INTECHNOLOGY

INTECHNOLOGY'S SIP TRUNKING SERVICE DEFINITION

SD032 v1.8
Issue Date 12th Sept 2013

inTechnology
MANAGED SERVICES

Revision Control

Revision	Author	Change	Date
SD032 V1.2	Phil Dyson	Updates after review	15th April 2011 August 2011
SD032 V1.3	Paul Storey	Added network diagrams, removed reference to PBX Connect ISDN service	18th August 2011
SD032 V1.4	Paul Storey	Updated Network Architecture Diagram. Added Billing and Bandwidth Calculation Table.	31st October 2011
SD032 V1.5	Paul Storey	Added number porting information	15th November 2011
SD032 V1.6	Paul Storey	Added note to Section 4	21st March 2013
SD032 V1.7	Paul Storey	Removed SLA and Service Credit sections	26th July 2013
SD32 V1.8	Paul Storey	Updated Section 5 Charging Policy	12th Sept 2013

Related Documents

Reference	Title and Purpose	Author	*Last version*
[1]	Interoperability Test Plan	S. Golding	V1.0

Scope

This document provides a technical overview of the InTechnology's SIP Trunk service "PBX Connect SIP". The document will describe the relevant network topology, network elements, supported service features and limitations.

The intended audience for this document includes InTechnology Sales, Support, Engineering, Product Management and contracted InTechnology customers.

SECTION 1: OVERVIEW

SERVICE OVERIEW

PBX Connect SIP is a Voice over IP (VoIP) service providing IP based connectivity to customers with existing IP PBXs. The service aims to provide full PSTN equivalence, delivered across InTechnology's managed IP network. The SIP trunk service is delivered from two centrally managed, software based, IP Voice Switches hosted in InTechnology's highly secure and resilient Data Centre environment.

The service is sold and billed as a combination of a onetime service activation fee and a per-channel monthly rental. The customer needs to define in advance the number of discrete concurrent voice channels required.

TARGET MARKET & BENEFITS

PBX Connect SIP is primarily targeted at customers that have PBXs capable of supporting direct IP connectivity and are looking to completely replace their existing ISDN BRI or PRI circuits. InTechnology's SIP Trunking service enables organisations to run both voice and data services via a single converged network connection, maximising investments made in their existing telephony equipment.

The PBX Connect SIP should be distinguished from InTechnology's "PBX Connect ISDN" product(s), whereby customers connect into InTechnology's VoIP network via a Q.931 PRI/BRI gateway.

PBX CONNECT BENEFITS

- The ability to make and receive calls to and from the PSTN (Public Switched Telephone Network) via InTechnology's managed IP network and Tier-1 interconnect partners.
- Flexibility – PBX Connect SIP can be increased delivered in increments of one after initial installation. The only limit is the customer bandwidth.
- Cost reduction through the reduction or removal of existing ISDN rental charges
- Free on-net calls between the organization's sites connected using the service
- Number porting from a number of service providers
- A Platform to deliver additional UC services.
- Improved network redundancy – InTechnology's SIP trunk service can be configured to deliver multiple levels of failover including (Trunk, Trunk Group, or user), providing customers with a level of resilience typically not implemented in ISDN connected solutions.
- Competitive tariffs, helping customers reduce ongoing call costs.

SECTION 2: SERVICE DESCRIPTION

Figure 1 details the typical enterprise SIP Trunk deployment models supported. Depending on the customer type and existing security policies, the customer may/may not choose to deploy their own enterprise Session Border Controller (SBC).

