Supplement to PTC207and PTC217

Summary of PTC requirements for Private Voice Networks connected to the PSTN/ISDN

DRAFT FOR COMMENT 23 October 2001

1
General

1.1

This Specification covers the requirements for a private voice networks which are connected as customer equipment to the Telecom PSTN/ISDN. It is not technology specific, and is intended to cover traditional circuit switched PABXs as well as VoIP networks and wireless technologies such as DECT. It should be read in conjunction with PTC207 or PTC217, and does not replace these Specifications.

1.2
Telepermits

Telepermits will be generally be granted for for complete systems – i.e. network interface (FXO) and packet phones and/or analogue ports (FXS). Telepermits may be also granted for components of a system, but must be tested as a complete system (see Appendix 1), and where a component is granted a Telepermit, the other compatible component required to make up a complete functional system must also be Telepermitted. Telepermits will generally be granted in the PTC207 series.

2
Transmission

2.1
General

This section covers parameters which deal directly with the quality of the voice signals themselves. They include transmission level plans, impedance plans (for analogue interfaces), delay plans, and methods of managing distortion. Independent parameters are measured and an overall transmission performance indication is calculated using the E-model.

2.2
E-model

When PTC 207 and PTC 217 were written, transmission impairments were dealt with using Quantization Distortion Units (QDUs). These were originally intended to quantify the distortion for an A or mu law codec pair in such a way that the overall effect of multiple conversions between analogue and digital could be assessed by adding the QDUs together. While QDUs were assigned to low bit rate encoders based on group assessment processes, these were found to be inadequate for all but very simple networks. The E-model was subsequently developed by ETSI to take account of all the impairments which lead to speech degradation, and in particular takes into account impairments which typically occur in packet based networks. In the E-model, impairment values are assigned to a number of independent parameters, which are then combined to give a transmission rating factor *R* as follows:

$$R = Ro - Is - Id - Ie + A$$

Ro represents in principle the basic signal-to-noise ratio, including noise sources such as circuit noise and room noise.

Is is a combination of all impairments which occur more or less simultaneously with the voice signal. This includes loudness, sidetone, and quantizing distortion from analogue/digital conversions.

Id represents the impairments caused by delay, which include Talker and Listener echo, and end to end delay.

Ie represents impairments caused by low bit rate codecs.

The expectation factor *A* allows for compensation of impairment factors when there are other advantages of access to the user, such as mobility.

- *Some parameters particularly receive and send loudness ratings impact on more than one impairment factor.*

- *Some factors are determined in terminal equipment. For example in digital networks which are in themselves lossless Loudness ratings are determined entirely by terminal equipment (phone) design. Other factors such as delay. are partly determined in terminal equipment design, partly determined in network design, and in packet networks, could vary according to network loading.*

2.3
Transmission Levels

This part of the specification covers the requirements for transmission levels in a private voice network which is connected as customer equipment to the Telecom PSTN/ISDN. The transmission plan is designed to allow good performance in an any to any call scenario, and the mixing of analogue and digital phones and interfaces.

2.3.1
Design Objectives

Proposed pads and levels are optimised for the long-term "all-digital" situation;

Ultimately, the circuit-switched public network will be replaced by an IP network which will directly connect into the private IP network with no FXO function;

Network planning and digital terminal design for voice functionality shall comply with the ITU Overall Loudness Rating objectives of 10 dB, with SLR of +8 dB, RLR of +2 dB and no circuit losses;

It is recognised that most traffic on a private network is usually "extension to extension", whether the terminals are all digital or mixed analogue and digital. As such, the FXS pad values are optimised for this situation and set at "standard values" for use in all circumstances;

