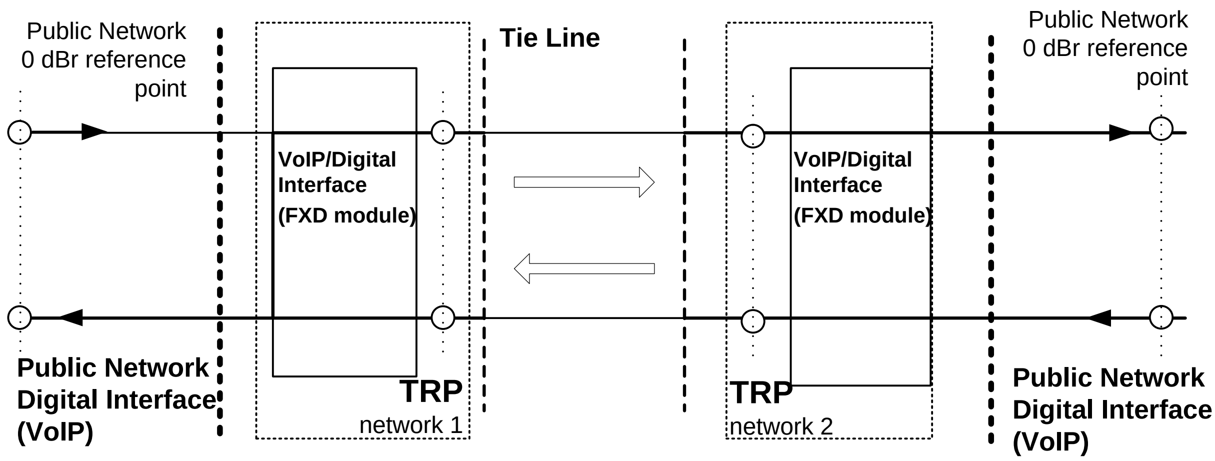


Figure 4.6.7 Two private networks connected by a tie line

Tie lines shall connect at the 0 dBr Transmission Reference Point (TRP) in each respective network.

Figure 4.6.7 shows two networks with digital (VoIP) FXD interfaces, connected by a tie line. The FXD interfaces both have 0 loss in the send and receive paths.

4.8 Quality of Service Considerations

4.8.1 Codec Support

(1) Voice samples shall be transported using the Real-time Transport Protocol (RTP) as described in RFC 3550.

(2) Any customer device that exchanges RTP traffic with the CLNE shall support ITU-T G.711 A-Law codec with a packetization rate of 20ms.

(3) The Telecom network supports G.711 A-law only; codecs offered by the PBX other than G.711 A-law will be discarded by the Telecom network.

4.8.2 Delay

The maximum delay acceptable between a PBX extension and the Service Delivery Point (SDP) is 50ms. (The PBX and extension telephone together shall contribute no greater than 40 ms of this delay, thus also allowing for a maximum of 10 ms delay due to switching/routing within the private network.)

4.8.3 Jitter Buffer

Adaptive jitter buffer control is required. The recommended dynamic range is 10 – 60ms.

Any customer device that supports pass-through transport of fax calls shall switch from any adaptive/dynamic jitter buffer in use to a fixed-length (100 ms) jitter buffer upon recognition of a 2100 Hz tone.

4.8.4 Echo Cancellation

ITU-T G.168 compliant echo cancellation shall be provided if analogue handsets are present. 64ms G.168 echo cancellers are required to eliminate any hybrid imbalance & handset conduction.

Any customer device that supports pass-through transport of fax calls shall switch off echo cancellers and non liner processors upon recognition of a 2100 Hz tone.

4.8.5 VAD and CNG

Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) shall be turned off.

4.8.6 PLC

It is recommended to turn on Packet Loss Concealment (PLC) algorithms.

4.8.7 QoS Recommendations

The following end-to-end QoS targets are shown for information. Where applicable the PBX and associated private network design should be consistent with these targets:

Type of Call	Max Jitter Avg	Max Packet loss %[1]	MOS Score Above[2]	R- value Above	ECHO 'TELR' Above	Max Post Dialling Delay
PBX to/from Telecom network	2ms	0.5	4.1	83	-55db	<2 Sec

Table 2 QoS SLA tables

Note 1: Packet Loss % is when using PLC ITU-T Rec. G.113

Note 2: MOS can be converted to R ("E" Model), see ITU-T Rec. G107, and G.109, and ASG0005.

4.8.8 RTCP QoS Monitoring

It is recommended that any customer device that originates and/or terminates RTP traffic supports RTCP (as described in RFC 3550).

QoS monitoring within the Telecom network is based on RTCP. For each call made, the Telecom network produces a Mean Opinion Score (MOS) equivalent rating derived from the RTCP information provided by the media endpoints.

4.9 Transport of DTMF Tones

Any customer device that exchanges RTP traffic with the CLNE shall support at least one of the two methods:

(a) Transport of DTMF tones using the RTP telephone-event payload format as described in RFC 2833 (which is Telecom's preferred method).
(b) Transport of DTMF tones in-band

4.10 Fax Support

(1) The Telecom network supports only the G.711 transparent pass-through mode for fax (ie T.38 fax relay is not supported).

(2) Any customer device that supports pass-through transport of fax calls shall upon recognition of a 2100 Hz tone:

(a) Switch off echo cancellers and non liner processors as per ITU-T G.168.
(b) Switch from any adaptive/dynamic jitter buffer in use to a fixed-length (100 ms) jitter buffer.

(3) If PLC techniques and algorithms are used, they shall be suitable for facsimile modulations.

4.11 Signalling Requirements

4.11.1 General

(1) The PBX shall utilize the Session Initiation Protocol (SIP) as the call control protocol as described in RFC 3261.

(a) SIP traffic shall always pass between the CLNE and the customer PBX.
(b) The PBX shall not utilize DNS NAPTR and SRV queries to determine the IP address, transport protocol and port number of the CLNE. These shall be pre-provisioned in the PBX.
(c) Both the PBX and the CLNE shall use port "5060" for SIP signalling.
(d) The PBX shall not perform SIP registration with the CLNE.
(e) The PBX will be considered as an untrusted SIP entity. Hence :
   i. The caller's private identity information (P-Asserted-Identity header as described in RFC 3325) received from the PBX will be discarded.
   ii. The privacy preference of the calling user (through the use of Privacy Header as described in RFC 3325) received from the PBX will be discarded.

4.11.2 Media Capability Negotiation (SDP Characteristics)

(1) The PBX shall utilize the Session Description Protocol (SDP) as described in RFC 2327, in conjunction with the offer/answer model described in RFC 3264, to exchange session information with the CLNE.