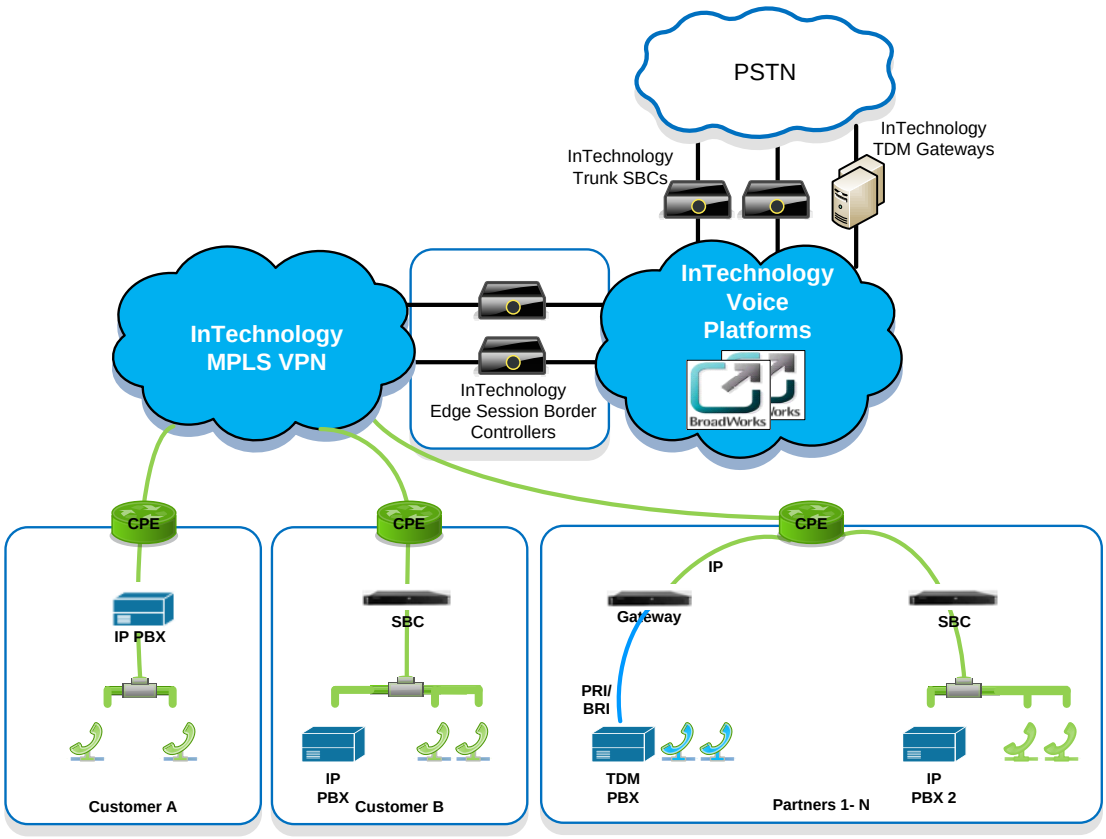

Figure 1: PBX Connect SIP Service Architecture

INTECHNOLOGY VOICE SWITCHES

InTechnology's PBX Connect SIP trunk service is delivered from two centrally managed, software based, IP Voice Switches hosted in InTechnology's highly secure and resilient DC (Data Centre) environment.

The IP Voice Switch technology is based on tried and tested SIP (Session Initiation Protocol) open standards software providing a highly, scalable, cost effective and future proof platform for the provision of IP voice services.

InTechnology's IP Voice platform has been deployed in a highly available configuration across InTechnology Data Centre locations in Harrogate and London. The core IP Voice Switches are maintained and supported by InTechnology 24 / 7 and access to the service is delivered across InTechnology's national IP MPLS Network and via a range of connection technologies including DSL, PPC leased lines and 10Mbps, 100Mbps and 1Gbps Ethernet as required.

SESSION BORDER CONTROLLER (SBC)

InTechnology have deployed clusters of high availability "Edge" Session Border Controllers (SBCs) to act as the points of service interconnect between the customer and InTechnology's SIP trunk service. The SBC acts a point of network demarcation, performs NAT/PAT translation and SIP signalling interoperability.

NETWORK CONNECTIVITY

PBX Connect SIP is delivered over InTechnology's IP MPLS network "LANnet". In order to assure call quality and performance across the network, InTechnology have implemented Quality of Service (QoS) systems throughout the LANnet network.

A direct connection from the customer site to the nearest LANnet Point of Presence (POP) is required to deliver the service. These connections are available in bandwidths from 1Mbps to 1Gbps.

NETWORK FAILOVER

SIP trunk failover can only be implemented as standard with InTechnology provided circuits. Due to the diverse nature of client networks any failover of the SIP trunk to another service provider's network will need a level of professional services design work to be carried out, and as such would require commercial coverage to be in place prior a design being produced.

PSTN CONNECTIVITY

For external calls, the service includes centralized, low-cost PSTN breakout from InTechnology's IP Voice switches. InTechnology's high speed IP and TDM PSTN interconnects enable callers to make and receive PSTN calls.

EMERGENCY CALLS

Emergency 999 calls may be made using the service where supported by the customer's telephony equipment and dial plan. The customer is responsible for ensuring that InTechnology are provided accurate caller location information.