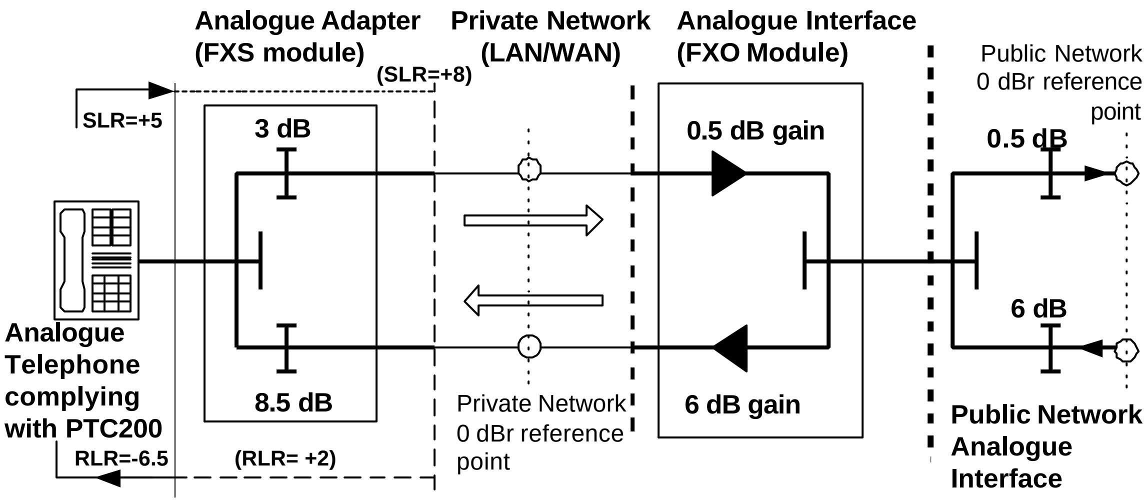
This level/loss plan recognises that digital trunks should always be used between the digital PSTN and the private IP network. In this case, the zero level point is simply extended from the PSTN into the private IP network, with no pads or gains in the interface.

In the event that analogue trunks are the only option available. The default FXO settings are gains to compensate for the T and R pads used in the PSTN. This means that private IP networks interfacing with the PSTN via analogue trunks suffer a transmission level loss relative to the optimum. This loss would be exacerbated in cases where long analogue trunks have to be used. While the FXO gains may be adjusted to compensate for analogue trunk loss in such cases, it is unlikely that long analogue trunks will be encountered.

2.3.2
Testing

All level measurements shall be performed with G.711 A-law codecs utilised. Performance using other codecs cannot be measured using standard techniques, and in the interim, the performance of non-waveform codecs should be determined by subjective comparison.
Any echo cancellers shall be turned on, and any silence suppression processes should be turned off.

2.3.3
Standard PTC200 phone port to analogue trunk

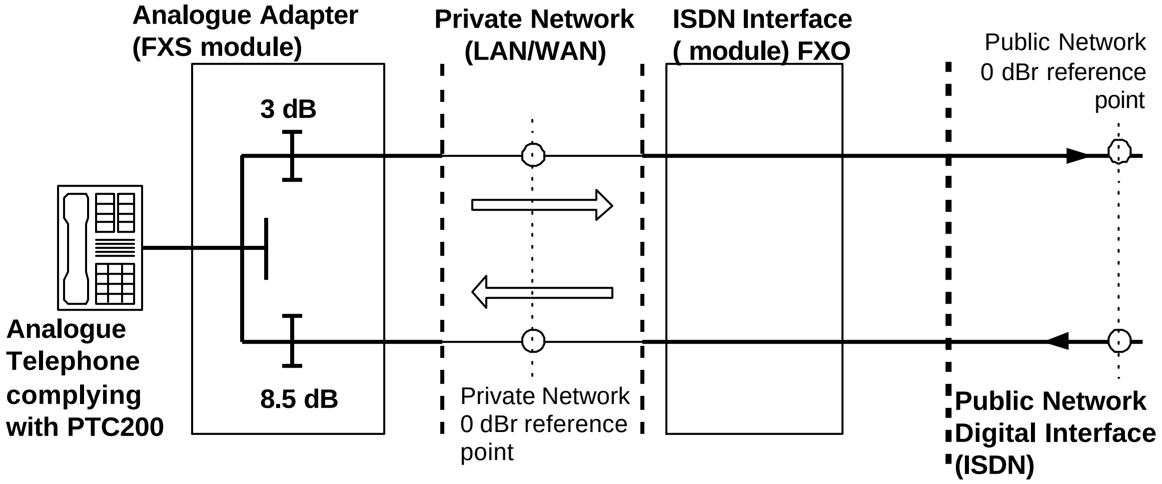


Loss from FXS port to FXO port: 1 to 5 dB, **Objective: 2.5 dB**
Loss from FXO port to FXS port: 1 to 5 dB, **Objective 2.5 dB**

