SIN 377

Issue 1.7
July 2015

Suppliers' Information Note

For The BT Network

BT Hosted VoIP Services

Service Description

This SIN is available in Portable Document Format (pdf) from: http://www.btplc.com/sinet/

Enquiries relating to this document should be directed to: sinet.helpdesk@bt.com

CONTENTS

1. Introduction

This Suppliers Information Notification document provides information for the BT Hosted VoIP services, BT Multimedia VoIP and BT VoIP Port.

2. Product Overview

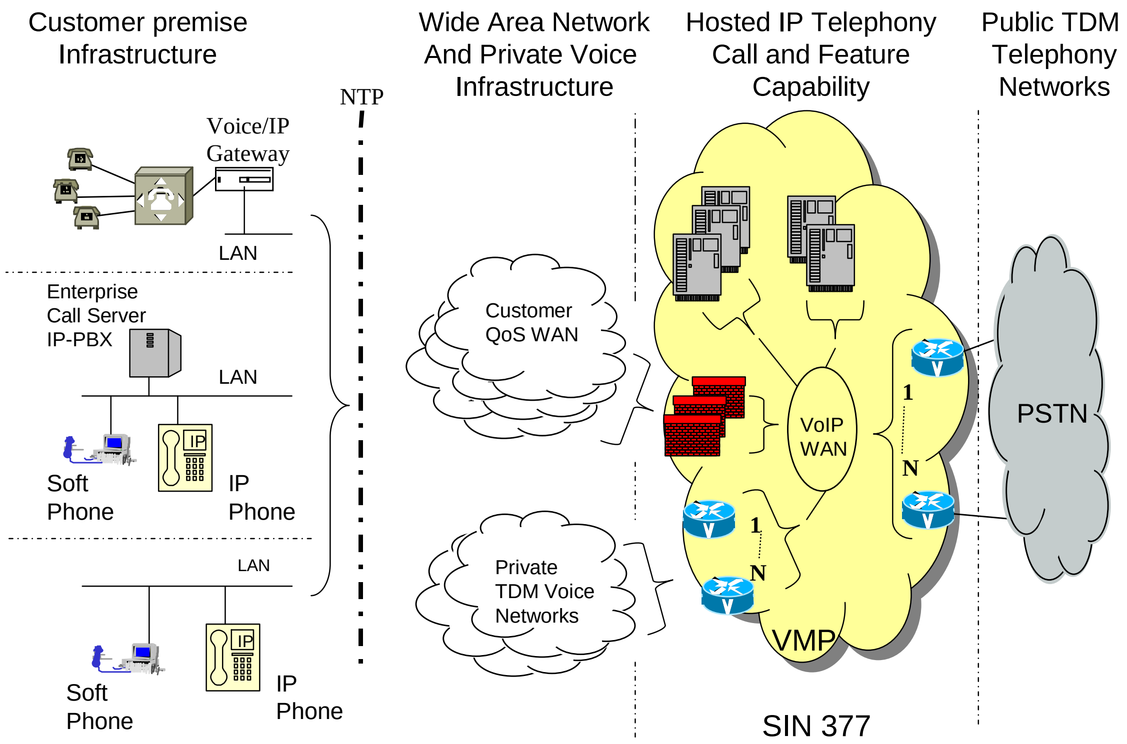
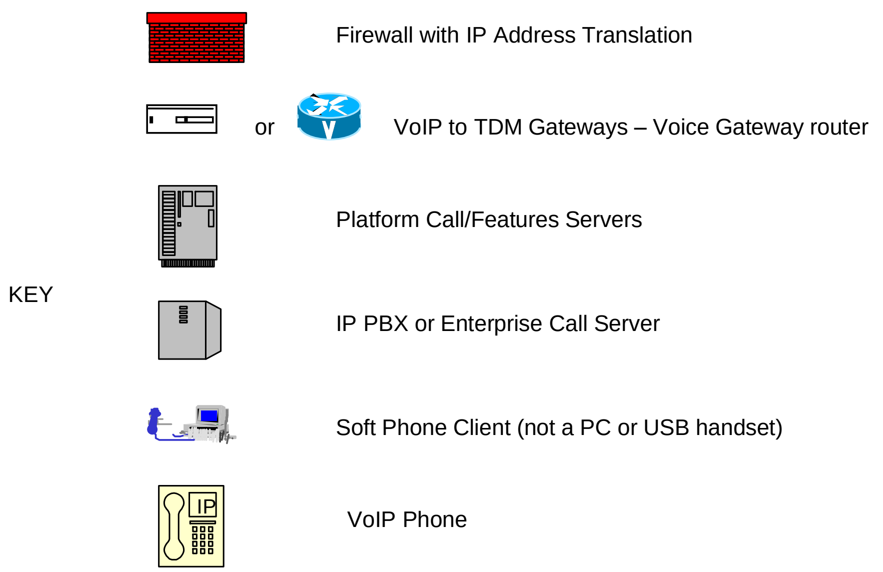


