REDCENTRIC

SIP TRUNKING

SERVICE DEFINITION

SD032 Version 2.1
Issue Date 21 June 2018

redcentric
business technology. managed.

1. SERVICE OVERVIEW

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

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. Redcentric's PBX Connect SIP enables organisations to run both voice and data services via a single converged network connection, maximising investments made in their existing telephony equipment.

1.1 KEY FEATURES & BENEFITS

- The ability to make and receive calls to and from the PSTN (Public Switched Telephone Network) via Redcentric's managed IP network and Tier-1 interconnect partners.
- Flexibility – PBX Connect SIP can be increased and 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 – Redcentric'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 on-going call costs.

2. SERVICE DESCRIPTION

2.1 FUNCTIONAL OVERVIEW

Figure 1 below details the PBX Connect SIP Service 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 (E-SBC).

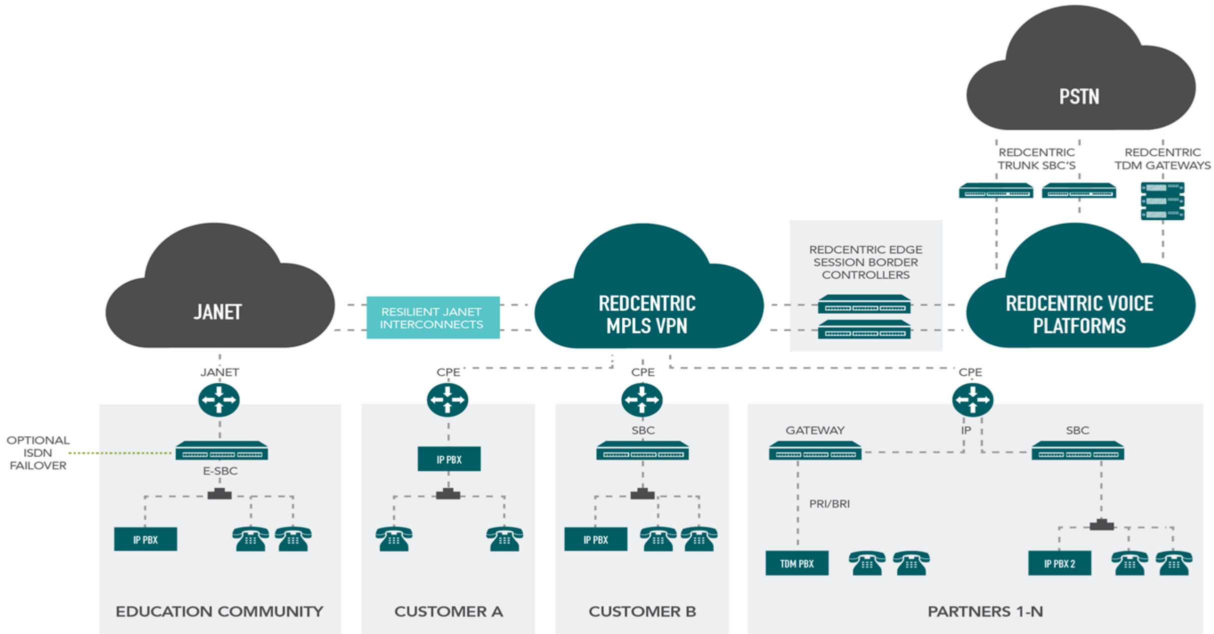

Figure 1: PBX Connect SIP Trunk Service Architecture

2.2 REDCENTRIC VOICE SWITCHES

Redcentric's PBX Connect SIP Service is delivered from two centrally managed, software based, IP Voice Switches hosted in Redcentric'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.

Redcentric's IP Voice platform has been deployed in a highly available configuration across Redcentric Data Centre locations in Harrogate and London. The core IP Voice Switches are maintained and supported by Redcentric 24 / 7 and access to the service is delivered across Redcentric'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.

2.3 SESSION BORDER CONTROLLER (SBC)

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

2.4 NETWORK CONNECTIVITY

PBX Connect SIP Service is delivered over Redcentric's IP MPLS network "LANnet". In order to assure call quality and performance across the network, Redcentric 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.

2.5 NETWORK FAILOVER

SIP trunk failover can only be implemented as standard with Redcentric 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.

2.6 PSTN CONNECTIVITY

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

2.7 EMERGENCY CALLS

Calls to the emergency services can be made using the PBX Connect SIP Service, subject to the service being available.