The losses/gains shall be measured at 1000Hz

The variation of gain/loss shall be not more than +/- 0.5dB of the 1000 Hz value across the frequency band 300 to 3400 Hz

2.3.4
Standard PTC200 phone port to digital (ISDN) trunk

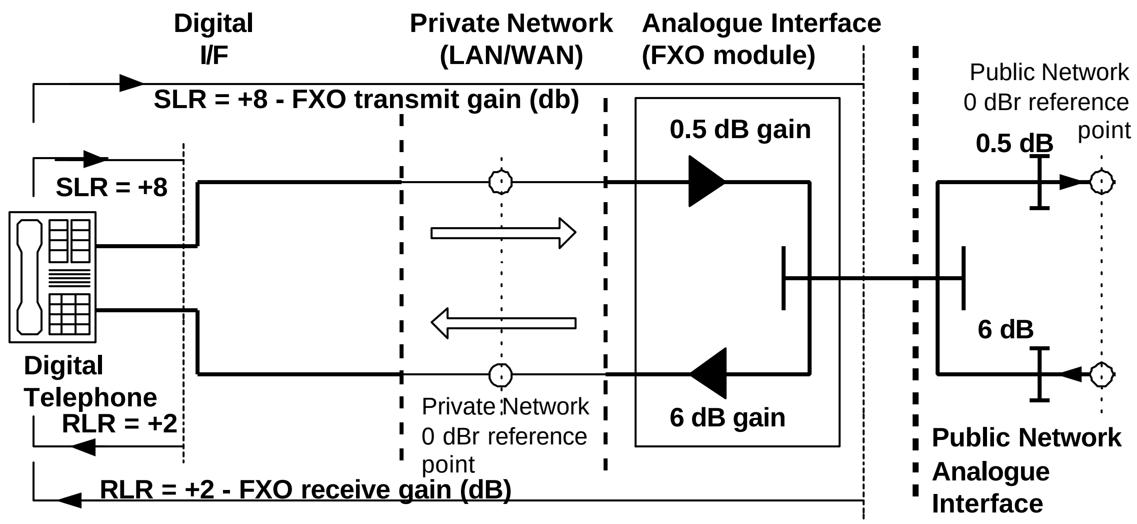


Loss from extension port to ISDN: 1 to 5 dB, **Objective: 3 dB**
Loss from ISDN to extension port: 6.5 to 10.5 dB, **Objective 8.5 dB**

The losses/gains shall be measured at 1000Hz

The variation of gain/loss shall be not more than +/- 0.5dB of the 1000 Hz value across the frequency band 300 to 3400 Hz