Figure 1 BT Hosted VoIP Service, Schematic



The BT Hosted Voice Over IP (VoIP) products provide the benefits of VoIP, multimedia and collaborative working to customers who want to further the capabilities of their existing data networks or IP infrastructure. BT Multimedia VoIP is a Network-Hosted Centrex type service, which is delivered by BT's VoIP & Multimedia Platform (VMP) and is available on a national basis. BT VoIP Port is a PBX connectivity product which compliments the BT Multimedia VoIP offering.

These services connect to a customer data network. Customers will be responsible for the provisioning and maintenance of their private data network to support the real-time VoIP services. It is preferable that private data networks use quality of service mechanisms in order to deliver VoIP data without delay to the voice packets, thus ensuring a desirable voice quality is experienced by the users. Call Servers within the BT VoIP & Multimedia Platform will provide call control to the voice and multimedia services enabling the voice terminals to use conventional E.164 telephone numbers.

Multimedia VoIP users will be able to “upgrade” any call in real time, to Multimedia, without having to start any additional programs or enter any additional information (such as end-user address). Multimedia, in this context, is defined to include the following real time end-to-end services:

- Video
- White-boarding
- File Transfer
- Document Sharing
- Application Sharing

More advanced features include click-to-call capability from PC-based directories. This includes both personal address books and corporate directories, where these are available to inter-work with the service. Customers can choose to combine the Multimedia benefits of the PC and the voice capabilities of the IP phones. By controlling the IP phone from the PC, users can make use of directory-enabled calling, use the IP phone for voice, and seamlessly interact with their PC for more advanced features such as multimedia.

The BT Multimedia VoIP service will also enable different customers who are hosted on its service to hold Multimedia calls between one another. Additionally, the VoIP & Multimedia Platform will inter-work with Voice VPNs (TDM based private voice networks) providing an evolution plan for customers who already utilise these voice capabilities.

Initially connection to all other PTO networks will be achieved using existing BT Wholesale C7 Interconnect links enabling BT Hosted VoIP customers to communicate (voice only) with customers of any other operator. Calls to the PSTN will be delivered using standard circuit switched signalling.

3. Abbreviations & Terms

Call Manager	Controls and manages users real time voice and multi-media applications.
Centrex	A service provided by the public telecommunications operator as an alternative to customers having their own Private Branch Exchange (PBX), with the service providing the switching of calls between terminals on the customer's site as well as calls to and from other users of the PSTN.
Dial Plans	These are only applicable to the integration of PABXs into the service offering. As with traditional voice networks, BT Hosted VoIP customers will also have a private number plan. In traditional networks, the private dial plan is made up of 2 key components; an SLC (site location code) and an extension number. The SLC is used to identify the location of the site where the extension exists. The extension digits define the termination within the site. BT Hosted VoIP follows these guidelines. For BT Hosted VoIP, each BT MM VoIP user will have an on-net extension number. Where necessary, to allow interworking, users will have an SLC.

