NET Certification PBX Guide for Voice Connect (GVC2) SIP-T Service
(PTC 229 – Limited Certification)

Circuit #2 - PTC Lab Reference Circuit.

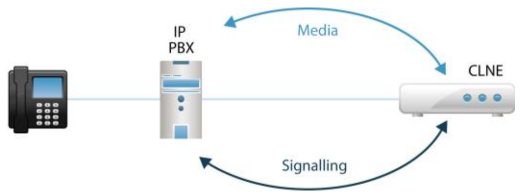
In order to connect the SIP-enabled CPE to Spark Network, the equipment must be tested for compliance to PTC 229 Specification.

To save time it would be preferable if the equipment was already configured upon the arrival at the test lab.

The SIP-enabled DUT or Customer PBX is directly connected to Telecom Test Circuit or CLNE.

The CLNE LAN interface is on the same subnet as the PBX's IP interface. This connection can be done physically, with the use of a crossover cable, or via an Ethernet switch. Neither the CLNE nor the PBX need routing information as there are no routers between them.

In this scenario all signalling and media traffic passes between the CLNE and the customer's PBX.



1. The CPE (PBX) configuration details are as follows:

Proxy and Registration:

- DNS IP Address: - Primary - Secondary	→ 122.56.237.1 → 210.55.111.1
- SIP Domain:	ptclab.co.nz
- SIP Outbound Proxy: - Option 1 - Option 2 - Option 3	→ wn0304-p01.telecom.co.nz (A RR) → gvc-wn0304.telecom.co.nz (SRV RR) → 122.56.254.104 (IP Address)
- Registration Required:	YES
- Use Outbound Proxy:	YES
- SIP Username:	44711860
- SIP Password:	To be advised

Note Example: - The SIP Registration User or Call Format is: 44711860@ptclab.co.nz

Subscriber Information:

- PBX IP address:	192.168.2.10 /24
- CLNE PBX faced port: (Default Gateway)	192.168.2.100 /24
- Display Name:	PTC lab
- Phone Numbers: - Pilot: - Number Range:	→ 44711860 → 44718380 to 89
- Make Call Without Registration:	NO
- Answer Call Without Registration:	NO
- Outgoing Call prefix:	NO
- Use DNS SRV:	YES
- DNS SRV Auto Prefix:	YES (if Option 1 used for SIP Outbound Proxy) NO (if Option 2 used for SIP Outbound Proxy)

Note: - The PBX IP Interface is connected to **CLNE Port GE 0/1.**

2. Additional Information:

2a) Domain Names and IP addresses;

SRV RR	A RR	IP Address
gvc-wn0304.telecom.co.nz	wn0304-p01.telecom.co.nz	122.56.254.104
	wn0304-s01.telecom.co.nz	122.56.254.105

2b) P-Assured Identity (PAI);

The use of PAI is a **MUST**, otherwise DID calls from PBX extensions which are programmed with DID numbers will not be supported.

DID calls from PBX extensions which are programmed with Pilot number will be supported.

2c) IP QoS Markings;

The following table lists the various different names / values for DSCP values:

DSCP Class	DSCP (bin)	DSCP (hex)	DSCP (dec)	ToS (dec)	ToS (hex)	ToS (bin)	ToS Prec. (bin)	ToS Prec. (dec)	ToS Delay Flag	ToS Throghput Flag	ToS Reliability Flag	TOS String Format
cs3	011000	0×18	24	96	0×60	01100000	011	3	0	0	0	Flash
af31	011010	0×1A	26	104	0×68	01101000	011	3	0	1	0	Flash
af41	100010	0×22	34	136	0×88	10001000	100	4	0	1	0	Flash Override
ef	101110	0×2E	46	184	0xB8	10111000	101	5	1	1	0	Critical

Note: - The Telecom Core routers require DSCP to be set to **cs3** or **af31.**

Telecom expects and strongly recommends for call quality reasons that all multimedia packets are marked with the following DSCP (DiffServ Code Point).

- Audio Stream: EF
- Video Stream: AF41
- SIP Signalling: CS3 (or AF31 *)
- Image (T.38): EF

* The SIP signalling will be presented from the Network using CS3, however customers that mark signalling traffic as AF31 will still be treated as though it was marked as CS3.

If a PBX does not set DSCP or has not set it as per the interface spec; the call will still be accepted.

If the PBX does not set the DSCP then it will be marked as TC4 - BusinessDataHigh (the minimum Gen-i forwarding class used for business services). This is substantially better than best efforts (TC5) and was a reasonable thing to do as it aligned with the Gen-i guide of nothing less than TC4.

If packets are incorrectly marked they may be discarded or the call quality may suffer as the incorrect treatment may be applied. This is especially important when MSSA (Multiple Services Single Access) functionality is used with GVC2.

2d) SIPconnect – RFCs Supported;

The SIPconnect Technical Recommendation is an industry-wide, standards-based approach to direct IP peering between SIP-enabled IP PBXs and VoIP service provider networks.
The table below identifies the RFC and standards that have been used when compiling the Voice Connect solution; it does not mean that Voice Connect is compliant to all options of an RFC.