(2) The SDP offers/answers from the PBX shall include the following:
(a) The IP address ("c=" field) of the PBX itself or the media endpoint (see the connection models in section 4.4).
(b) G.711 A-law as the codec (ptime (packetization rate) is not a mandatory field. If present, it shall be set to 20).
(c) If it is supported, RFC 2833 DTMF relay as the DTMF mode (The Telecom network only supports events 0-15).

(3) The PBX may offer codecs other than G.711 A-law, but these will be discarded by the Telecom network.

(4) <u>Example of SDP offer/answer from PBX:</u>

```
v=0
o= SIP 2000 1 IN IP4 192.168.4.2
s=SIP Call
c=IN IP4 192.168.4.2
t=0 0
m=audio 28292 RTP/AVP 8 101
a=rtpmap:8 PCMA/8000
a=ptime:20
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
```

(5) The PBX shall support the ability to receive session descriptions that have the "c=" field set to all zeros (0.0.0.0), or an "a=inactive" or "a=sendonly" attribute as an indication that the RTP bearer path shall be put on hold.

4.11.3 SIP Methods to support

The minimum set of SIP methods that require PBX support includes:

Method	Use

INVITE	The SIP INVITE method is used to invite another party to participate in a call session. The INVITE method can also be used within an existing dialog to change SDP characteristics once a call session has been established (in which case the INVITE is commonly called a REINVITE).	Mandatory
UPDATE	The SIP UPDATE method is used to modify SDP characteristics before a 200 OK (INVITE) has been received. If the PBX or the Telecom network wishes to update SDP characteristics (i.e. perform a second SDP offer/answer exchange) prior to call answer (i.e. before a 200 OK INVITE is sent or received), the UPDATE method shall be used. If the PBX receives an UPDATE request with a SDP offer, the 200 OK response shall contain a SDP answer.	Recommended (at minimum to receive)
ACK	The SIP ACK method confirms that a client has received a final response (2xx, 3xx, 4xx, 5xx or 6xx response) to an INVITE request. The PBX shall be able to send and receive SIP ACK requests. If the INVITE (or REINVITE) request sent to the PBX did not contain a SDP offer, then the SDP offer shall be included in the 200 OK (INVITE), and the SDP answer shall be included in the ACK. The PBX shall be able to receive SDP answers within the ACK requests.	Mandatory
BYE	The SIP BYE method terminates a call. The PBX shall be able to send and receive a SIP BYE request. The PBX shall only send a BYE if an INVITE dialog is confirmed (that is, a 200 OK INVITE and ACK have been successfully exchanged between the PBX and the CLNE). If the dialog has not reached the confirmed state, a SIP CANCEL shall be used instead.	Mandatory
CANCEL	The SIP CANCEL method terminates a pending INVITE before a 200 OK (INVITE) has been received. The PBX shall be able to send and receive a CANCEL request. The PBX shall only send (or receive) a CANCEL if an INVITE dialog is not confirmed (that is, a 200 OK INVITE and ACK have not been successfully exchanged between the PBX and the CLNE). If the dialog is confirmed, a SIP BYE shall be used instead.	Mandatory

4.11.4 Outgoing Call to the Telecom Network

(1) The PBX user will go off-hook, dial the escape code digit "1", receive external dial tone and then dial the destination number. Dial tone shall stop after the first digit of the destination number.

(2) At the end of the dialling the PBX shall send an INVITE request to the Telecom network to initiate a new call. The INVITE request shall have the following characteristics:

(a) From: <SIP URI containing the identity of the PBX user, i.e. calling party number in 0+National Number format, e.g. 042292510>,

   (i) "From" header shall include a valid number, a number provisioned in the Telecom network against the customer.

      - For a given call, the PBX shall choose which of its valid numbers to use on a per-call basis. For example, on a call from a user without a dedicated telephone number (i.e. DDI number) the PBX may choose to assert its "main" number (i.e. the pilot number), while a call from a user with a dedicated DDI number could use the number of that user's specific telephone number.

(b) To: <SIP URI containing the complete set of dialled digits including the escape code digit "1" minus the end-of-dialling character, e.g. 1077792620>

(c) Request-URI: <SIP URI containing the complete set of dialled digits including the escape code "1" minus the end-of-dialling character, e.g. 1077792620>

(3) Example:

```
INVITE sip:1077792620@4.111.120.131:5060 SIP/2.0
Via: SIP/2.0/UDP 192.168.4.2:5060;branch=z9hG4bK14b6df4a7f6
From: <sip:042292510@192.168.4.2>;tag=7a16344f
To: <sip:1077792620@4.111.120.131>
Call-ID: a18f4480-9dd1aeb6-843-204a8c0@192.168.4.2
CSeq: 1 INVITE
Max-Forwards: 70
```

(4) The initial INVITE request from the PBX shall include an SDP offer (refer to section 4.10.2).

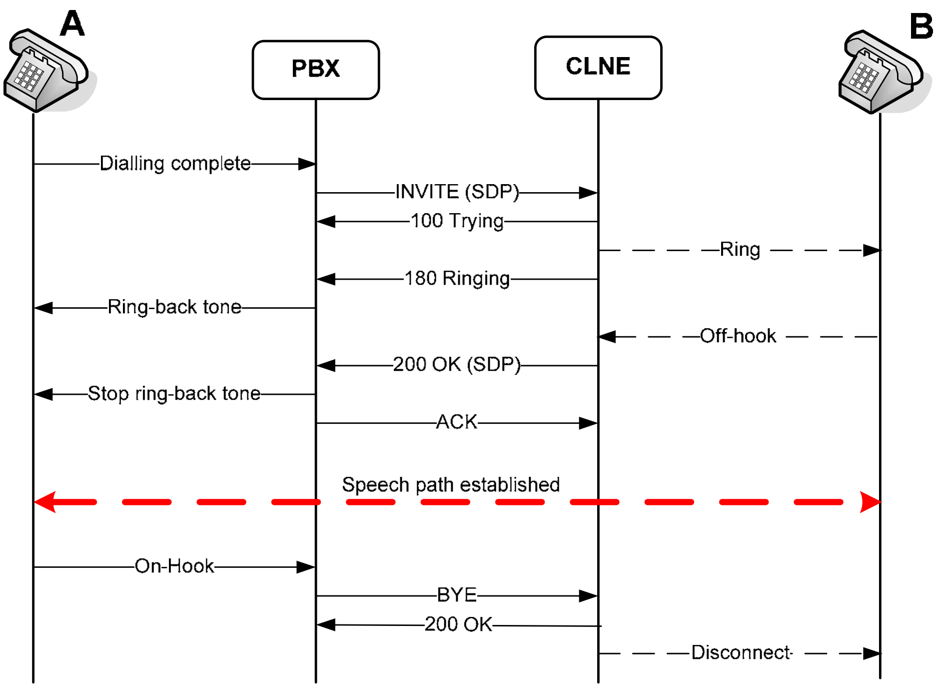
(5) The Telecom network will not challenge INVITE requests from the PBX. Hence, PBX support of 407 Proxy Authentication Required is not needed.