SECTION 3: DEPLOYMENT PROCESS

It is necessary to ensure that Customer Premises Equipment (CPE) is compatible with the service. In order to do this, an audit of the PBX equipment may be necessary. Any reconfiguration required for the PBX to work with the service must be undertaken by the customer or their appointed contractor. InTechnology can provide PBX reconfiguration or PBX support services via a partner should the customer require PBX configuration support.

PBX INTEROPERABILITY TESTING

Due to the proprietary variations of the SIP standard that exist across different vendor platforms, InTechnology operates an interoperability test process for each IP-PBX that requires connecting to the SIP trunk service.

This process has been implemented to ensure that a customer's IP-PBX operates in accordance with InTechnology's Interoperability test plan. [1]

TESTING PROCESS

1. The customer must configure their PBX to route some non-business critical, test calls via the provided SIP trunk. The configuration details of the SIP trunk will be provided by InTechnology. (The SIP trunk provided for the purposes of interoperability testing will be with temporary phone numbers only.)
2. The customer may use an InTechnology LANnet connection or a suitable Internet connection of sufficient bandwidth as connectivity for the purposes of the interoperability testing.
3. An InTechnology technical consultant will liaise with an appointed customer contact to carry out the InTechnology Interoperability test plan and subject to successful testing, sign off the test plan with the customer.

InTechnology will aim to complete the interoperability testing process within 30 days.

During the interoperability testing phase:

- The customer **will not** be charged any monthly service charges.
- The customer **will be** charged for any chargeable calls made.

Whilst every effort will be made, InTechnology offers no assurances that all IP-PBXs will work successfully with the service. In the unlikely event that an IP-PBX cannot successfully operate with the service in accordance with the InTechnology test plan, the customer may cancel the service during the first 30 days by providing InTechnology with written notice. If such notice is not received, the full minimum contract term will become applicable.

Note: the customer will be responsible for all configuration works required on the PBX.

DECOMMISSIONING PROCESS

Upon expiration of the service contract where the customer chooses not to renew with InTechnology, the following steps are followed as part of the decommissioning process:

Phase 1: Contractual
Expiration of the service contract and a decision by the customer not to renew. May also include early termination by the customer, subject to payment of early termination fees.

Phase 2: Service decommission
Cessation of the service and removal of user accounts, user access and administrative access from the platform.

Phase 3: Number porting
Porting of any telephone numbers from InTechnology to the gaining party where this party has a porting agreement with InTechnology.

SECTION 4: SERVICE COMPONENTS

OFFICE CONNECTIVITY

The PBX Connect SIP service is delivered over an InTechnology LANnet connection. If a LANnet connection is already installed, it's usually possible to supply the service over the existing connection without significant modification provided that there is sufficient bandwidth capacity available.

PBX CONNECT TECHNICAL DATA

The PBX Connect SIP supports both G.729a, and G.711 A-law voice encoding with a sampling period of 20msecs. The encoding algorithm results in a per call bandwidth usage of between 50 – 100kbs.

Quality of Service (QoS) is implemented using ingress and egress traffic control at both ends of the LANnet connection. As part of the managed service, InTechnology provides a fully managed multi-service router for each customer site to shape traffic, ensuring that voice traffic is not negatively impacted by other traffic (such as Internet traffic) using the LANnet connection.

Each customer is provisioned with a private voice network over LANnet. This ensures security and business grade call quality between sites. LANnet is InTechnology's dedicated 10 GB national MPLS network and is specifically designed to carry high volumes of both voice and data traffic without causing degradation to the voice service.

If all available voice channels to a customer site are in use, InTechnology's IP Voice platform will prevent additional calls being setup, therefore, protecting the existing active calls being negatively impacted from call degradation. An incoming call which is unable to complete because the circuit is congested, or the subscribed maximum simultaneous call limit is reached will be treated as if the person they were calling was on the phone.

Suitable LANnet connections for PBX Connect SIP service are InTechnology managed ADSL 1000 and managed ADSL 2000, PPC and Ethernet connections.

Note: calls made between phones on the same site will not consume a channel because it will route internally through the PBX.

CONCURRENT SIP CHANNELS

The maximum number of concurrent channels = the available bandwidth/total bandwidth, so if for example a customer has a 1Mb SIP trunk using G.729a with a sample period of 20 ms there will be 1Mb of bandwidth required = 20 concurrent channels available.

Table 1 provides an estimate of the bandwidth requirements for VoIP calls using G.711, and G.729a, and the corresponding Mean Opinion Score (MOS). By comparison, GSM has a MOS of 4.1. The scores range from 1 (worst) to 5 (best).