Standard	Description
RFC 791	Internet Protocol (IPv4)
RFC 2327	SDP: Session Description Protocol
RFC 2782	A DNS RR for specifying the location of services (DNS SRV)
RFC 2833 (Obsolete)	RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals RFC 2833 is Spark'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).
RFC 3261	SIP: Session Initiation Protocol The transport methods supported are UDP (RFC 768) and TCP (RFC 793). UDP is Spark's preferred transport method. The PBX-specific agreed transport method (i.e. UDP or TCP) shall apply in both directions between the PBX and the service.
RFC 3262	Reliability of Provisional Responses in the Session Initiation Protocol (SIP)
RFC 3263	Session Initiation Protocol (SIP): Locating SIP Servers
RFC 3264	An Offer/Answer Model with Session Description Protocol (SDP)
RFC 3311	The Session Initiation Protocol (SIP) UPDATE Method
RFC 3515	The Session Initiation Protocol (SIP) Refer Method
RFC 3550	RTP: A Transport Protocol for Real-Time Applications
RFC 3891	"Replaces" Header.
RFC 3892	Referred-By Mechanism.
RFC 3960	Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP),
RFC 4028	Session Timers in the Session Initiation Protocol (SIP)

RFC 4244	An Extension to the Session Initiation Protocol (SIP) for Request History Information
RFC 4566	SDP: Session Description Protocol (obsoletes RFC 2327)
RFC 4629	RTP Payload Format for ITU-T Rec. H.263 Video
RFC 4904	Representing Trunk Groups in tel/sip Uniform Resource Identifiers (URIs)
RFC 5806	Diversion Indication in SIP (network accepts does not send)
ITU G.168	Echo cancellation

2e) Status Codes;

Telecom will provide relevant SIP message back to the PBX, i.e. SIP 4xx or 5xx or 6xx response messages, rather than tone or announcement.

This means that the PBX will have to generate the appropriate tone for the PBX users and display message for the phone screens.

Telecom will send the following SIP responses towards the PBX based on the release cause:

Incoming SIP Status Code	Release Cause	Recommended Tone Generated by CPE
404 Codes below are mapped to 404: 410, 484, 604	USER_NOT_FOUND	Number Unobtainable Tone
413 This code is mapped to 404	TRANSLATION_FAILURE	Number Unobtainable Tone
486 600 is mapped to 486:	BUSY	Busy Tone
480 Codes below are mapped to 480: 401, 407, 606	TEMPORARILY_UNAVAILABLE	Disconnect Tone
403, 603 These codes are mapped to 480	FORBIDDEN	Disconnect Tone
408 This code is mapped to 480	REQUEST_TIMEOUT	Disconnect Tone
400, 402, 405, 406, 409, 411, 414, 415, 420, 422, 481, 482, 483, 485, 487, 488, all other 4XX not listed above These codes are mapped to 480	REQUEST_FAILURE	Disconnect Tone
All 5xx	SERVER_FAILURE	Disconnect Tone
All other 6XX not listed above	GLOBAL_FAILURE	Disconnect Tone

Telecom recommends that the received SIP Status Code is mapped on the PBX to an appropriate tone/message and a CPE display message (where applicable).

2f) Tone Plan;

The Standard TNZ Tone expected from CPE devices are defined below:

Tone Name	Abbrev	Frequency (Hz)	Output Level (dBm)	Cadence
Busy Tone	BT	400Hz	-15	500 ms ON 500 ms OFF Sequence repeated until timeout (45 sec)
Disconnect Tone (also known as Congestion Tone)	DSCT	400Hz	-15	250 ms ON 250 ms OFF Sequence repeated until timeout (45 sec)
Number Unobtainable Tone	NUT	400Hz	-15	75 ms ON 100 ms OFF 75 ms ON 100 ms OFF 75 ms ON 100 ms OFF 75 ms ON 400 ms OFF Sequence repeated until timeout (45 sec)
Ring Back Tone (also known as Ringing Tone)	RBT	400Hz + 450Hz	-18	400 ms ON 200 ms OFF 400 ms ON 2.0 sec OFF Sequence repeated until CPE ringing timeout (300 sec, unless preceded by answer, abandonment or other timeout), then Busy Tone

Basic Functional tests: PTC 229 – Limited Certification

PBX Model:	
H/F version:	
F/W version:	
Note: PCaps traces or SIP Messages to be captured for all tests.	

Test	Call Flow Example	Test Description	Result		Comments
			Pass	Fail	
1		Incorrect Password action?			
2		Is Registration to SIP Server Successful?			
	7.1	…..what is the Retry Interval?			
	7.3	…..is there a Back OFF process followed?			
	7.4	....Acceptable is repeated Retries at greater than 1 minute intervals.			
	7.5	….Acceptable is retries at the standard SIP Retry Intervals of .5, 1, 2, 4, 8, 16, 32, 64 Seconds and maybe 1, 2, 4, 8 etc. Minutes.			
	7.6	….Not Acceptable is 1 second Repetitions.			
3		Can outgoing calls be made from the Pilot extension?			
4		Can incoming calls be made to Pilot extension?			
5		Can outgoing calls be made from the DID extension? (related to P-Assured Identity (PAI))			
6		Can incoming calls be made to DID extension?			
7		Is CLID correct for the DID Extension?			
8		Can outgoing calls be made from a non-DID extension?			
9		Is CLID correct for the non-DID Extension?			
10		Does the DTMF signalling use RFC2833?			
11		Is the DTMF signalling satisfactory?			
12		DSCP QoS marking: CS3 or AF31?			
13		RTP QoS marking: EF?			
14		Is the speech satisfactory for both parties on call?			
15		Consulted Call transfer to external number?			
16		Immediate Call transfer to external number?			

Equipment for testing should be sent to:

Access Standards,
L12, Spark Central, Boulcott,
42-52 Willis Street, WELLINGTON

Attn:	Bill Dawid (+64 4 3825730) (email: bill.dawid@spark.co.nz)	or	Shaun Godfrey (+64 4 8029860) (email: shaun.godfrey@spark.co.nz)

Please do not hesitate to contact us if you require any further information.

Regards,
Shaun / Bill