(6) Ring-back tone shall be output to the PBX user when 180 Ringing is received from the CLNE (Prior to receipt of 200 OK, the PBX shall also be prepared to accept 183 Session Progress, this is covered in (9) - (11)).

(7) When the call is answered, 200 OK will be sent to the PBX. It will have the SDP answer to the initial offer made in the INVITE.

(8) A bothway speech path shall be established after the PBX confirms the receipt of 200 OK with the ACK method.

Figure 4.5 Outgoing Call to the Telecom Network



Note 1: Signalling considerations between the PBX and the customer devices is outside the scope of this document.
Note 2: Signalling considerations beyond the CLNE is outside the scope of this document.

(9) It is possible that the Telecom network may send 180 Ringing, followed by 183 Session Progress with SDP slightly later in the call flow. The PBX shall apply local ring-back tone when 180 Ringing is received, and shall then stop ring-back tone and open the RTP bearer path when 183 Session Progress with SDP is received.

(10) Conversely, if the PBX receives 183 Session Progress with SDP followed by 180 Ringing, the PBX shall first establish a bearer path when 183 Session Progress is received, and then break the bearer path and apply local ring-back tone when 180 Ringing is received.

(11) An SDP answer received 183 Session Progress shall be considered to be the official SDP answer and the PBX shall not expect to receive any SDP in 200 OK when the call is answered. This means that the PBX shall preserve the SDP answer received in 183 Session Progress so that it can re-establish the RTP bearer path when 200 OK is received.

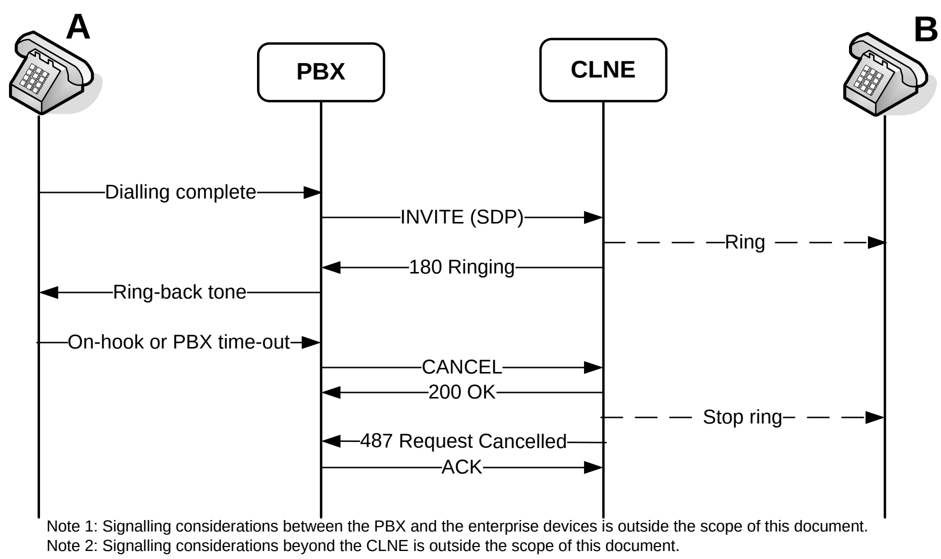
(12) The PBX may support PRACK functionality for provisional responses (180 and 183 responses) following the rules in RFC 3262.

(13) Main exceptions to normal call establishment

(a) The PBX user may abort the call during call setup (in ringing state). The PBX shall signal this by sending a CANCEL to the CLNE (see Figure 4.5).

(b) The PBX may time-out if there is no answer (200 OK) for at least 3 minutes and send a CANCEL to terminate the call (see Figure 4.5).

Figure 4.6 Call aborted by PBX



(c) Alternatively, the PBX may not have a timer running. If there is no answer (200 OK) from the called party for 4 minutes, the Telecom network will terminate the call by sending 480 Temporarily Unavailable to the PBX.

(d) It is possible for the Telecom network to send 180 Ringing, followed by a final INVITE response other than 200 OK (i.e. a 4xx, 5xx, or 6xx response). In this case, the PBX shall apply ring-back tone when 180 Ringing is received, and shall then stop it and provide an appropriate call progress tone (e.g. busy tone if 486 is received) based on the specific received response.

4.11.5 Incoming Call to the Customer Network

(1) The CLNE will send an INVITE request to the PBX to signal an incoming call. The INVITE request will have the following characteristics:

(a) From: <SIP URI including the calling party identity>
   (i) The PBX shall use the “From” header as the source of the CLI information.
   (ii) If the called PBX identity has been assigned the Calling Line Identity Presentation (CLIP) subscription service, the Telecom network will populate the “From” header with the calling

party number in the form of a 9-digit 0+National Number (e.g. 047792620) for New Zealand origins, or in the full international format (e.g. 0061222011887).

(iii) If the called PBX identity has been assigned CLIP, but the calling party has requested to remain anonymous, the Telecom network will set the "From" header to be <sip:anonymous@anonymous.invalid> so that the PBX can display "anonymous" or "private", or something similar.

(b) To: <SIP URI containing the called number in the form of a 9 digit 0+National Number, e.g. 042292510>

(c) Request-URI: <SIP URI containing the called number in the form of a 9 digit 0+National Number, e.g. 042292510>

(i) The PBX system shall be able to handle the incoming 9 digits and convert these digits to the PBX extension number.

(2) Example:

```
INVITE sip:042292510@192.168.6.5:5060 SIP/2.0
Via: SIP/2.0/UDP 4.111.120.132:5060;branch=z9hG4bK4C789F
From: <sip:077792620@4.111.120.132>;tag=4427E900-2697
To: <sip:042292510@192.168.6.5>
Call-ID: 57228965-241011DE-8A5CD79A-3D6AB2A5@4.111.120.132
CSeq: 101 INVITE
Max-Forwards: 66
```

(3) Incoming INVITE requests may or may not contain a SDP offer. The PBX shall be prepared to receive an INVITE or REINVITE request that does not contain SDP. When this occurs, the PBX shall include a SDP offer in 200 OK (see section 4.10.2).