Extra-net Calling	Extra-net (IP) calls can be made to other users within other customer’s networks connected to the BT Voice & Multimedia Platform. These calls utilise the feature-sets (Table 1) that are implemented through the use of the H.323 protocol.
IP	Internet Protocol.
MS TAPI	MICROSOFT Telephony Application Programmers Interface.
Off-net Calling	Full PSTN break-in and break-out is available as part of the service. These calls utilise the feature-sets (Table 1) that are implemented through the use of the H.323 protocol. Calls to the PSTN are voice services only, the multi-media aspects of H.323 are not currently supported within the PSTN.
On-net Calling	On-net (IP) calls can be made to other users within the customer's network. These calls utilise the feature-sets (Table 1) that are implemented through the use of the Cisco Skinny or H.323 protocols. On-net calls can include voice calls to a PABX whose numbering plan is integrated with the service, functionality is then limited to that shown in Table 1.
NTP	Network Termination Point.
PABX	Private Automatic Branch Exchange.
PSTN	Public Switched Telephone Network.
QoS	Quality of Service.
SNMP	Simple Network Management Protocol.
SoftPhone	For BT Multimedia VoIP, an IP SoftPhone can be used as a stand-alone telephone to place and receive all calls through a computer rather than an IP Phone. In this mode, the media stream is terminated by the computer. This means that the computer "rings" in response to an incoming call. A computer headset or computer handset will be required to speak and listen during a call. The microphone and speaker volume is controlled by the volume controls on the IP SoftPhone user interface. To use an IP SoftPhone as a stand-alone telephone, a PC must have a full-duplex sound card with the proper drivers installed, and a computer headset or computer handset.
TDM	Time Division Multiplex.
VMP	VoIP & Multimedia Platform.
VoIP	Voice over Internet Protocol.
VPN	Virtual Private Network - these are used by a company or private group to make inter-site connections either for telephone speech or data as if there were dedicated leased lines between these sites. The equipment used is located within the public telecommunications operators’ premises and forms an integral part of the public network, but is software- partitioned to allow for a genuinely private network.
™	indicates it is a trade mark.
®	indicates it is a registered trade mark.

4. Service Availability

The service is available (where capacity exists) anywhere within the UK, at locations where the customer has a private data network.

5. BT NTP Interface

5.1 Physical Interfaces

5.1.1 ATM
The ATM interface used for the delivery of this service has all the attributes of the BT CellStream service, technical details of the ATM interface can be found in SIN 264 (current issue available from http://www.btplc.com/sinet/).

5.1.2 Ethernet

5.1.2.1 BT Internet Protocol Transport Services
Delivery will also be available using the BT QoS enabled wide area network services, offering an Ethernet interface as described in SIN 302 BT Internet Protocol Transport Services and SIN386 IP Clear. (Current issues available from http://www.btplc.com/sinet/).

5.1.3 Voice VPNs and Retail Interface for Messaging Service Providers

5.1.3.1 Featurenet, Embark, and PABX interworking
Customers using Featurenet (SIN 357) or Embark (SIN324) will be able to integrate their Voice VPN networks and private numbering plans with sites connected to the BT Hosted VoIP Services. This will allow the two networks to appear as one virtual network for each individual customer.

5.1.3.1.1 2048 kbit/s Digital Private Network Signalling System No 1 Interface

The gateway will support E1, 2048 kbit/s, G.703 interface with G.704 framing with either RJ45 120 Ohm or BNC 75 Ohm connection. The signalling is compliant to BTNR188 Issue 6 - including Update Sheet Sept. '96, see Annex B.

5.1.3.1.2 PABX connectivity

5.1.3.1.2.1 2048 kbit/s(E1) G.711 (A law) ETSI PRI interface
The gateway will support E1, 2048 kbit/s, G.703 interface with G.704 framing with either RJ45 120 Ohm or BNC 75 Ohm connection.

Signalling is compliant to ETS 300 403-1/C1 [ITU-T Q.931] (see http://www.btplc.com/sinet/ SIN 261 for further, more detailed references).

5.1.3.1.2.2 2048kbit/s Digital Private Network Signalling System no.1
The gateway will support E1, 2048kbit/s, G.703 interface with G.704 framing with either RJ45 120 Ohm or BNC 75 Ohm connection. The signalling is compliant to BTNR188 Issue 6 - including Update Sheet Sept. '96, See annex B.

5.2 Call Control Protocols

The BT Multimedia VoIP and BT VoIP Port services supports four customer interface types which interwork with IP Phones and Softphones, and for BT VoIP Port gateway devices such as Q.931, these are:-

- Cisco proprietary, Skinny terminal access,
- H.323 Version 2,
- H.323 Version 2 with SNMP,
- Microsoft TAPI client.
- Media Gateway Control

5.2.1 Cisco Skinny Protocol

The " Skinny terminal access" document (see doc refs) details the message set, and includes sample message sequence charts for registration and call control, the final part of this document is a description of the message set.