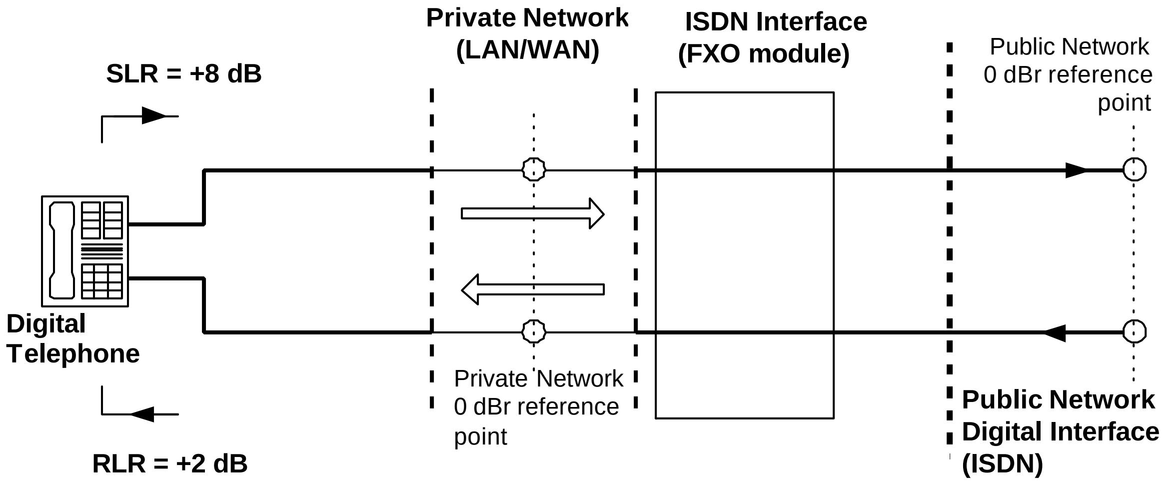
2.3.5
Digital Phone to analogue trunk



Send and Receive Loudness Ratings to be measured between IP Phone handset and 2-wire analogue network interface
Send Loudness Rating (SLR) +11 dB to +2 dB, **Objective: +5 dB**

Receive Loudness Rating (RLR): -1 dB to -10 dB (-14 dB with volume control), **Objective: -6.5 dB**

2.3.6
Digital Phone to digital (ISDN) trunk



Send Loudness Rating (SLR): +10 dB to +6 dB, **Objective: +8 dB**

Receive Loudness Rating (RLR): +4 dB to 0 dB (-8 dB with volume control), **Objective: +2 dB**

2.3.7
Additional gain
To overcome losses in access cable between PSTN and Analogue FXO module, the additional gain of up to 3 dB may be added symmetrically in each direction in the FXO module. This is subject to the stability criteria of PTC 200 clause 4.12.

Where an analogue FXS port is used (ref clause 3.1 of this specification), a PTC200 compliant phone with SLR of <+5 dB and RLR of <–6.5 dB shall be connected to the port, and tested as per PTC 200 clause 4.12.

2.3.8
Noise
Noise shall meet the requirements of PTC 109 section 7.

2.3.9
Codec Distortion
Impairments due to codec distortion are given in Table 4.1 for standard codecs. If other Codecs are to be used, the Impairment factors shall be calculated as per ITU-T Recommendation G.113.

2.3.10 Delay

Delay can occur in several parts of an end to end call. Some delay occurs in the customer equipment – e.g. IP phone and gateway, some in the customer IP network, and some in public and other networks (particularly international links) between the customer gateway and the other end of the call. In general, if end to end delay (one way) exceeds 150ms users begin to experience "double talk" (both parties talking at once after a pause), and the call becomes perceived as being unacceptable. In a packet world the situation is further complicated by the fact that the delay can vary depending on network loading. To give a reasonable chance of calls being satisfactory, it is necessary to allocate delays to different parts of the end to end call. The worst case situation is an international call with VoIP private networks at both ends. The following table gives maximum delays to the halfway point in the international call.

Network element	**Delay** **G.711 codec**	**Delay** **Low bit codec e.g. G.729**
IP phone & Gateway	1 ms	20 ms
Customer IP network*	21.5 ms	2.5 ms
PSTN/ISDN	20 ms	20 ms
½ International Link**	57.5 ms	57.5 ms
Total	**100 ms**	**100 ms**

* This is the maximum delay which may be introduced by the customer network and still keep within the 200 ms total end to end delay.