Site location information is held by Redcentric, which defines the physical location address associated with each telephone number. This address information may be provided to the emergency services if they are contacted to direct them to the source of the call. As such the accuracy of this information is critical. The Customer is responsible for notifying Redcentric of any change address where voice services are provisioned. Moving services to a new location without providing updated site location information is in breach of the General Condition 4 of the General conditions of Entitlement enforced by OFCOM. As the SIP Trunking over JANET Service includes functionality to facilitate mobile and remote working, all telephone numbers are registered with the emergency services as 'nomadic'. This means that during an emergency services call, the geographical location of the caller will be confirmed.

2.8 OFFICE CONNECTIVITY

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

2.9 PBX CONNECT TECHNICAL DATA

The PBX Connect SIP Service 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, Redcentric 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 Redcentric'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, Redcentric'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 Redcentric 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 Customer PBX.

2.10 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

Table 1: VoIP Bandwidth Calculations

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

2.11 RFC SUPPORT

The SIP Trunk service supports:

- 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 Signalling Messages

2.12 NETWORK AND 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 PBX Connect SIP Service. 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:

Service	Service Description
	○ 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: 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

Service	Service Description
	incoming call does not match any of the criteria, normal call handling applies. Definable criteria include: • 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.**

2.13 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 Redcentric'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 Redcentric'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.

2.14 PC ASSISTANT TOOLBAR TECHNICAL DATA

Requirement	Description
Internet connectivity requirements	**All Platforms** • Access to **ipt.Redcentric.co.uk** on TCP / UDP port 80, 2208, 443 • Access to **portal.ipt.Redcentric.co.uk** on TCP / UDP port 80, 2208, 443 • Access to **bsews1.ipt.Redcentric.co.uk** on TCP / UDP port 80, 2208, 443 • Access to **bsews2.ipt.Redcentric.co.uk** on TCP / UDP port 80, 2208, 443 • Access to **ecom.Redcentric.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

Requirement	Description
	• 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

2.15 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 forincludes: configuring the network and calling services.

2.16 NUMBERING & PORTING

To facilitate inbound telephone calls, the service may be specified with geographic or non-geographic telephone numbers (DDIs). Redcentric can provide new number /DDI ranges to use with the service, as well as porting a Customer's existing number, or number ranges from existing Service Providers.

Number porting forms a vital part of Redcentric's service delivery process, Redcentric have current Geographic & Non-Geographic and porting agreements as listed below in tables 2 & 3 that facilitate porting number ranges onto the PBX Connect SIP Service, meaning that Customers can retain their existing number(s)as part of the PBX Connect SIP 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
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

Redcentric 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 most 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. Redcentric 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 Redcentric'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 Redcentric will complete the relevant industry documentation and submit the porting request to the losing operator (and range holder where different) together with the LOA. 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 Redcentric 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, Redcentric may charge the Customer for porting resubmission.

2.17 SERVICE ORDERING

New PBX Connect SIP Service is ordered by way of Redcentric standard contract process. Subsequently, orders can be placed by authorised person completing the online information via Redcentric's "Inform" portal.

Orders can be placed using the following part code(s):

2.18 CORE PRODUCTS

Part code	Description
N-PBX-111	PBX Connect SIP (per channel)
N-PBX-115	PBX Connect SIP Service Establishment (Per Customer)
N-PBX-116	Connect SIP Service Establishment (OFFNET)

2.19 OPTIONAL SERVICES

The optional services listed in the table below can be added to the PBX Connect SIP Service *at any point during, or after the initial service has been configured and/or deployed.*

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

2.20 SIP TRUNKING USER MOBILITY PACK

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

2.21 ON-BOARDING PROCESS

Redcentric's on-boarding comprises of a project managed, five phase technical and business consultative process.

Phase 1 - Redcentric presales to work with the Customer to clearly capture & document the current requirement(s).

Phase 2 – Perform PBX Connect SIP Service interoperability test to ensure compatibility with SIP trunk service. Customer sign-off required prior to next phase.

Phase 3 - If required, porting requests will be submitted to the Losing Communications Providers (LCPs).

Phase 4 - On the day of the port, Redcentric will create the Customer's account and assign either ported, or new numbers into the Customer's account.

Phase 5 - Optional account based services added to the PBX Connect SIP Service.

2.22 DECOMMISSIONING PROCESS

Upon expiration of the PBX Connect SIP Service contract where the Customer chooses not to renew with Redcentric, the following steps are followed as part of the decommissioning process:

Phase 1 - Contractual

Expiration of the service contract or the Customer decides not to renew. This may also include early termination by the Customer, subject to payment of early termination fees.

Phase 2 - Service Decommission

Cessation of the PBX Connect SIP Service and any associated optional features. Plus, the removal of user accounts from the core platform.