To allow other vendors to build equipment to work on the service, Cisco Systems have agreed to licence their Intellectual Property Rights on fair and reasonable terms to any applicant.

5.2.2 H.323

H.323 is a sophisticated specification published by the ITU-T. It includes references to several further specifications relating to registration of end points, admission of calls to a network with the purpose of managing bandwidth, call status, multi-media call control, video and audio encoding, and the transport of information streams over the chosen physical layer.
This service will use the gatekeeper-routed model, whereby all calls will be signalled to the BT gatekeeper (Call Manager) for onward routing.
All calls will use the E.164 style of addressing. These addresses will be translated by the gatekeeper into the IP address of the called party.
As H.323 is an International standard there are many clients that are capable of using this service. However, their level of sophistication is considered to be uneven and consequently the BT Hosted VoIP service will only support simple call set up (see Table 1).

Station to Station Dialling	√
Caller ID – Number	√
Caller ID – Name	√
Direct Dialling Inward	√

Table 1 H.323 Simple Call Set-up

The H.323 documentation is published and can be found at

http://www.itu.int/publications/telecom.htm - ITU-T Recommendations Online

5.2.2.1 H.323 between IP PBX/Gatekeeper and BT V&MPGateKeeper

- An IP PBX/GW and a BT V&MP GateKeeper, will view each other as peer H.323 entities. Therefore, neither the IP PBX/GW or the BT V&MP GateKeeper will be required to support the RAS procedures across this interface, as defined in the ITU-T Recommendation H.225.
- Both the IP PBX/GW and the BT V&MP GateKeeper will use the Gatekeeper routed model when initiating H.225 call signalling, as defined in H.323v2 section 7.3.1.
- Both the IP PBX/GW and the BT V&MP GateKeeper will keep the Call Signalling Channel open for the duration of the call, as defined H.323v2 section 7.3.1.
- Both the IP PBX/GW and the BT V&MP GateKeeper will ensure that H.245 Control Channel messages are routed between themselves, as defined H.323v2 section 7.3.2 and not directly between the endpoints.

5.2.2.2 H.323 Call Establishment

- The call establishment procedures will be as defined in H323v2 Figure 24. Routing is performed via the E.164 Called Party number analysis.
  Note:- As is normal practice in private networks; private network Gatekeepers will have a numbering plan covering internal/local extensions, and for the BT V&MP Gatekeeper (public network). Calls not deliverable within the private network will be passed to the BT V&MP Gatekeeper for processing.
- Consequently, this SIN does not require the use of RAS procedures between the IP PBX/GW and the BT Gatekeeper. Therefore, the LRQ/LCF messages are not required.

5.2.3 H.323 Version 2 with SNMP

It is planned to also offer a managed primary rate (E1) Q.931 interface that allows a customer sited PBX to be included in the customer network environment.

Management will be carried out using SNMP, IETF RFC 1902. SNMP is an IETF RFC, SNMPv2 is described in RFC 1902, "SNMP is an application-layer protocol that facilitates the exchange of management information between network devices - and is part of the Transmission Control Protocol/Internet Protocol (TCP/IP) protocol suite."

5.2.4 Microsoft TAPI

The client software supplied with this product uses the MICROSOFT ™ Telephony Application Programmers Interface (TAPI) architecture. The multi-media client running on the user's PC is able to make and receive voice and video calls, and in conjunction with NetMeeting can share documents (ref. H.323 and T.120). It is also possible to use the multi-media client to exercise third party control over an associated IP telephone.

Remote terminal services can be created by third parties using the TAPI interface.

5.2.5 Media Gateway Control

Media Gateway Control Protocol is used to control the customer premise located Voice Gateway router. In conjunction with MGCP, SCTP (DUA classes 0 and 1) are used to backhaul DPNSS signalling to and from the BT VMP Call Agent. These protocols are used to convey DPNSS messages end-to-end between PBXs connected to customer premise Voice Gateway routers. These gateways need to be compliant to the following specifications. See Fig. 1 for location of the Voice Gateway.