** International link made up of 15 ms codec (G729(A) + 100 ms delay.

- *If an international call is made, the local VoIP system must be set to G.711, or there is very little budget for customer network delay.*

- *In practice delay in some international routes is likely to exceed 100 ms.*

- *This table allows a total of 200 ms end to end delay. While this is above the 150 ms, it is a worst case scenario, which will not always occur in practice, i.e. the phones at both ends of the call will not always be at the end of VoIP networks, and other factors, particularly loudness ratings can overcome the effects of delay to a small effect.*

For the purposes of Telepermitting equipment, the delay in ms shall be measured for each codec (where applicable) as follows:

Delay in ms FROM	**TO**
Digital Phone (acoustic)	Digital Network Interface
Digital Phone (acoustic)	Analogue Network Interface
Analogue 2-wire port	Digital Network Interface
Analogue 2-wire port	Analogue Network Interface
Digital Network Interface	Digital Phone (acoustic)
Digital Network Interface	Analogue 2-wire port
Analogue Network Interface	Digital Phone (acoustic)
Analogue Network Interface	Analogue 2-wire port

For testing, the IP network, should be a direct connection between the gateway and either the IP phone or the analogue adapter.

2.3.11
Echo

Where the mean one way propagation time exceeds 15ms echo cancellers should be deployed. VoP and wireless technologies are likely to have inherent delays exceeding 15ms and would be expected to have echo cancellers fitted. Assuming echo cancellers are fitted, values of 65dB and 110dB are used for TELR and WEPL respectively in calculating the delay impairment factor Id.

- *Ref ITU-T Recommendation G.107*

3
Electrical Interface Parameters

3.1
Analogue network interface
(a) PTC 200 clause 4.2(2)

- *Applicable with system dependant phone.*

(b) PTC 200 clauses 4.3 (noise & crosstalk), 4.4 (distortion), 4.5 (impedance), 4.8 (impedance balance ratio to earth)

(c) PTC 200 clause 4.9 (frequency response)

- *To be measured between analogue trunk and extension port and/or between analogue trunk and system dependant phone.*

(d) PTC 200 clause 4.11 (STMR), 4.12 (Instability), 4.13 (Acoustic shock protection), 4.14 (adjustable volume control), 4.15 (Telephone security in on-hook condition)

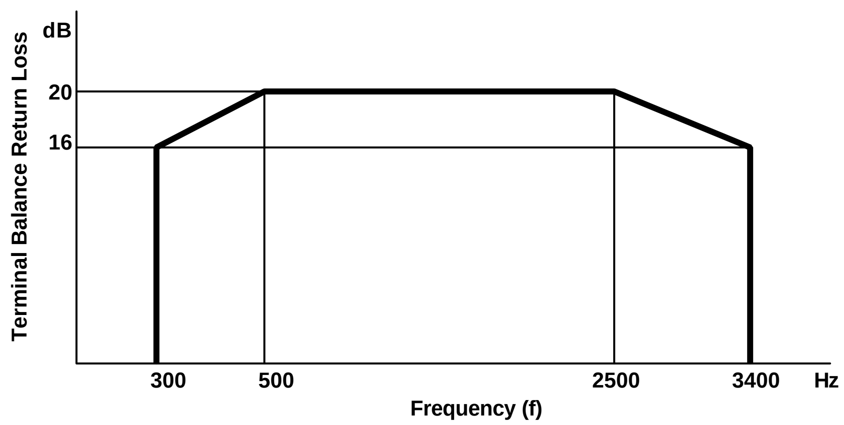
- *Applicable with system dependant phone only.*

(e) PTC 200 clause 4.16 (recorded message quality)

- *For in-built voicemail systems only.*

3.2
Analogue Port
PTC 200 clause 4.3 (noise & crosstalk), 4.6 (port impedance), 4.8 (Impedance balance ratio to earth)

Terminal Balance Return Loss (TBRL)
ref ITU-T Recommendation Q.552 clause 3.1.8
Using the test method described in Appendix 1 of this specification, the TBRL shall exceed the limits shown in the figure below.



4
Signalling
4.1
Analogue Network Interface

DTMF signals to be measured at Trunk interface shall meet the requirements of PTC200 Section 5.2

- *DTMF signals shall be generated by both standard extension phones as well as System Dependant phones.*

4.2
Analogue Port

DTMF Receivers in extension card to meet 5.3 with limits changed as follows:

Clause 5.3 (1) (b): change to +/- 1.8%,

Clause 5.3 (2): change on & off times to 55 ms

4.3
Digital Trunk

(a) Signalling shall comply with Q.931 or PTC108. Compliance with Q.931 shall be by compliance with CTR3 or CTR4 with supplementary tests as per Access Standards Newsletter no. 125 Appendix 1.