(4) The PBX shall not challenge INVITE requests received from the CLNE.

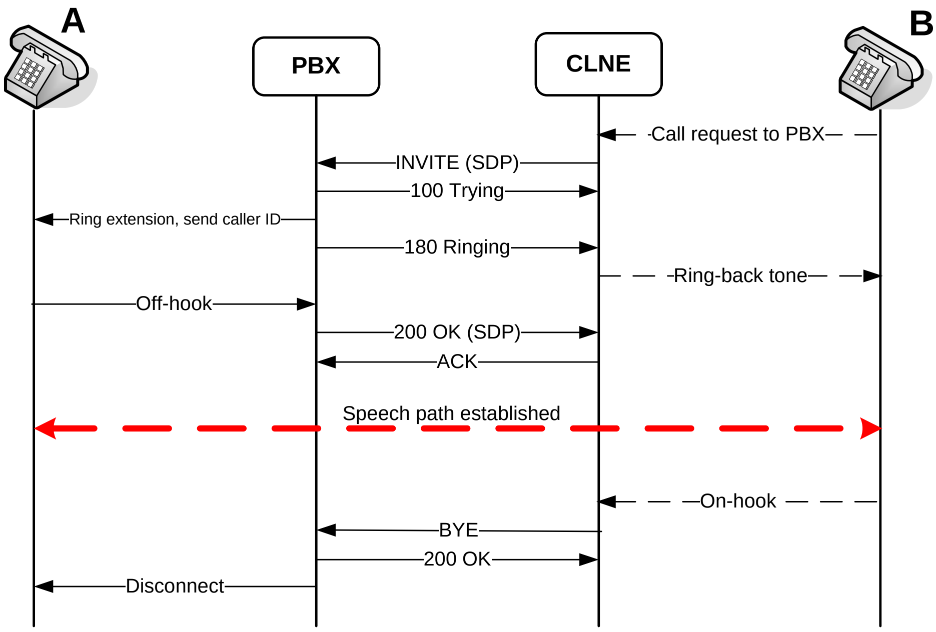
(5) The PBX shall respond to the CLNE with 180 Ringing, alert the extension that a new call has arrived and provide the extension with the calling party number. 180 Ringing shall not include SDP.

(6) When the PBX user goes off-hook, the PBX shall return 200 OK to the CLNE. 200 OK shall have the SDP answer to the initial offer made in the INVITE (see section 4.10.2).

(7) A bothway speech path shall be established after the CLNE confirms the receipt of 200 OK with the ACK method.

(8) The PBX may support of PRACK functionality for 180 Ringing following the rules in RFC 3262.

Figure 4.7 Incoming Call to the Customer Network

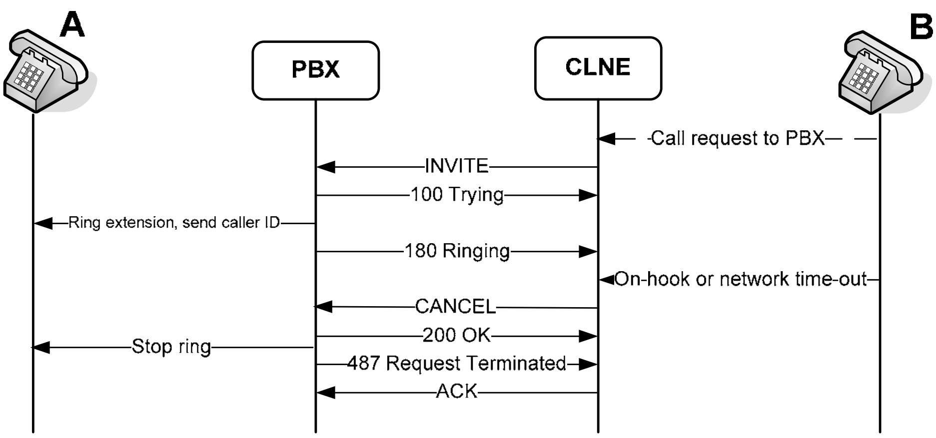


Note 1: Signalling considerations between the PBX and the customer devices is outside the scope of this document.
Note 2: Signalling considerations beyond the CLNE is outside the scope of this document.

(9) <u>Main exceptions to normal call establishment</u>

(a) The calling party may abort the call during call setup (in ringing state). The CLNE will signal this by sending a CANCEL to the PBX (see Figure 4.7).

Figure 4.7 Call to aborted by network



Note 1: Signalling considerations between the PBX and the customer devices is outside the scope of this document.
Note 2: Signaling considerations beyond the CLNE is outside the scope of this document.

(b) The PBX may time-out if there is no answer (200 OK) from the PBX extension for at least 3 minutes and send 480 Temporarily Unavailable or 487 Request Terminated to terminate the call.
(c) Alternatively, the PBX may not have a timer running. The Telecom network will terminate the call by sending a CANCEL to the PBX if there is no answer (200 OK) from the PBX user for 4 minutes (see Figure 4.7).

4.11.6 Calls in Progress

(1) Bothway audio connection shall be maintained until one of the parties clears the call (Note that the Telecom network will limit the maximum call duration to 24 hours).

(2) The PBX may apply either call waiting or busy treatment (respond to the CLNE with 486 Busy) when a third party calls a PBX extension that is already in conversation.

(3) While a call is in progress, either RFC 2833 DTMF relay or in-band methods may be used to communicate DTMF tones.

(4) Refer to section 4.10.2 to see how the PBX should handle the indication from the Telecom network that a call in progress (an RTP bearer path) shall be put on hold.

(5) Refer to section 4.11.1 to see how the PBX may put a call in progress (an RTP bearer path) on hold.

4.11.7 Call Tear-down

(1) To end the conversation either the PBX user or the remote party may hang-up.

(a) If the PBX user hangs up, the PBX shall send a BYE method to the CLNE to terminate the call (see Figure 4.4). The CLNE will respond to the PBX with 200 OK (BYE).
(b) If the remote party hangs up, the CLNE will send a BYE method to the PBX to terminate the call (see Figure 4.6). The PBX shall respond to the CLNE with 200 OK (BYE), and may generate disconnect tone.

4.11.8 Call Progress Tones

(1) The PBX shall locally generate NZ call progress tones or announcements in response to the following subset of standard SIP response codes. Selection of the particular tone/announcement is left to the equipment manufacturer's discretion.