Network	Codec	Sample period	Encoded sound bandwidth	IP/UDP/RTP overhead	Network overhead	Total bandwidth*	MOS†
Ethernet	G.711	20 ms	64 kbps	16 kbps	15.2 kbps	95.2 kbps	4.4
	G.729	20 ms	8 kbps	16 kbps	15.2 kbps	39.2 kbps	3.9
DSL	G.711	20 ms	64 kbps	16 kbps	26 kbps	106 kbps	4.4
	G.729	20 ms	8 kbps	16 kbps	18.4 kbps	42.4 kbps	3.9

Table 1: VoIP Bandwidth Calculations

Note: InTechnology's SIP trunk customers IP-PBX **MUST** support both G.711 and G.729a encoding. In the event calls are made into InTechnology's managed ADSL user base, calls will be delivered with G.729a as the preferred codec to ensure quality and performance.

PBX CONNECT SIP TRUNK SUPPORTS THE FOLLOWING RFCs

- RFC 3261 - SIP: Session Initiation Protocol
- RFC 2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals

Note: The IANA Registered Port 5060 should be utilized for all SIP Signaling Messages

NUMBERING & NUMBER PORTING

In order to facilitate inbound telephone calls, the service may be specified with geographic or non-geographic telephone numbers (DDIs). InTechnology can provide new number /DDI ranges to use with the service, as well as port existing numbers and number ranges from existing service providers.

Number porting forms a vital part of InTechnology's the service delivery process, InTechnology have current Geographic & Non-Geographic and porting agreements as listed below in tables 1 &2 that facilitate porting their number ranges onto our service, meaning the customer can retain their existing inbound numbers as part of the Unity or Unity PBX Connect service.

Non Geographic Number Portability (NGNP)
BT
Cable & Wireless (Ex-Energis)
Your Communications/Thus
Virgin Media (NTL & Telewest
Cable & Wireless
Magrathea
Telephony Services
Global Crossing
KComms/Affinity
Colt Telecom
Verizon

Table 2: Non-Geographic Porting Agreements

Geographic Number Portability (GNP)
BT
Virgin Media (NTL, Telewest & Eurobell)
Thus (incl. Your Comms)
Cable & Wireless (Ex-Energis)
Cable & Wireless UK (Inner&Outer London, Home Counties, Gloucester, Norwich, Nottingham, Hertford, Guilford, Ipswitch, Northampton, Wolverhampton, Brighton, Shrewsbury & Scotland)
Inclarity
Telephony Services
Viatel (UK) Ltd
Magrathea
Global Crossing

Geographic Number Portability (GNP)
Colt Telecom
Spitfire
KComms (does not include Kingston originating numbers in Hull)
Verizon
Voxbone
Storacall (X-On)
Opal Telecom (Subsequent, LCP import only)
Lumison (Subsequent LCP import only)
Primus (Subsequent LCP import only)

Table 3: Geographic Porting Agreements

InTechnology pre-sales consultants will gather the necessary information to complete the number port but will need the customer's assistance to identify the current Service Provider and Range Holder of the numbers, the site information including postcode and any associated direct dial inwards (DDI) numbers attached to main billing number. This information will then be included in the management summary of the contract so both parties have full visibility of the porting scope. In the majority of cases the presales consultants are able to obtain this data with just the main billing number and installation postcode from a customer's current invoice. InTechnology will in addition, request the customer provides on letter headed paper a standard letter of authority (LOA). The LOA together with the porting request form allows InTechnology's porting desk to talk directly to the losing operator's helpdesk on any number discrepancies.

Once established that a customer's numbers can be ported InTechnology will complete the relevant industry documentation and submit the porting request to the losing operator (and range holder where different) together with the letter of authority. Porting lead-times are a regulated and critical milestone in the overall delivery of the service and it's vital that the data received from the customer and presented to the losing operator and range holder is accurate, as this could result in the porting request being rejected. Once the porting order is accepted by the losing operator, the lead-time commences between the range holder and gaining service provider. To set a realistic expectation this can take up to 22 working days and in very rare cases longer. The InTechnology project manager, assigned to the delivery, will advise on the delivery date once confirmation is received back from the losing operator, the losing operator ultimately controls the date that the port will happen. Until the porting date confirmation is received back from the range holder and service provider any dates quoted will be on an indicative and estimated basis.

The installation price a customer is quoted includes any required number porting as long as the port is scheduled to take place during normal working hours (Monday-Friday/9-4), porting request outside of the stated hours will incur additional out of hours costs (bank and public holidays will be treated as an out of hours request).