Phase 3 - Number Porting

Porting out of any telephone numbers out from Redcentric to the gaining party where this party has an existing porting agreement with Redcentric

3. IMPLEMENTATION AND ACCEPTANCE

3.1 ACCEPTANCE CRITERIA

The following acceptance criteria will be demonstrated during the service delivery process:

- SIP Trunk Interoperability test plan Customer sign-off
- Email notification that numbers porting in have successfully been ported into Redcentric network.
- Where Customers are porting numbers into Redcentric, a number of test calls will be made to pre-defined DDIs to confirm numbers are live on the Redcentric network prior to final Customer handover to live service.

3.2 CUSTOMER DEPENDENCIES

It's the Customer's responsibility to:

- Provided all required information to allow Redcentric to submit appropriate porting requests where appropriate.
- Provide signed LOA (Letter of Authority) to facilitate number porting requests where required.
- Maintenance of on-site telephony equipment (unless equipment is provided and covered by Redcentric maintenance agreement).
- Provision of site access for engineer where required
- Provision of mains power to line installation point where required
- The accurate reporting of any service related faults
- PBX telephone system; availability, operation, configuration and maintenance.
- PBX telephone handsets; operation, configuration and maintenance.
- On-going moves /adds / changes associated with the PBX (unless equipment is provided and covered by Redcentric maintenance agreement).
- All fixed cabling including data and voice; suitability for use with PBX, operation, maintenance.
- All LAN configuration, operation and maintenance (unless equipment is provided and covered by Redcentric maintenance agreement).
- 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.
- Implementation of suitable security policies to prevent fraudulent use of the PBX and the service.

3.3 SERVICE CAPABILITIES & LIMITATIONS

Capacities

PBX Connect SIP Service can be deployed with a minimum of 10 channels. Subsequent additional channels can be added in increments of 1 channel.

3.4 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 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

3.5 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

3.6 LIMITATIONS WITH TRANSFERRING & 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.

Redcentric 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.

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.

3.7 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)

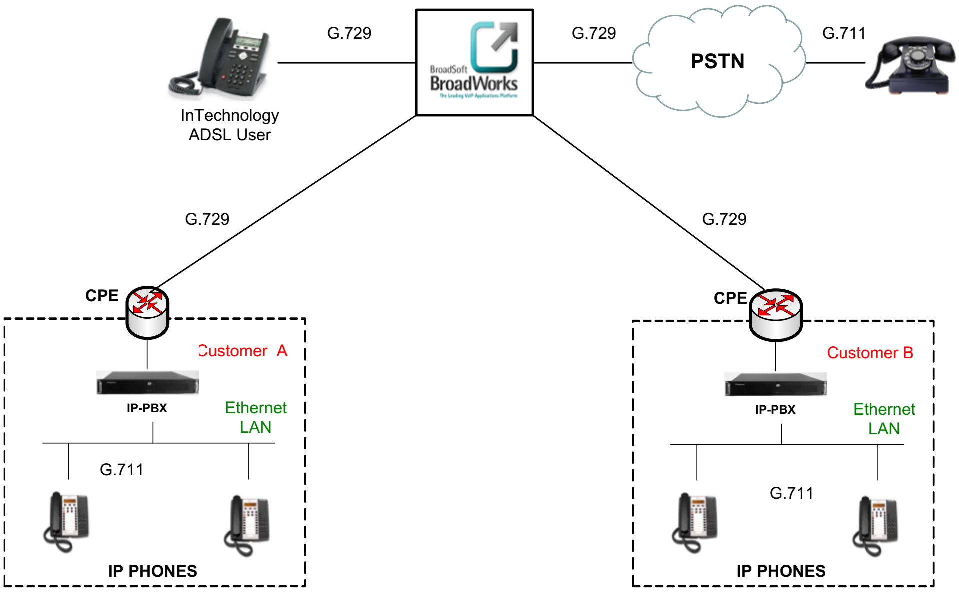

Figure 2: Network Call Scenario

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

3.8 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 Redcentric's hosted IP Telephony service

3.9 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

3.10 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

Call Scenario	Description	Call Quality Supported
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

4. SERVICE LEVELS AND SERVICE CREDITS

4.1 SERVICE LEVELS

The Service Levels applicable to the PBX Connect SIP Service is as follows:

Service Level: Availability Measurement Period: Month	
Service Level	Not less than 99.99%

4.2 EXCLUSIONS FROM AVAILABILITY

In calculating Availability, in addition to the exclusions listed in clause 6.7 of the General Terms the following shall be excluded:

Call completion not possible due to busy signal – fully utilised Trunk line or network capacity for example.