SIP Response Code
180 Ringing
403 Forbidden
404 Not Found
408 Request Timeout
480 Temporarily Unavailable
486 Busy Here
487 Request Terminated
500 Server Internal Error

(2) In addition to the response codes outlined above, PBX systems should generate some form of NZ call progress tone or announcement for the remaining set of standard SIP response codes (where a call progress tone is applicable). Selection of the particular tone/announcement is left to the equipment manufacturer's discretion.

4.11.9 Early Media

(1) Early media refers to exchange of media (one-way or two-way) prior to answer of a call. This is required to provide progress tones and announcements from the Telecom network (e.g. ringback tone for calls to the PSTN, announcement treatment for unsuccessful calls).

(2) In order to support delivery of in-band announcements and call progress tones, upon receipt of SDP information in any '183 Session Progress' or '200 OK' message the PBX shall immediately disable any locally generated call progress tones and cut-through the early media to the end-user as described in RFC 3261.

4.12 PBX Service Interworking

Some telephony services may be provided by the PBX itself. The details provided in this section describe the requirements to be met by the PBX for proper interworking with the Telecom network.

4.12.1 Call Hold

(1) A PBX user may place an active call on hold and retrieve the held call. For these purposes, the Telecom network supports the use of REINVITE transactions to modify the RTP bearer path for a call according to the held/retrieved state.

(a) The PBX may use the direction attribute in an SDP offer as specified in RFC 3264 (e.g., a=sendonly, a=inactive, a=sendrecv).
(b) The PBX may also use the method of setting the media connection (IPv4) address to zero (0.0.0.0) in the SDP offer to place the RTP bearer path on hold as specified in RFC 3264.

(2) Alternatively, the PBX may stream recorded music (music on hold) to the remote party without changing the SDP characteristics of the ongoing SIP session.

4.12.2 Conference

(1) A PBX may provide conferencing services where one or more of the legs of a conference call may involve external users in the Telecom network.

(a) The occurrence of conferencing at the PBX shall be transparent to the Telecom network where the legs involving external users simply appear as DDI/pilot number calls from the perspective of the Telecom Network.
(b) The use of SIP REFER with Replaces (per RFC 3515 and RFC 3891) is not supported.

4.12.3 Call Transfer

(1) A PBX user may transfer an established call where the transferee and/or the transfer-to parties may involve external users in the Telecom Network.

(a) The occurrence of the transfer at the PBX shall be transparent to the Telecom network where the call legs involving the transferee and transfer-to parties simply appear as DDI/pilot number calls from the perspective of the Telecom network. The Telecom network may see REINVITE transactions to hold/retrieve the call legs and/or to direct the RTP bearer paths from the call legs between the transferee and transfer-to parties. In this case, the PBX is expected to remain in the signalling path of the transferred call.
(b) The use of SIP REFER with Replaces (per RFC 3515 and RFC 3891) is not supported.

4.12.4 Call Forwarding

(1) A call to a PBX user may be forwarded by the PBX where the forward-to party may refer to an external user in the Telecom Network.

(a) The occurrence of forwarding at the PBX shall appear as a new call in to the Telecom Network. The INVITE for the new (forwarded) call leg shall include a valid PBX customer number (forwarding user’s DDI or customer’s pilot number) in the “From” header.

APPENDIX 1: PTC 228 TEST PLAN

TEST REPORT TO SPECIFICATION PTC 228

Product Manufacturer: ....

Product Type: ....

Version Number: ....

Serial Numbers Tested: ....

Overall Compliance:

YES or NO

COMMENTS:

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Item	Test Required	Expected Result	Complies?
	1. External Laboratory		
§2.1	Electromagnetic interference (PTC 220 §1.3) - Compliance (as applicable) with Radiocommunications Act 1989, Radiocommunications (Radio) Regulations 1993 and AS/NZS 4252	Complies: Yes/No	Yes/No
§2.1	Electrical safety (PTC 220 §2.7) - Compliance (as applicable) with AS/NZS 60950	Complies: Yes/No	Yes/No
	Note: Compliance with Electromagnetic interference and Electrical safety requirements shall be the subject of separate tests conducted by a recognised independent test laboratory.		

Item	Test Required	Expected Result	Complies?
§4.10.4 §4.10.5	**2. Basic Calls**		
	Step 1. Originate a call according to the call matrix below. Confirm ringing and ring-back tone, answer the call and generate voice traffic. Continue the call to meet an average call holding time of 40 seconds. Note the time the call was made. Measure QoS (ie MOS, Jitter and delay) for a <u>selection</u> of calls, using a Sage 960B test instrument.	Call is established and both parties are able to hear each other clearly.	
	Step 2. Release the call and note which end terminated the call. Note the time the call was terminated.	Call is released without error.	
	Step 3. Collect the relevant traces for the time the calls were made.	G.711 A-Law negotiated for all calls. Target MOS, Jitter and delay scores are acceptable.	
	<u>Call matrix</u>		
	A. Local call: PBX under test – PBX under test (dial 1+DN)	As per Steps 1-3 above	Yes/No
	B. Local call: PBX under test – PLV (dial 1+DN)	As per Steps 1-3 above	Yes/No
	C. Local call: PBX under test – PSTN (dial 1+DN)	As per Steps 1-3 above	Yes/No
	D. National call: PBX under test – PSTN (dial 1+0+AC+DN)	As per Steps 1-3 above	Yes/No
	E. International call: PBX under test – International (dial 1+00+CC+AC+DN)	As per Steps 1-3 above	Yes/No
	F. Mobile call: PBX under test – 027 mobile	As per Steps 1-3 above	Yes/No
	G. Mobile call: PBX under test – 021 mobile	As per Steps 1-3 above	Yes/No
	H. Local call: PLV – PBX under test (dial 1+DN)	As per Steps 1-3 above	Yes/No
	I. Local call: PSTN – PBX under test (dial 1+DN)	As per Steps 1-3 above	Yes/No
	J. National call: PSTN – PBX under test (dial 1+0+AC+DN)	As per Steps 1-3 above	Yes/No