Note: If customer provided numbering information results in a subsequent port rejection, InTechnology may charge the customer for porting resubmission.

NETWORK & INDIVIDUAL CALLING FEATURES

A range of optional network and calling features can be assigned to the service. Some of these features are chargeable.

Some of these features are assigned at group level, meaning they apply to all DDIs within the SIP trunk. Other services are assigned at a DDI level and apply only to that specific DDI.

Service	Service Description
Calling Line Identity Presentation (CLIP)	Enables the called party to receive and display the calling party's telephone number before answering the call. The called party will only receive this information if the caller has not restricted the sending of their number (CLI). The called party will require suitable CPE in order to use CLIP. **CLIP is assigned at an individual DDI level.**
Calling Line Identity Restriction (CLIR)	Customers can request that their identities (telephone numbers) are not revealed at any time. This service is available free of charge. **CLIR is assigned at an individual DDI level.**

Outgoing Calling Plan	Restricts the ability to make outbound calls.
	
	Options include blocking all outbound calls and selective blocking based on call type. Call types include:
	• Calls within the group
	• Local calls
	• National calls
	• Freephone calls
	• International calls
	• Directory enquiry calls
	• Premium rate calls including:
	  ○ 070
	  ○ 0871
	  ○ 090
	
	Outgoing Calling Plan is assigned at a group level and applies to all DDIs within that group.

Incoming Calling Plan	Restricts the ability to receive calls. Options include: • Allow external inbound calls • Partial – allow / block external inbound calls only if transferred by another user in the group • Block external calls **Incoming Calling Plan is assigned at a group level and applies to all DDIs within that group.**
Call Forwarding	Redirects inbound calls to an alternate destination. Call forwarding incorporates three sub-services:

Service	Service Description
	Call Forwarding Always – redirects all incoming calls to that DDI unconditionally to the specified alternate destination. Call Forwarding Busy – redirects incoming calls to the alternate destination if the line associated with the DDI is busy. Call Forwarding No Answer – redirects incoming calls to the alternate number if the call remains unanswered within a specified number of rings. Please note: The forwarded leg of the call will be charged to the customer at their pre-agreed tariff. Calls can only be forwarded to telephone numbers permitted by the outgoing calling plan **Call Forwarding is assigned at an individual DDI level.**
Anonymous Call Rejection (ACR)	Anonymous Call Rejection enables the rejection of calls from calling parties who have withheld their number. When activated, callers who have withheld their number are informed by an announcement that the person they are calling does not accept calls from anonymous callers. **Anonymous Call Rejection is assigned at an individual DDI level.**
Call Forwarding Advanced	Call Forwarding Advanced provides the capability to forward inbound calls intended for a particular DDI to another destination, when the incoming call matches pre-specified criteria. If the incoming call does not match any of the criteria, normal call handling applies. Definable criteria includes: • A time schedule – e.g. applies all day every day, or outside of business hours • Whether the calling party line ID is private or unavailable • A list of up to 12 phone numbers The criteria can be combined within predicates (for example, incoming call from this number and within business hours and during work week). Multiple predicates can be defined and the call is forwarded when at least one of the predicates is met. The user can associate a different destination with each predicate, or use the same destination for all predicates. **Call Forwarding Advanced is assigned at an individual DDI level.**

SIP TRUNK USER MOBILITY PACK

The SIP trunk user mobility pack is designed to allow a user to take advantage of the mobility features available on InTechnology's voice platform whilst still utilising their existing PBX.

Each DDI may be specified with this optional service pack that provides the user of that DDI with additional functionality delivered directly from InTechnology's voice platform. This pack provides the user with the 'Unity' PC Assistant Toolbar. This software application sits on the user's Windows PC desktop and integrates into Microsoft Outlook and Internet Explorer. The toolbar provides quick and easy control of the features contained within the Mobility pack including:

- Call Forwarding Always
- Call Forwarding Busy
- Call Forwarding No Answer
- CommPilot Express
- Outlook Integration
- Remote Office
- Simultaneous Ring Personal
- Sequential Ring

The PC Assistant Toolbar has a minimum supported level of hardware, software and network requirements. These are defined in the Technical Data section of this service definition.

WEB ADMINISTRATION PORTAL

The PBX Connect SIP service includes access to a secure web portal which may be used to administer certain features of the service. Access to this portal should only be provided to appointed personnel who are authorized to administer the organization's telephony services.