- Media Gateway Control Protocol (MGCP) v1.0: RF3435
  http://www.zvon.org/tmRFC/RFC3435/Output/index.html

Stream Control Transmission Protocol (SCTP): RFC2960 http://www.ietf.org/rfc/rfc2960.txt

DUA classes 0 and 1 http://www.ietf.org/internet-drafts/draft-ietf-sigtran-dua-05.txt

Annex B describes the compliance to DPNSS BTNR 188 provided by the BT VoIP Port using MGCP v1.0

5.3 Voice and Fax

Voice will be conveyed by the Real Time Protocol (RTP) see IETF RFC 1889.
Voice codecs conforming to G.711 and G.729 will be supported codecs.
Group 3 Fax machines will be supported via Gateways (BT VoIP Port product) using the IT-U T.38 protocol.

6. Customer Data Network, QoS requirements

The BT Hosted VoIP products allow customers to converge their voice and data on to a single network.
To support real-time voice and multimedia on the BT Hosted VoIP services, there are a number of requirements on the LAN / WAN infrastructure, which are:

- end-to-end delay less than 60ms
- packet loss less than 1%
- packet delay variation of less than +/- 10ms (i.e. packet delay within 10ms of the average)

Exceeding these parameters is likely to cause speech degradation, and it is therefore important that a suitable LAN / WAN infrastructure is in place.

Due to the diversity of customer's data infrastructure it is not possible to give generic requirements. BT offers a process for interested customers, which includes two stages of assessment of the LAN and WAN.

The first stage will be a paper-based health-check that will be made available, without charge, to allow customers to assess the suitability of their existing infrastructure. This will indicate any major issues, and provide indicative requirements for any alterations to their network. Information discussing QoS requirements can be found at http://www.cisco.com/univercd/cc/td/doc/product/voice/ip_tele/avvidqos/index.htm

The second stage will be a physical assessment, providing a more detailed assessment report, and recommending alterations where necessary. This will be a chargeable option.

7. Features and Functionality

7.1 Multimedia

Multimedia, in the context of this service, is defined to include the following real time end-to-end services:

* Video * Whiteboarding * File Transfer
* Document Sharing * Application Sharing

"Document sharing" and "Application sharing" are also referred to as "Data Collaboration" services. It is envisaged that all these services require software on the PC / laptop. Video will require additional hardware (PC camera). Multimedia will only be available, to those users who are subscribed to the service.

7.2 Service Features and Functionality

Functionality	**Cisco Skinny Protocol**		**H.323 Protocol**
	IP Phone	**Soft Phone***	**Soft Phone***
Call Forward – Busy	√	√	
Call Forward – No Answer	√	√	
Call Forward – Unconditional	√	√	
Call Hold and Retrieve	√	√	
Call Park	√	√	
Call Pickup Group – Directed	√	√	
Call Pickup Group – Universal	√	√	
Call Status per Line – State, Duration, Number	√	√	
Call Transfer – Blind	√	√	
Call Transfer – Supervised	√	√	
Call Waiting / Retrieve	√	√	
Caller ID – Name			
Caller ID – Number	√	√	√
Centralised Configuration	√	√	
Click to Dial from Web Browser	√		
Data Collaboration, White Boarding, File Transfer	√**		√
Direct Dialling Inward	√	√	√
Distinctive Ringing (Internal versus External)	√	√	
Distinctive Ringing per Phone	√	√	
Hands-free Full Duplex Speakerphone	√	√	
HTML Help Access from Phone	√	√	
Message Indicator	√	√	
Multi-Line Support	√	√	
Network based Voice messaging	√	√	√
Privacy	√	√	
QoS Statistics available at the Phone	√		
Single Directory Number for Multiple Phones	√	√	
Speed Dialling	√	√	
Station to Station Dialling	√	√	√
Video	√***	√	√
Voice Mail Integration	√	√	

* Soft Phones are PC based client implementations that allow the use of a PC as an integrated computing / voice communications package.
** IP Phone needs a softphone controlling the IP Phone
*** With suitable video-equiped IP Phone