Item	Test Required	Expected Result	Complies?
	K. International call: International – PBX under test	As per Steps 1-3 above	Yes/No
	L. Mobile call: Mobile – PBX under test	As per Steps 1-3 above	Yes/No
	M. Emergency: PBX under test – 111 and 911	Call answered and acceptable call quality	Yes/No
	N. Telecom service call: PBX under test – 120 and 123, hang up after hearing announcement	Announcement received "welcome to Telecom" (or similar)	Yes/No
	O. Calling Card: PBX under test dials 0125, then enters calling card number. PIN entered after request, then terminating number.	Receive announcement & able to enter calling card number and PIN. Call answered and acceptable call quality. DTMF is sent from the PBX as per RFC2833.	Yes/No
	P. Directory Assistance: PBX under test dials 018	Call answered and acceptable call quality.	Yes/No
	Q. Conference Bridge: PBX under test dials 083033, and following announcement, enters Conference ID	Receive conference Bridge announcement and able to join conference.	Yes/No
	R. 0800 Number: PBX under test dials 0800 123456	Call answered and acceptable call quality.	Yes/No
	S. 0900 Number: PBX under test dials 0900 12345	Call answered and acceptable call quality.	Yes/No
	T. International Operator Assistance: PBX under test dials 0170	Call answered and acceptable call quality.	Yes/No
	U. Long duration calls: (1) PBX under test dials a PSTN number, call answered and remains set up for >24 hours (2) PSTN number dials PBX under test, call answered and remains set up for >24 hours	) Calls torn down) by Telecom) network after 24) hours	Yes/No Yes/No

Item	Test Required	Expected Result	Complies?
§4.10.4 §4.10.5	**3. Unsuccessful Calls**		
	A. Busy call. PBX under test dials a PSTN number that is busy.	Caller hears Busy Tone.	Yes/No
	B. Busy call. PSTN telephone dials a PBX under test extension number that is busy.	Caller hears Busy Tone.	Yes/No
	C. Unallocated number. PBX under test dials a PSTN number that is unallocated.	Caller hears Number Unobtainable (NU) Tone or similar (ref section 4.10.8).	Yes/No
	D. Unreachable Mobile: PBX under test dials 027 mobile that is currently switched off. Release the call and collect traces for the time the call was made.	Caller hears announcement, call is released without error and all resources are released successfully.	Yes/No
	E. Unreachable Mobile: PBX under test dials 021 mobile that is currently switched off. Release the call and collect traces for the time the call was made.	Caller hears announcement, call is released without error and all resources are released successfully.	Yes/No
	F. Time-out – PBX under test calls PSTN number, no answer from PSTN number	Ringing stops after 3-4 minutes	Yes/No
	G. Time-out – PSTN number calls PBX under test, no answer from PBX user	Ringing stops after 3-4 minutes	Yes/No
	H. Abandoned call: PBX under test calls PSTN number, no answer from PSTN. PBX user abandons call after a few ringing cadences.	Normal call teardown PBX and PSTN	Yes/No
	I. Abandoned call: PSTN number calls PBX under test, no answer from PBX. PSTN number abandons call after a few ringing cadences.	Normal call teardown PSTN and PBX	Yes/No
	J. Dialling error: PBX under test calls PSTN directory number (DN) in different local calling area without dialling 0 + area code.	Announcement received “your call cannot be completed as dialled” (or similar)	Yes/No

Item	Test Required	Expected Result	Complies?
§4.10.5	**4. Number Presentation/Restriction:**		
	Step 1: Assign (terminating) CLIP to numbers assigned to the PBX under test.		
	Step 2: Make local and national calls from PSTN numbers without CLIR to the numbers in step 1.	PSTN number is displayed on PBX under test's phone prior to answering (in 0+AC+DN format)	Yes/No
	Step 3: Make international calls from PSTN numbers without CLIR to the numbers in step 1.	PSTN number is displayed on PBX phone prior to answering (in 00+CC+AC+DN or 0000 format)	Yes/No
	Step 4: Make local and national calls from PSTN numbers with CLIR to the numbers in step 1.	PSTN number is not displayed on PBX under test's phone prior to answering (Anonymous is displayed)	Yes/No
	Step 5: Make local and national calls from PBX under test to PSTN numbers.	PBX under test number is displayed on PSTN phone prior to answering (in 0+AC+DN format)	Yes/No
	Step 6: Repeat Step 5 with dialled numbers prefixed in turn by 0197 (local and national) and 197 (local only).	PBX under test number is not displayed on PSTN phone prior to answering (Anonymous is displayed)	Yes/No
	Step 7: Assign (originating) temporary CLIR to numbers assigned to the PBX under test.		
	Step 8: Make local and national calls from PBX under test numbers in Step 7 to PSTN numbers.	PBX under test number is not displayed on PSTN phone prior to answering (Anonymous is displayed)	Yes/No

Item	Test Required	Expected Result	Complies?

	Step 9: Repeat Step 8 with dialled numbers prefixed in turn by 0196 (local and national) and 196 (local only).	PBX under test number is displayed on PSTN phone prior to answering (in 0+AC+DN format)	Yes/No

Item	Test Required	Expected Result	Complies?
	5. Miscellaneous		
§4.9	A. Fax Calls: Make calls from a Fax machine connected to the PBX under test to a Fax machine connected to the PSTN (and vice versa).	All Fax pages successfully received	Yes/No
§4.10.9	B. Announcements: Make calls from PBX under test to network Call Forwarding interrogation codes 1561, 1563 and 1567	Service announcement received	Yes/No
§4.10.2	C. Call Hold: Make a call from the PBX under test to PSTN and answer the call. The PSTN user puts the call on hold, makes a separate call and then retrieves the original call.	Call hold and retrieval successful, and PBX caller hears tone or music	Yes/No
§4.11.1	D. Call Hold: Make a call from PSTN to the PBX under test and answer the call. The PBX user puts the call on hold, makes a separate call and then retrieves the original call.	Call hold and retrieval successful, and PSTN caller hears tone or music	Yes/No
§4.11.4	E. PBX-provided Call Forwarding: Make a call from the PSTN to the PBX under test. The PBX then forwards the call to another PSTN number.	Calling Party Number of the forwarded call is the PBX number	Yes/No
§4.10.4	F. Post-dialling delay. Measure post-dialling delay for local (1+DN) and national (1+0+AC+DN) calls	No greater than 2 seconds	Yes/No

Item	Test Required	Expected Result	Complies?
	6. Transmission Tests		
§4.6.1	A. Loss Plan. Measure TRP losses as per Appendix 2A.	0.5 dB to -0.5 dB	Yes/No
§4.6.2	B. Attenuation Frequency Distortion. Measure loss distortion using the method outlined in Appendix 2A.	As per section 4.6.2	Yes/No
§4.6.3	C. Variation of Gain with Input Level. Measure gain variation using the method outlined in Appendix 2A.	As per section 4.6.3	Yes/No
§4.6.4.1	D. FXS Loss Plan. Measure FXS loss as per Appendix 2B.	As per section 4.6.4.1	Yes/No
§4.6.4.2	E. FXD Loss Plan. Measure FXD loss as per Appendix 2A.	As per section 4.6.4.1	Yes/No
§4.6.5.1	F. Digital Phone Loudness Rating. Measure as per Appendix 2B.	As per section 4.6.5.1	Yes/No
	7. PBX SIP Trunking Interoperability Specification		
§2.2.4	PBX Configuration	Document	Yes/No
	CLNE Configuration	Document	Yes/No

APPENDIX 2: PTC 228 TRANSMISSION TEST METHOD

This Appendix contains a basic description of the test methods to be used in order to collect the transmission test results in relation to transmission tests specified in Appendix 1.