(b) DTMF digits to be mapped on to correct Q.931 messages during call setup, DTMF tones to be on appropriate B channel during call in accordance with PTC200 Section 5.2 except that levels shall be –6 to –15 dBm

5
d.c. characteristic

(a) Analogue network interface to meet PTC 200 clauses 6.3 to 6.8.

(b) Analogue port to meet requirements of PTC 200 clauses 6.10.

(c) System to meet requirements of PTC 200 clauses 6.11.

6
Ringing

(a) Analogue trunk card to meet PTC 200 clauses 7.5 &7.6

(b) Analogue Adapter card to meet requirements of PTC 200 clauses 7.7.4 & 7.7.5.

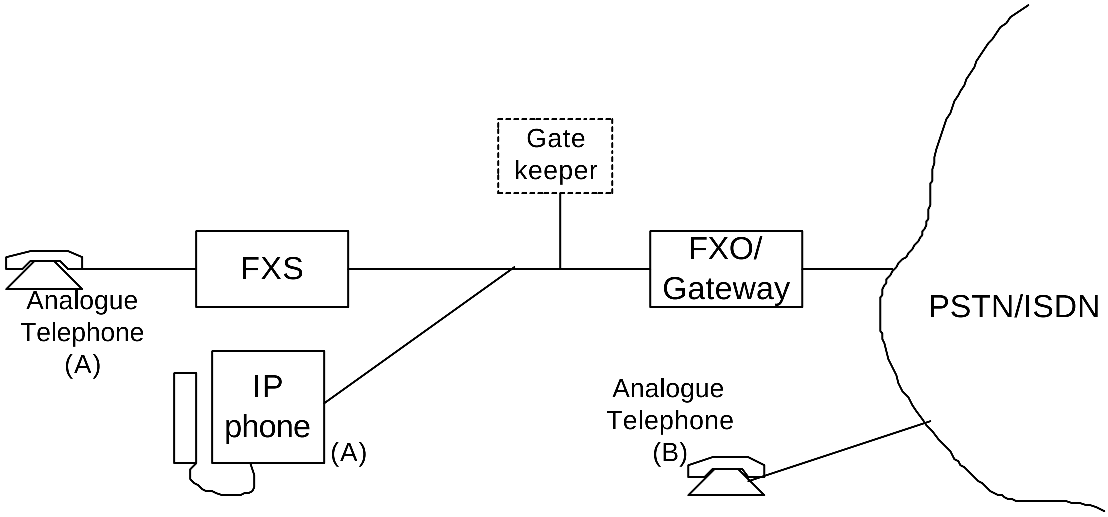
(c) Alerting device in IP Phone to operate or ringing voltage applied at 2-wire port of analogue adapter to be applied, within 500ms of ringing, or incoming call message being received by gateway

7
Automatic Call Set-up, answering & recording functions
System shall meet the appropriate clauses of PTC 200 Section 8

- *For in-built voicemail systems only.*

8
Functional Requirements
8.1
General
The aim of this set of tests is to check the functionality of a IP phone or analogue phone and analogue adapter/ Gateway combination. Basic call functionality is tested, more sophisticated functions which may be available are considered as marketing features.

8.2
Test Set-up
A standard analogue phone (A) is connected to the FXS module (analogue adapter) or, an IP phone is connected to the gateway (FXO module) which is in turn connected to the appropriate public network (ISDN/PSTN). If a gatekeeper is necessary for basic call control this should be included as part of the test set-up. A standard analogue phone (B) is connected to the PSTN.



8.3
Call from IP phone to PSTN/ISDN

8.3.1
Call Set-up
(a) Pre dial supervisory (dial tone) ………………………

- *Dial tone is optional, some systems may work on the basis of keying in the number followed by a send key.*

- *Dial tone is preferred, but user prompt could be a visual display message.*

(b) Signalling (correct number dialled at gateway/network interface) ………………

(c) Call progress tones received from network:
Ringing tone:………………..
Busy tone:…………………..
Disconnect tone:……………
NU tone:……………………..
Recorded announcments:……………….