Table 2 Service Features and Functionality

8. Document References

Document Ref.	Document Title	Document Source
BTNR188	DPNSS Digital Private Network Signalling System No. 1. - Issue 6 including Update Sheet Sept. '96	BT
SIN 261	BT ISDN 2e and ISDN 30e Services using full ETSI Call Control	BT
SIN 264	BT CellStream Service Description	BT
SIN 324	Featurenet Embark Service Description	BT
SIN 357	BT Featurenet™ Service Description	BT
Skinny terminal access	Cisco proprietary, Call Management Protocol	Cisco
RFC 1889	RTP: A Transport Protocol for Real-Time Applications	IETF
RFC 1902	SNMP (Simple network management Protocol) is an application-layer protocol that facilitates the exchange of management information between network devices - and is part of the Transmission Control Protocol/Internet Protocol (TCP/IP) protocol suite.	IETF
RFC 3435	MGCP 1.0 – Media Gateway Control Protocol	IETF
G.711	Recommendation G.711 (11/88) - Pulse code modulation (PCM) of voice frequencies. Appendix I (09/99) to Recommendation G.711 - A high quality low-complexity algorithm for packet loss concealment with G.711. Appendix II (02/00) to Recommendation G.711 - A comfort noise payload definition for ITU-T G.711 use in packet-based multimedia communication systems	ITU
G.729	Recommendation G.729 (03/96) - C source code and test vectors for implementation verification of the G.729 8 kbit/s CS-ACELP speech coder	ITU
H.323	Recommendation H.323 (11/00) - Packet-Based Multimedia Communications Systems	ITU
Q.931	Recommendation Q.931 (05/98) - ISDN user-network interface layer 3 specification for basic call control	ITU
T.38	Procedures for real-time Group 3 facsimile communication over IP networks	ITU
T.120	Recommendation T.120 (07/96) - Data protocols for multimedia conferencing	ITU

These documents may be obtained from the following sources:

BT	http://www.btplc.com/sinet/
Cisco	-Andy Oldfield on aoldfield@cisco.com or Tel +44 (0)20 8824-4048 OR -contact Cisco General enquiries Tel +44 (0)208 824 1000 request UK Voice Technology Group prime
IETF	Http://www.ietf.org
ITU	Http://www.itu.org or Http://www.itu.int/publications/telecom.htm - ITU-T Recommendations Online

9. History

Issue 1	12 February 2001	First Published.
Issue 1.1	04 April 2001	Addition of section 6, QoS requirements.
Issue 1.2	24 July 2001	Support for Featurenet / Embark / PABX interworking & additions to clarify the interworking ability of the service.
Issue 1.3	12 July 2002	Addition of sections 5.2.2.1 & 5.2.2.2 Figs 1 & 2 updated to show IP PBXs in customer environment.
Issue 1.4	15 January 2003	Modification to Cisco contact details in section 8.1.
Issue 1.5	3 July 2003	Introduction of Media Gateway Control Protocol & T.38 support. Brand name and editorial changes.
Issue 1.6	26 August 2005	Update MGCP spec, refresh references, and update DPNSS annex.
Issue 1.7	July 2015	Change SINet site references from http://www.sinet.bt.com to http://www.btplc.com/sinet/

Annex A: Interworking Functionality

The service supports both H.323 V2 and Cisco Skinny protocol. The information below illustrates the features and functionality available and supported for the Service:-