The proposed testing methods are based on the availability of a 4-wire E&M test circuit at 0 dBr TRP. This circuit, located in the Telecom NIL laboratory (refer section 2.2.2), has been specifically set up for transmission testing.

A. Loss / Level Tests

The Loss / Level related tests can be carried out utilising a Telecom 4-wire Test Circuit, specifically set up for the transmission and end-to-end performance testing. The test circuit is located in the Telecom PTC Lab, in the Telecom NIL.

On the CPE side, a suggested way of setting up a 4-wire transmission path is through utilising a soft phone application, coupled with the sound card within a PC or a laptop which provides for the Tx and Rx interconnection.

Test Procedure for Loss / level testing for PBX SIP Trunking – VoIP/SIP Interface:

1. A PC or a Laptop equipped with a sound card is connected to an Ethernet port on the Enterprise Voice Network.
2. A soft phone application is launched (e.g. SJ Phone) and a SIP call is established to the Telecom 4-wire test circuit.
3. Once the end-to-end call is established, the HP PCM 3776A test instrument is connected to both ends. The SEND and RECEIVE level traces may thus be collected (see Figure A for the test set up).

NOTE: It is important to ensure that the sound card is calibrated before testing, in order to make sure that no additional loss or gain is being introduced. The calibration test circuit is defined in Figure B.

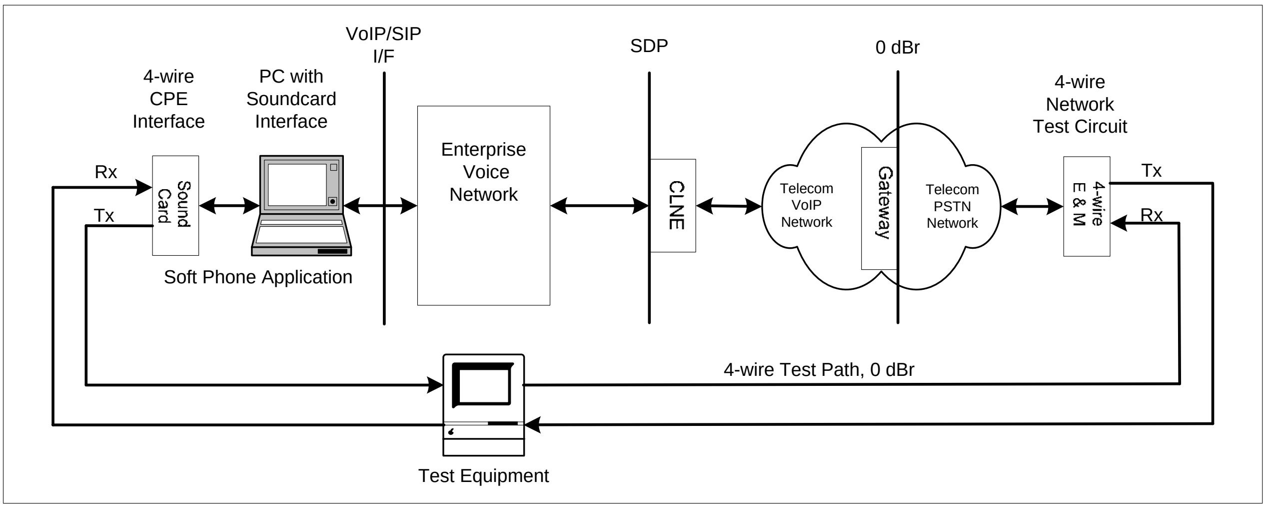

Figure A: Transmission Loss / Level Test Setup for PBX SIP Trunking VoIP/SIP Interface

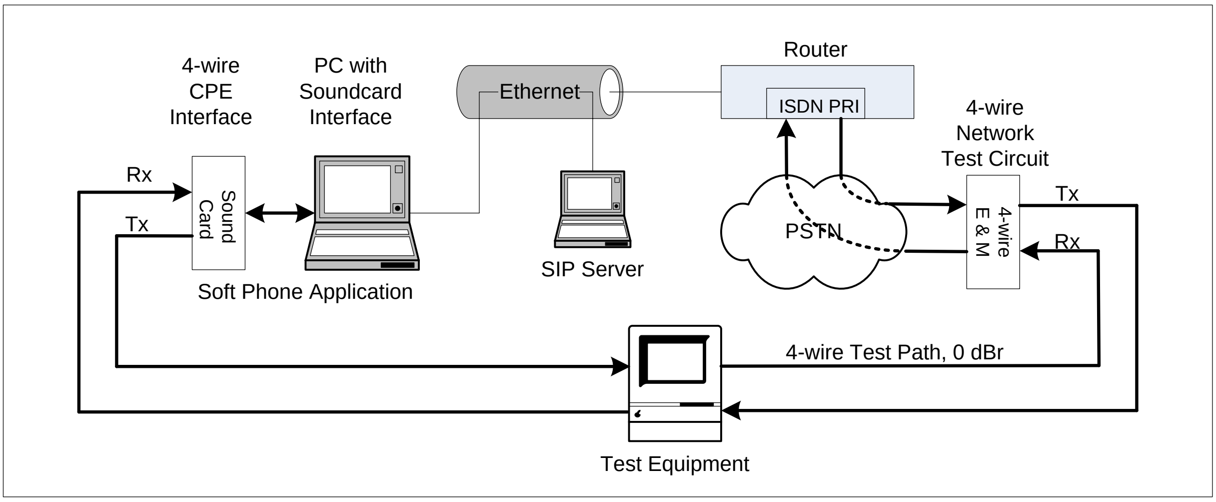

Figure B: Calibration Circuit for a Soft Phone and the PC Sound Card

<u>Test Procedure for Loss / level testing for PBX SIP Trunking – FXS Interface:</u>

1. An Analogue Telephone is connected to an FXS port on the Enterprise Voice Network.
2. A call is established from the Analogue Telephone to the Telecom 4-wire test circuit.
3. Once the end-to-end call is established, the HP PCM 3776A test instrument is connected to both ends. The SEND and RECEIVE level traces may thus be collected (see Figure C for the test set up).

