Service Description

SIP Trunking

Version 1.5

Introduction

This document is the copyright of Claranet intended for internal use and Customer distribution only. This description is subject to the Claranet Business Master Services Agreement.

Service in this document is defined as any combination of the various services outlined below and ordered by the Customer ("**Service**").

The purpose of this document is to provide a reference guide to the processes and procedures Claranet employs when delivering the Service(s) and to define the Service level metrics that Claranet will endeavour to deliver against when providing SIP Trunking.

This Service Description details Claranet's and the Customer's key duties, obligations and responsibilities related to the provision of the Services described herein. This document forms part of the Agreement between the Parties.

Contents

Definitions

Term	Description
SIP	SIP stands for Session Initiation Protocol and is the means by which a real-time data stream is established between user's endpoints and the central platform for the purpose of voice, video or desktop sharing.
IP-PBX	An IP-based voice switch capable of handling incoming and outgoing calls for an organisation.
Voice Gateway	A SIP-aware device, used to convert a traditional TDM-based PBX into one that can use a SIP Trunk service.
ESBC	An Enterprise Session Border Controller, which is a smaller version of the network-based IP voice 'firewalls', designed to protect Customer networks from intrusion via VoIP services.
IP Phone	A physical telephone device that can connect to its controlling voice switch over Internet Protocol.
On-Net	Calls made between users within the same Customer.
Off-Net	Calls made to destinations outside of the Customer organisation.
ACD	Automatic Call Distribution. A means by which an incoming call can be picked up by one of a team of agents, leaving the remaining agents to take other calls.
VoIP	Voice Over Internet Protocol is a general phrase for the capability to make and take voice calls over a data connection.
LCP	Losing Communication Provider. The provider of a phone number which is moving to a new provider as part of a number porting activity.
GCP	Gaining Communication Provider. The new provider of a phone number that has moved from an LCP.
End User	The individual who is the ultimate consumer of the Service. I.e. the person using the Service to make and receive phone calls and other communications via the Service.
BT Retail	A division of British Telecom that sells a variety of communication services directly to businesses.
BT Wholesale	A division of British Telecom that sells a variety of communication services to Service Providers such as Claranet for onward sale to businesses.
Emergency Services	The UK emergency services organisations i.e. Police, Fire, Ambulance. Typically reached by dialling 999 from a telephone.
SIP Trunk	A group of SIP Channels which are delivered to a single identified end-point.

Term	Description
SIP Channel	A VoIP 'channel' capable of carrying a single voice call. Multiple channels are required in order to have more than one call happening at a time.

1. Executive summary

SIP Trunking from Claranet provides a highly reliable IP-based voice connection, suitable for connection to a wide variety of Customer-premise based IP-PBX and Voice Gateway platforms.

This service is designed to replace traditional ISDN multi-channel voice services and to be provided over a multi-purpose IP connection (i.e. not a dedicated voice connection).

The SIP Trunking platform is hosted within our voice partner's network and Claranet Customer's IP-PBXs can connect via Claranet MPLS connectivity or via a suitable internet connection. The platform provides break-in and break-out for UK calls and a number of routing options in both directions. The Customer's IP-PBX will continue to provide all the call control and user features.

SIP Trunking is a fully managed service in the sense that Claranet is required to manage all changes to the Customer's SIP Trunk service. Claranet will not manage or maintain any IP-PBX or gateway equipment. Claranet will help the Customer match their concurrent call volume needs to the capabilities of the Service and propose a trunk and channel structure that meets those requirements. Claranet will deploy the service with the Customer to an agreed initial configuration and will manage any changes to the service structure that the Customer requires subject to assessment for Professional Services charges for that work to be carried out.

Claranet's voice partner in this service is British Telecom (BT) who provide and manage the core call handling platform and other aspects of the Service including number management and call break-in/out. Some of the terms of our agreements with BT for this service are necessarily passed through to our Customers and are detailed below.

Claranet SIP Trunking is constructed from the following components:

Product Element	Description	Charging Model
Call Usage	Charges for outbound off-net calls.	Pence Per Minute, Charged Monthly.
Geographic numbers	Telephone numbers (01/02) which can be assigned to the Customer's SIP Trunks.	One-off costs for new or ported numbers.
SIP Trunk Groups	A logical group of SIP Channels designed to connect to an IP-PBX or Gateway.	Per Trunk Per Month Costs.
SIP Channels	A SIP Channel capable of carrying a single call. Multiple channels will be required to carry multiple concurrent calls.	Per Channel Per Month.
SIP Routing Configurations	Routing rules for inbound and outbound calls to and from the Customer's IP-PBX.	No charge.