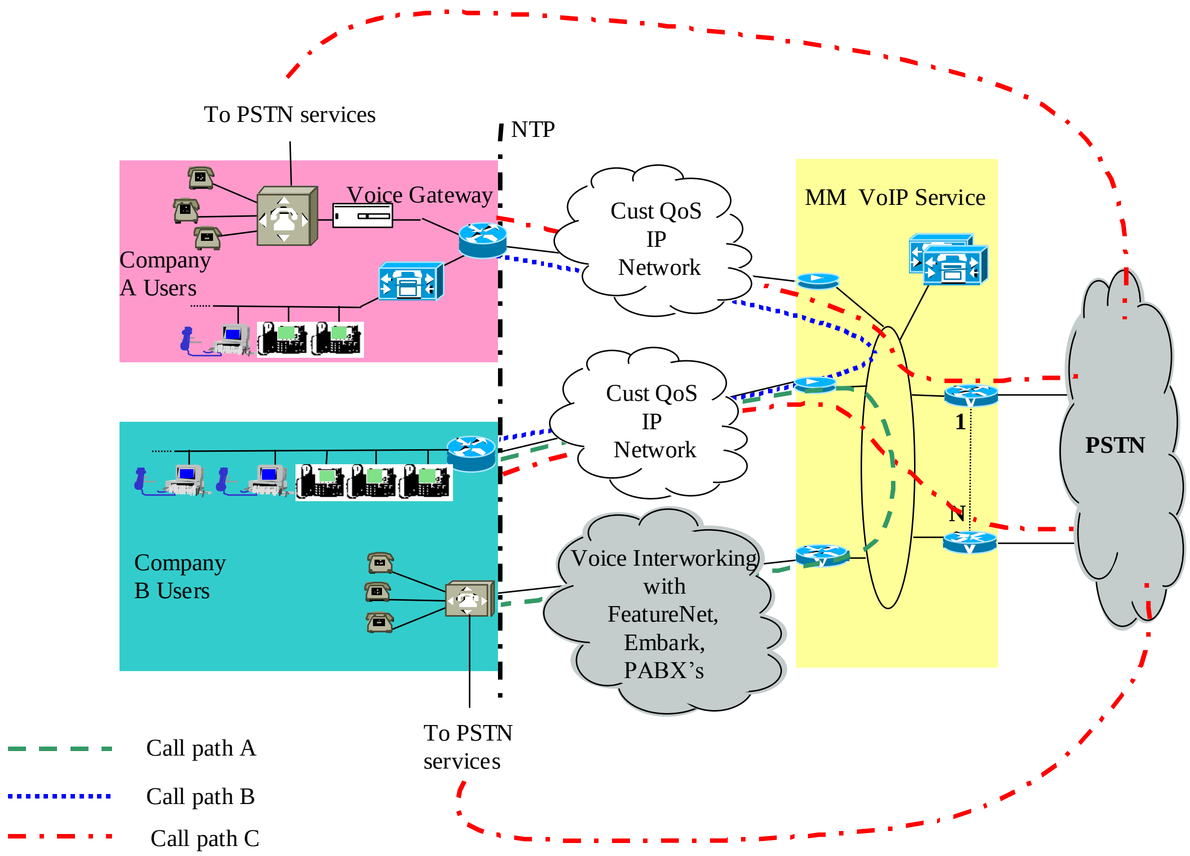


Figure 2 Interworking Functionality, Call Paths

Call path A:- This call path allows IP based users to interwork (at a voice level, not multimedia) with users on TDM based PABXs used by the same company, this facilitates integration of systems as companies upgrade their systems.

Call path B :- This call path allows interworking of IP based users between company A and company B, the level of interworking depends upon the terminal equipment used, it is limited to the functionality indicated in Table 2 Service Features and Functionality under the column "H.323 protocol".

Call path C :- This call path allows users access to PSTN services and users.

In the above diagram a "Company" could be single-sited or multi-sited.

On-net Calling (call path A)	On-net (IP) calls can be made to other users within the customers network, i.e. within customer B. These calls utilise the feature-sets (Table 2) that are implemented through the use of the Cisco Skinny or H.323 protocols. This can include voice calls to a PABX whose numbering plan is integrated with the service, functionality for these calls is then limited to that shown in Table 1.
Extra-net Calling (call path B)	Extra-net (IP) calls can be made to other users within other customer's networks connected to the "BT Hosted VoIP Service" network, i.e. between customer A and customer B. These calls utilise the feature-sets (Table 2) that are implemented through the use of the H.323 protocol
Off-net Calling (call path C)	Full PSTN break-in and breakout is available as part of the service, i.e. from customer A or B to or from the PSTN. All IP phones have an E.164 number allocated to them that is dial-able by any telephony user These calls utilise the feature-sets (Table 1) that are implemented through the use of the H.323 protocol. Calls to the PSTN are voice services only, the multi-media aspects of H.323 are not currently supported within the PSTN

Annex B: BT VoIP Port DPNSS Conformance to BTNR 188

The table below indicates feature support for PBXs connected via customer premise gateways to VoIP Port.

- Column 3 - a PBX E1 interface supporting DPNSS connected to a customer gateway and then onward using VoIP Port protocol MGCP (see para. 5.2.5) via a WAN (and VoIP Port Call Agent) to another customer gateway and PBX.
- Column 4 - As above but interworking via a gateway to Featurenet (See SIN357).
- Column 5 - As above to PSTN
- Column 6 - As above from a PBX connected via a gateway to BT VoIP Port and onward via a WAN to a BT Multimedia VoIP termination.