Examples of what the portal may be used for include configuring the network and calling services.

PRODUCT ORDERING INFORMATION

Part code	Description
N-PBX-111	PBX Connect SIP (per channel)
N-PBX-115	PBX Connect SIP Service Establishment (Per Customer)

INDIVIDUAL CALLING & NETWORK FEATURES

Part code	Description
N-PBX-120	Calling Line Identity Presentation (CLIP)
N-PBX-121	Calling Line Identity Restriction (CLIR)
N-PBX-122	Outgoing Calling Plan
N-PBX-123	Incoming Calling Plan
N-PBX-124	Call Forwarding
N-PBX-125	Call Forwarding Advanced
N-PBX-126	Anonymous Call Rejection

SIP TRUNK USER MOBILITY PACK

Part code	Description
N-PBX-130	SIP Trunk User Mobility Pack

SECTION 5: CHARGING POLICY

CHARGING STRUCTURE

SIP Trunk service charges are structured as follows:

- One time service setup charge
- Fixed recurring monthly charge calculated based on the number of contracted SIP channels
- Per minute charges for voice calls
- Fixed monthly recurring charges for optional services

Details of the actual prices for each element are detailed in the SIP Trunk service master price list.

INVOICING

SIP Trunk services will be invoiced on the 6th of every month.

BILLING

Billing is post-paid. Reconciliation is done daily, using a process to verify that all CDRs have been collected. A customer can access the InTechnology 'InForm' portal to review all unbilled data.

By default all calls will be billed to individual users. Alternatively, if required the service can be configured to bill to a "Charge To" number, which is equivalent to an ISDN Main Bill Number (MBN). The final billing setup and requirements will be agreed and tested during the setup of the service.

CALL DETAIL RECORDS

Call Detail Records (CDRS) will be produced for all calls, rated daily and made available for the customer to review and download via InTechnology's online 'Inform' portal.

REPORTING

The following billing reports are available for customers to view and download via the 'Inform' web portal.

- Itemised Monthly Service Charges
- Itemised Unbilled Calls
- Usage By Site/Sub-site/User
- Top Ten Spend Analysis

SECTION 6: SERVICE CAPABILITIES & LIMITATIONS

CAPACITIES

PBX Connect SIP service can be deployed with a minimum of 20 channels. Subsequent additional channels can be added in increments of 1

SERVICE LIMITATIONS

- The maximum number of simultaneous PSTN calls that can be made using the service is limited to the amount of available bandwidth on the customer WAN connection
- The service can only be delivered via managed InTechnology network circuits of adequate bandwidth
- The service cannot be delivered over ADSL Max, ADSL 512kbps, SDSL or any form of bonded xDSL circuit
- The service is delivered subject to successful completion of interoperability testing

FEATURE LIMITATIONS

The following ISDN type Network & Calling Features **may not** be supported. If any of the listed features are required, it must be tested and validated during the initial Interoperability phase.

- Connected Line Identity Presentation (COLP)
- Call Deflection
- Caller Redirect
- Customer controlled call forwarding of calls (using the PBX)
- Malicious call indication
- Data services are currently not supported via the service
- Signalling between PBXs is not supported using the service
- Fax is not supported; this includes analogue or ISDN fax machines connected via the PBX or directly to the gateway
- Analogue devices such as modems, bank machines and franking machines are not supported

LIMITATIONS WITH TRANSFERING & FORWARDING CALLS

Call Quality on Transferred and Forwarded Calls – PBX Connect SIP

The conversion from G.711 (PCM) to G.729a (also called transcoding) compresses the call, reducing the bandwidth required across the network to ~50kbps per call.

In certain call scenarios, calls can end up being transcoded from G.711 (PCM) to G.729a more than once. In this event, call quality will become degraded.

InTechnology cannot provide support for call quality issues on calls that have been transcoded more than once. Please see the "Supported and Unsupported Call Scenarios" section of this document.

Note: It is mandatory that G.729a codec must as a minimum be supported, if not the preferred codec.

Number Presentation on Forwarded Calls

The customers PBX must supply the number that diverted the call to the network, if the PBX fails to do this it must send the pilot number for all diverted calls.