4.3 FLOOR SERVICE LEVEL

The Floor Service Level applicable to the PBX Connect SIP Service in respect of Availability shall be 85% in any given Month.

4.4 SERVICE CREDITS

The Service Credits applicable to the PBX Connect SIP Service shall be calculated as follows:

$$\textit{Service Credit} = \frac{C \times S}{MS}$$

Where:

S = the number of seconds by which Redcentric fails to meet the Service Level for Availability in the relevant Month (subject to the provisions of paragraph 4.2 above)

C = total Charges payable in respect of the SIP Trunking Service *for the same Month*

MS = total number of seconds in the same month

In the following table:

"≥" means "greater than or equal to"

< means "less than"

"MS" means the total Charges payable in respect of the PBX Connect SIP Service for the same Month

Applicable SIP Trunking Service	Service Availability	Service Credit
	99.99%	none
	≥<99.95% but <99.98%	10% of MS
	≥<99.75% but <99.94%	15% of MS
	<99.75%	25% of MS

5. DATA PROCESSING

5.1 DATA PROCESSING SCOPE

- Redcentric does not access, alter or use any application data that is running on the SIP Trunking Service except as specifically stated below.
- In terms of operating the SIP Trunking Service, API commands are passed into the SIP Trunking associated supporting servers to orchestrate the build/management of identified users that have subscribed to the Service.
- Users that have the appropriate role/privileges assigned to them can access the Service via a secure web portal to manage their individual SIP trunk configuration.
- The agreed roles and responsibilities are provisioned based on documented Customer requirements.

5.2 DATA STORAGE AND UNENCRYPTED DATA

- All SIP Trunking service associated data is stored within Redcentric's privately owned and managed data centre facilities.
- All access to data within the SIP Trunking service is via secure portal.
- All access to data is restricted to Customer identified users.

5.3 DATA PROCESSING DECISIONS

- In the normal course of business Redcentric does not make any data processing decisions in relation to the SIP Trunking Service. Processing is automated and instigated by the Customer.
- Redcentric Support can be asked by the Customer to intervene in the event of an issue with the Service. In such a case Redcentric may make decisions that can affect data processing, but such actions will only be undertaken at the request of and in conjunction with the Customer.

5.4 SERVICE CONFIGURATION WITH RESPECT TO DATA

- The initial Service configuration is built using a combination of Redcentric and Customer provided information.
- As data controller, Redcentric holds the following information on Users on Redcentric's BroadWorks telephony service delivery platform:

    - **Company Information**: Company Name, Associated Site Address, Postcode
    - **SIP Trunk User Data**: Telephone Number on Users:

5.5 DATA BACKUP

- Redcentric performs daily database data backups.
- Backup data is encrypted during transit.
- Backup data is encrypted whilst in storage.

5.6 SUB-PROCESSORS

- No other parties are involved in delivering the SIP Trunking Service, and there are no sub-processors appointed by Redcentric.

5.7 CUSTOMER ACCESS TO DATA

- The Customer has login rights to the SIP Trunking Service via secure web portal.
- Access to the SIP Trunking Service is based on roles and responsibilities defined by the Customer as part of the service setup.
- Redcentric can access reporting data, but would only do this after a formal support request by the Customer/authorised user.

5.8 SECURITY ARRANGEMENTS AND OPTIONS

- Customers have access via a secure portal to manage their own SIP Trunk user configuration, but they are unable to interact directly with the back-end systems to modify any service wide configurations.
- Customer access to the portal uses role-based access controls (RBAC), integrated with Redcentric core voice platform
- All locations meet physical security standard ISO27002 section 11.1 or equivalent.

HARROGATE (HEAD OFFICE)
Central House
Beckwith Knowle
Harrogate
HG3 1UG

LONDON
Lifeline House
80 Clifton Street
London
EC2A 4HB

THEALE
2 Commerce Park
Brunel Road
Theale
Reading
RG7 4AB

HYDE
Unit B
SK14 Industrial Park
Broadway
Hyde
SK14 4QF

CAMBRIDGE
Newton House
Cambridge Business Park
Cowley Road
Cambridge
CB4 0WZ

INDIA
606-611, 6th Floor
Manjeera Trinity Corporate
JNTU – Hitech City Road
Kukatpally, Hyderabad – 72

READING
3-5 Worton Drive
Reading
RG2 0TG

0800 983 2522
sayhello@redcentricplc.com
www.redcentricplc.com

redcentric
business technology. managed.

bsi. ISO 9001 Quality Management FS 603185
ISO 22301 Business Continuity Management BCMS 603194
ISO/IEC 27001 Information Security Management IS 603187
ISO/IEC 20000-1 Information Technology Service Management ITMS 668453