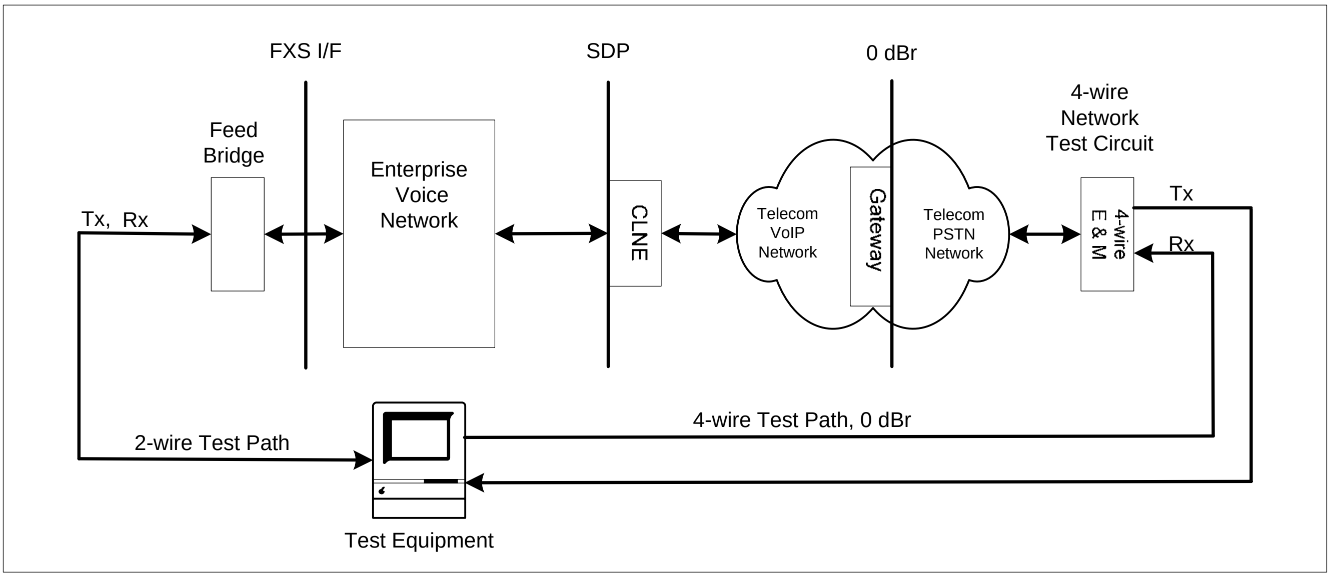

Figure C: Transmission Loss / Level Test Setup for PBX SIP Trunking FXS Interface

B. Loudness Rating Tests

The Loudness Rating testing, which incorporates a Telephone instrument as well as the Enterprise Voice Equipment, is referred to as the Applied Loudness Rating. This meaningful measurement is very useful when is difficult to separate a Telephone instrument from the proprietary equipment (e.g. a Digital Set).

In the case of the FXS or VoIP/SIP ports, the Applied Loudness Rating measurement may be used in order to measure the overall loss, provided that the SLR and RLR values of each stand-alone Telephone are known.

The Applied Loudness Rating tests can be carried out at the VoIP/SIP, FXS and Digital or Proprietary interfaces for PBX SIP Trunking and extended to the Telecom 4-wire test circuit (see Figure D for the test set up).

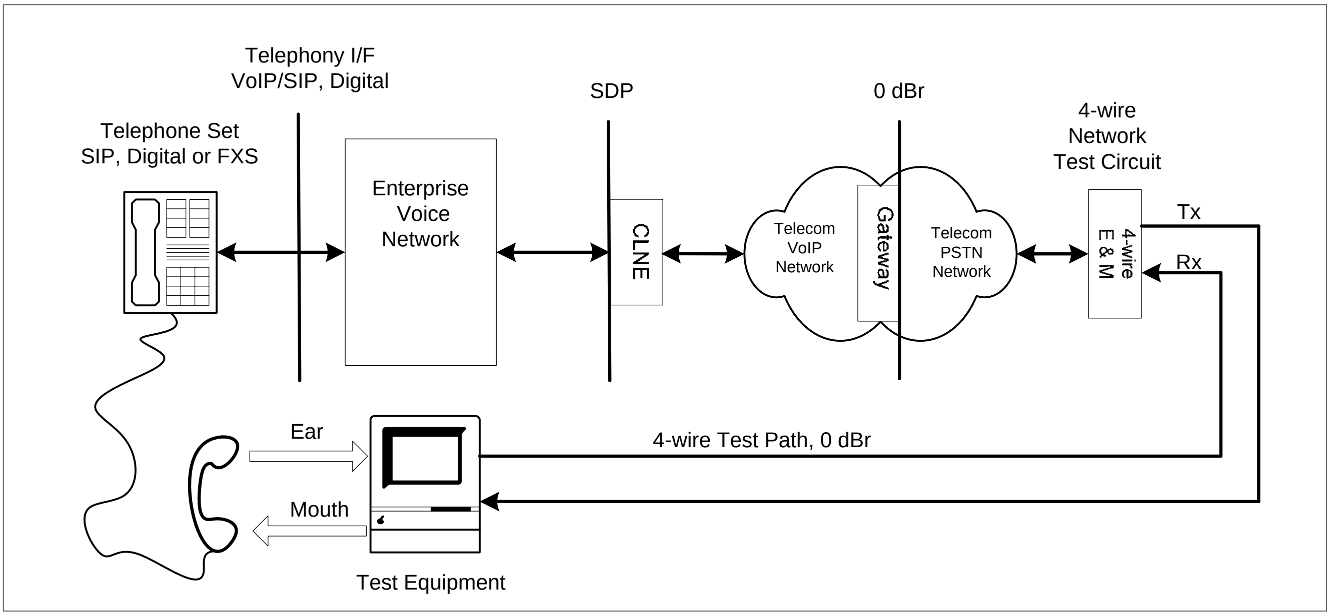

Figure D: Loudness Rating Test Setup for PBX SIP Trunking Digital, SIP or FXS Telephone Set

Test Procedure for PBX SIP Trunking Loudness Rating testing:

1. An Analogue, SIP or Digital Telephone is connected to an appropriate port on the Enterprise Voice Network.
2. A call is established from the Telephone to the Telecom 4-wire test circuit.
3. Once the end-to-end call is established, the PTC Port Tester is connected to the Telecom 4-wire test circuit with the handset fixed into the B&K 4905 Acoustic Test Head.
4. The test application is invoked and the Loudness Rating data collected.

END