This is not applicable to transferred calls.
Supported and Unsupported Call Scenarios

This section applies to:

PBX Connect SIP service where transcoding between G.729a and another codec occurs on the customer PBX (i.e. the handsets are not running G.729a)

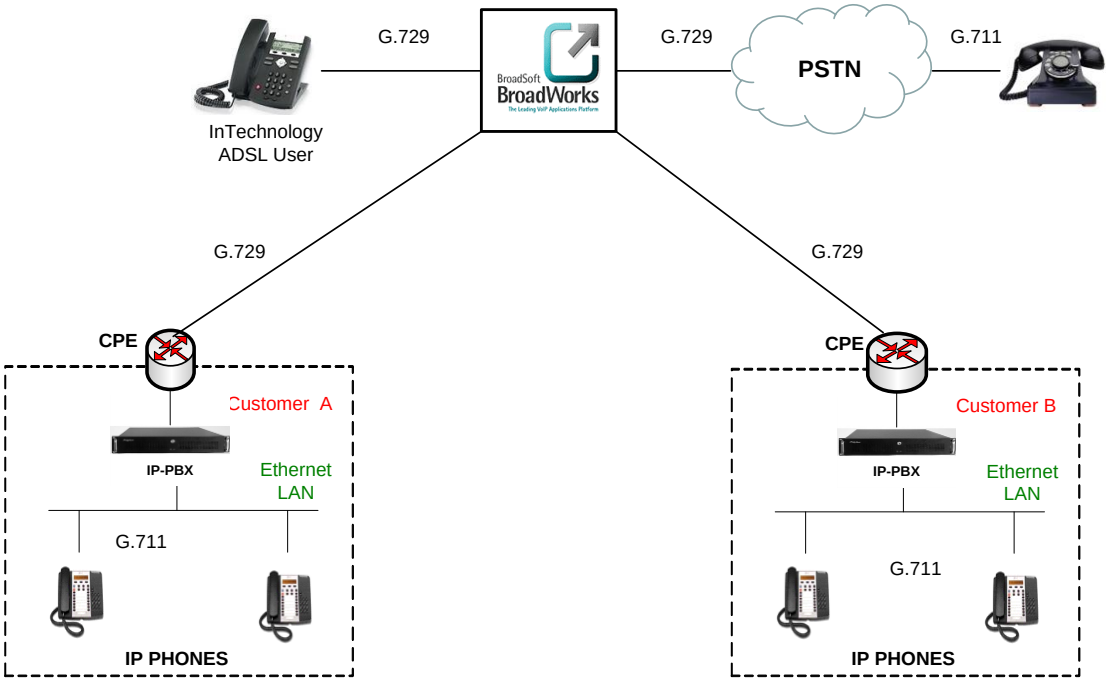

Figure 2: Network Call Scenario

InTechnology will not provide support for **call quality issues** for call scenarios that are not stated as supported in this section of the service definition.

SUPPORTED CALL SCENARIOS

- The PBX may make calls to the PSTN, to Unity IP Voice* users and to another PBX connected to the service
- The PBX may receive calls from the PSTN, from Unity IP Voice users and from another PBX connected to the service
- The PBX may receive calls from the PSTN that have been transferred or forwarded by a Unity IP Voice user
- Calls made by the PBX to Unity IP Voice users may be transferred or forwarded by the Unity IP Voice user to the PSTN, to other Unity IP Voice users or to another PBX connected to the service

*Unity IP Voice is InTechnology's hosted IP Telephony service

EXCEPTIONS

- Inbound calls may not be transferred or forwarded by the PBX, except to other internal telephone extensions connected to the same PBX
- Where more than one PBX is connected to the service, calls made between the PBXs may not then be transferred or forwarded