(d) Bothway end to end audio path set up on cessation of ringing……………….
(e) If call aborted during call set-up, gateway to clear down network connection within 5 seconds: …………

8.3.2
Call in Progress
(a) Bothway audio connection maintained……………………

(b) Ability to generate in-band DTMF tones ……………………

- *See section 3 for electrical details*

(c) Switch-hook flash (to Telecom network) implemented correctly:……………….

- *Switch-hook flash through to the Telecom network is mandatory, it may be used for PBX features within the packet network.*

8.3.3
Call Clear (from Packet phone)
(a) Gateway clears network connection when phone hangs up……………

(b) Gateway clears network connection if phone disconnected or loses ability to function – e.g. loss of local power. Time limit for disconnection 15 seconds………………….

Timeout maybe up to 5 minutes provided the billing implications are made clear in the User Documentation:………………..

- *The longer timeout can be used to allow a phone to be disconnected from one location and re-connected at another location while a call remains active.*

8.3.4
Call Clear (from other party)
(a) Supervisory Tone (Disconnection Tone)…………………
Other supervisory method –e.g. visual………………….

- *A call clear indication is desirable but not essential. If a disconnect tone is not used, then the user should get new dial tone or some visual indication rather than have calls just end in silence.*

8.3.5
Incoming Call
(a) Incoming call alert (visual/audible) within 250ms of network signal to Gateway…………….

(b) Alerting signal to cease and bothway audio to be established within 100ms of pickup:………………….

- *Pickup may be picking up a handset or hitting a key for headset or handsfree operation*

(b) Caller ID implemented……………………….
Correct number displayed……………….
Dialback (if implemented) correct number
Insert zero for national toll calls ………….
Either remove area code or insert zero for local calls……………

8.4
Call to PSTN from phone (A) (via analogue adapter)

8.4.1
Call Set-up
(a) Pre dial supervisory (dial tone) presented to (A) within 200 ms of phone (A) going off-hook ……………………….

(b) Signalling (number dialled at phone (A) regenerated correctly at gateway/network interface) ……………….

- *Correct D channel message for ISDN or DTMF for PSTN gateway interfaces*

(c) Call progress tones received at phone (A) from network:
Ringing tone:………………..
Busy tone:…………………..
Disconnect tone:……………
NU tone:……………………..
Recorded announcments:……………….

- *These tones may be extended from the network or generated locally*

(d) Bothway end to end audio path set up on cessation of ringing……………….

(e) If call aborted during call set-up, gateway to clear down network connection within 5 seconds: …………

8.4.2

Call in Progress

(a) Bothway audio connection maintained……………………

(b) Ability to generate in-band DTMF tones:
phone (A) to network ……………………

network to phone (A) …………………...

- *See section 3 for electrical details*

(c) Switch-hook flash (to Telecom network) implemented correctly:……………….

- *Switch-hook flash through to the Telecom network is not mandatory, it may be used for PBX features within the packet network. If SHF does not operate correctly, a warning is required in the equipment User Manual stating that access to some Telecom services is not possible e.g. Call Waiting, Three Way calling, and other functions in a Centrex environment*

8.4.3

Call Clear (from phone (A))

(a) Gateway clears network connection when phone hangs up……………

(b) Gateway clears network connection if FXS module is disconnected or loses ability to function – e.g. loss of local power. Time limit for disconnection 15 seconds………………….

8.4.4

Call Clear (from phone (B))

(a) Supervisory Tone (Disconnection Tone) received at (A)…………………

8.4.5

Incoming Call (to phone (A))

(a) Ringing fed to analogue port within 250ms of network signal to Gateway…………….

(b) Ringing signal to cease and bothway audio to be established within 100ms of pickup by (A):………………….

(b) Caller ID implemented………………………
Correct number displayed on Telepermitted Caller ID unit……………

- *Caller ID display hardware required for test if this feature is implemented.*