Chapter	Feature Name	PBX to PBX	PBX to FN5000	PBX to PSTN	PBX to MMVoIP
1-5		C	C	C	C
6	Simple Telephony Call	C	C	C	C
7	Data Call	NS[1]	NS[1]	NS[1]	NS[1]
8	Swap	NS[1]	NS[1]	NS[1]	NS[1]
9	Call Back When Free	C	C	NS	NS
10	Executive Intrusion	C	C	NS	NS
11	Diversion	C[2]	PC[8]	PC[5]	PC[5]
12	Hold	C	C	C	C
13	Three Party	C[2]	C	PC[6]	C
14	Call Offer	C	C[8]	NS	PC[9]
15	Non Specified Information	C	PC[8]	NS	NS
16	Service Independent Strings	C	PC[8]	NS	NS
17	Call Waiting	C	NC[8]	NS	PC[10]
18	Bearer Service Selection	NS[1]	NS[8]	NS	NS
19	Route Optimisation	C[2]	C	C[13]	NS
20	Extension Status	C	NS	NS	NS
21	Controlled Diversion	C	NC	NS	NS
22	Redirection	C	C[8]	NS	NS
23	Series Call	C	NC[8]	NS	NS
24	Three Party Takeover	C	NS	NS	NS
25	Night Service	C[2]	NC[8]	NS	PC[11]
26	Centralised Operator	C	PC	NS	PC[12]
27	Traffic Channel Maintenance	NS	NS	NS	NS
28	Remote Alarm Reporting	C	NC	NS	NS
29	Add-On Conference	C	NC[8]	NS	C[14]
30	Time Synchronisation	C	NS	NS	NS
31	Call Back When Next Used	C	NS	NS	NS
32	Do Not Disturb	C	NS	NS	NS
33	Remote Registration of Diversion	C	NS	NS	NS
34	Remote Registration of DND	C	NS	NS	NS
35	Priority Breakdown	NS[3]	NS	NS	NS
36	Call Back Messaging	C	NS	NS	NS
37	Loop Avoidance	NS[4]	NS	NS	NS
38	Forced Release	C	NS	NS	NS
39	Text Message	C	NS	NS	NS
40	Charge Reporting	C	NS	NS	NS
41	Network Address Extension	C	NS	NS	NS
42	Call Park	C	NS	NS	NS
43	Call Distribution	C	NS	NS	NS
44	Route Capacity Control	NS	NS	NS	NS

45	Wait on Busy	**C**	**NS**	**NS**	NS
46	Call Pickup	**C**	**NS**	**NS**	NS
47	Travelling Class of Service	**C**	**NS**	**NS**	NS
48	Number Presentation Restriction	**C**	**C**[8]	**PC**[7]	PC[7]

C Compliant
PC Partially Compliant
NS Not Supported
NC Not Compliant

Notes

1. Non telephony SICs are not supported because only voice and G3 fax traffic are supported over IP.
2. RM messages will transit the network, but will not branch in the network.
3. Priority breakdown information is passed transparently, but the network does not lower priority calls on congestion.
4. This information is passed transparently but loop counters are not decremented
5. Diversion service supports Divert on Immediate, Busy and No Answer via separate channel. Diversion via the same channel is not supported. Follow Me diversion is not supported. Diversion bypass is not supported.
6. The Three Party service is mainly informational, apart from AD-V which is acknowledged allowing conferences to include PSTN and MM-VoIP parties.
7. Supports restriction of Calling Party Number only via NPR-A parameter.
8. See FeatureNet caveats in Annex A of SIN 357 for PBX to FeatureNet calls.
9. Call Offer currently maps to Call Waiting.
10. The called party is given an indication that there is another incoming call to his line. The called party can accept or reject the new incoming call. This is a called-party service.
11. Only supported on the DPNSS PBX side
12. This feature is not supported from CCM to DPNSS PBX). The Centralized Operator must be in the DPNSS network. Further, Series Call, Controlled Diversion and Executive Intrusion are not supported. Also, Three Party Take Over will only be available between DPNSS PBXs, not between DPNSS PBX and CCM.)
13. Route Optimisation is available for VoIP Port call cases, but not VoIP Port/Call Manager cases
14. Add-on Conference – Permits a conference call controller to add more parties (four or more). All parties on the call can make a further enquiry call and add the enquiry call party on to the conference.

< END >