OVERVIEW OF SUPPORTED & UNSUPPORTED CALL SCENARIOS

Call Scenario	Description	Call Quality Supported
PBX to PSTN	A call is made from the PBX to the PSTN	Yes
PSTN to PBX	A call is made from the PSTN to the PBX	Yes
PBX to Unity	A call is made from the PBX to a Unity IP Voice user	Yes
Unity to PBX	A call is made by a Unity IP Voice user to the PBX	Yes
PSTN to PBX to PSTN	A call from the PSTN is transferred or forwarded by the PBX to another party on the PSTN	No
PSTN to PBX to Unity	A call from the PSTN is transferred or forwarded by the PBX to a Unity IP Voice user	No
PSTN to Unity to PBX	A call from the PSTN is transferred or forwarded by a Unity IP Voice user to the PBX	Yes
PBX to Unity to PSTN	A call from the PBX is transferred or forwarded by a Unity IP Voice user to another party on the PSTN	Yes
Unity to PBX to PSTN	A call from a Unity IP Voice user is transferred or forwarded by the PBX to another party on the PSTN	No
Unity to PBX to Unity	A call from a Unity IP Voice user is transferred or forwarded by the PBX to another Unity IP Voice user	No
PBX to PBX	A call is made from a PBX connected via the service to another PBX connected via the service	Yes
PSTN to PBX to PBX	A call from the PSTN is transferred or forwarded by the PBX to another PBX also connected via the service	No
PBX to PBX to PSTN	A call made from a PBX connected via the service, to another PBX connected via the service, is transferred by the 2nd PBX to another party on the PSTN	No
PBX to PBX to Unity	A call made from a PBX connected via the service, to another PBX connected via the service, is transferred by the 2nd PBX to a Unity IP Voice user	No
Unity to PBX to PBX	A call made from a Unity IP Voice user to a PBX connected via the service is transferred or forwarded to another PBX connected via the service	No
PBX to Unity to PBX	A call made from a PBX connected via the service is transferred or forwarded by a Unity IP Voice user to another PBX connected via the service	Yes
PSTN to PBX to PBX to PSTN	A call made from the PSTN is transferred or forwarded by a PBX connected via the service to another PBX connected to the services. This call is then transferred or forwarded by the 2nd PBX to another party on the PSTN	No

PC ASSISTANT TOOLBAR TECHNICAL DATA

Requirement	Description
Internet connectivity requirements	**All Platforms** ▪ Access to **ipt.intechnology.co.uk** on TCP / UDP port 80, 2208, 443 ▪ Access to **portal.ipt.intechnology.co.uk** on TCP / UDP port 80, 2208, 443 ▪ Access to **bsews1.ipt.intechnology.co.uk** on TCP / UDP port 80, 2208, 443 ▪ Access to **bsews2.ipt.intechnology.co.uk** on TCP / UDP port 80, 2208, 443 ▪ Access to **ecom.intechnology.co.uk** on TCP port 80
Software requirements	▪ Windows 2000 with SP4 (or higher), Windows XP, Windows Vista, or Citrix Presentation Server 3 or 4 ▪ Windows Installer 2.0 ▪ Internet Explorer 6.0, 7.0, or 8.0 (required for IE toolbar edition) ▪ Mozilla Firefox 2.0 or 3.0 (required for Firefox toolbar edition) ▪ Outlook 2000 SP3, 2002/XP SP2, 2003, 2007 (required for Outlook toolbar edition) **Citrix Presentation Server Platform** The application can be published on a Citrix server via the Management Console for MetaFrame. **Citrix ICA Client Workstation Platform** No additional software requirements.
Hardware requirements	**Microsoft Windows Platform** ▪ 2 GHz Intel Pentium 4 or equivalent CPU ▪ 512MB of RAM ▪ 60MB of free hard disk space ▪ 8MB Video Graphics Card ▪ SVGA monitor (15” or larger) ▪ 800 x 600 minimum screen resolution
	Citrix Presentation Server Platform ▪ 2 GHz Intel Pentium 4 or equivalent CPU ▪ Minimum 2GB of RAM ▪ 60MB of free hard disk space
	Citrix ICA Client Workstation The hardware requirements for a Citrix ICA client workstation include: ▪ 1.2 GHz or higher, Pentium 3, or compatible CPU ▪ 128 megabytes (MB) of RAM ▪ Video graphics card with 8 MB of RAM minimum ▪ 800 x 600 screen resolution minimum ▪ Network connection of minimum 56 Kbps speed

SECTION 7: RESPONSIBILITIES

Customer Responsibilities
PBX telephone system; availability, operation, configuration and maintenance.
PBX telephone handsets; operation, configuration and maintenance.
Ongoing moves /adds / changes associated with the PBX.
All fixed cabling including data and voice; suitability for use with PBX, operation, maintenance.
All LAN configuration, operation and maintenance.
Any re-configuration or upgrades (hardware or software) to the PBX that may be required in order for the PBX to operate with the service.
Provision and ongoing supply of mains power to any InTechnology hardware required in order to provide the service.
Implementation of suitable security policies to prevent fraudulent use of the PBX and the service.

InTechnology Responsibilities
Service provision and availability.
Provision of analogue telephone lines to service analogue devices where requested by the customer

InTechnology Managed Services
Central House
Beckwith Knowle
Harrogate
HG3 1UG

Tel: +44 (0)1423 850 000
Fax: +44 (0)1423 850 001
contact@intechnology.com
www.intechnology.com

inTechnology
MANAGED SERVICES

Certificate No. FS 578568

Certificate No. IS 578569