SIP Trunks are deployed to the locations of a Customer's IP-PBX whether that is in a datacentre or on a Customer's physical sites. The product can provide multiple trunks to a single Customer to multiple locations or to a single location as required.

Please note that this service can only connect to IP-PBX/Gateways that have been approved by BT – please check carefully that the Customer's IP-PBX/Gateway is on the Approved Equipment List on the Online Service Catalogue.

2. SIP Trunking Service Components

2.1. Core Voice Platform

The core SIP Trunking platform has a number of configuration options, and can be configured for specific Customer requirements.

SIP Trunk Functionality
Native SIP protocol support – allows a Customer to remove/eliminate the need for on-site VOIP2TDM gateways. (approved equipment list only)
Number Porting – carry existing (geographic) phone numbers across to Wholesale SIP Trunking
PSTN break-in – receive inbound calls from the Public Switched Telephone Network (national, mobile, international)
PSTN break-out - make outbound calls to destinations normally reachable over the Public Switched Telephone Network (national, mobile and international) some restrictions may be applied due to existing fraud issues.
Transport of the Calling Line Identity of the caller onto the IP PBX
Presentation Number screening
Trunk level Call Admission Control – specify the maximum number of simultaneous calls a particular trunk (site) can handle
Trunk Group level Call Admission Control – specify the overall maximum number of simultaneous calls the platform needs to cater for your estate
Channel aggregation across sites
Round Robin Trunk call distribution – to load share telephony traffic across multiple sites
Priority Based Trunk call distribution – to specify main sites from back-up/overflow sites
Trunk level Call Barring features – to prevent certain calls from taking place
Call Diversion features – point incoming calls to different destinations depending on the conditions of the Trunk
Emergency Call Handling – ensure that 999 calls are treated correctly and the appropriate address information is displayed to the emergency operator per number
Directory Listing – add or keep your numbers in the telephone book

2.2. Restrictions & Other Terms

The following general restrictions apply:

- Claranet will not install, manage or maintain any on-premise IP-PBX or gateway devices.
- Claranet will not install, manage or maintain any on-premise LAN cabling between our router and the Customer IP-PBX or gateway.
- All equipment connected must be on the approved equipment list.
- Maximum number of Linked sites = 6
- Up to 50 calls per second can be supported
- New numbers can be ordered in blocks of up to 100 numbers or multiples of. (i.e. 500 = 5 blocks of 100)
- Maximum number of channels per trunk group is 5000
- There is no interworking with other VOIP services
- Video is not supported
- Franking machines are also not supported.
- Dial up internet calls are not supported.
- Short message service and text messaging calls are not supported.

The Customer take all reasonable steps to ensure that the Service is not used:

- To make Nuisance Calls;
- To send, knowingly receive, upload, download, use or re-use material which is offensive, indecent, defamatory, obscene or menacing;
- In a way that does not comply with the terms of any legislation or any licence applicable to the Customer or End User;
- In a manner that is in any way unlawful, fraudulent or in bad faith or, to the knowledge of the Customer, has any unlawful, fraudulent or bad faith purpose or effect; or
- In a manner that in BT Wholesale's reasonable opinion could materially affect the quality of any telecommunications service, including the Service, provided by BT Wholesale.

2.2.1. Redcare

Redcare is a security service provided by BT Retail that involves the monitoring by BT Retail of a phone line installed a site and connected to an approved alarm system.

It is not possible to migrate Redcare services to a SIP Trunk (or any other IP-based voice service) and the PSTN line used for that service should be retained by the Customer.

2.2.2. Analogue Devices and Fax Machines

Analogue devices can be connected to the Customer's IP-PBX and use the SIP trunk for PSTN connectivity.

Examples of this may include:

1. Fax machines.
2. PDQ Machines for Credit/Debit Card transactions.
3. Door entry systems.

4. Hardened devices such as warehouse phones or outdoor phones.

In these cases the Customer will need to incorporate an ATA device which converts analogue signals into IP. We will not supply the analogue devices themselves.

Please note that Fax over IP is not wholly reliable. In fax transmission, timing is critical and the conversion to and from IP can introduce a tiny but significant delay in the transmission, and can cause an inbound and outbound fax to fail.

If fax is critical to the Customer business it is often better to retain an individual PSTN phone line, dedicated to the fax machine.

2.2.3. Automatic Diallers

Automatic Diallers are devices that create a high volume of outbound call attempts, usually as part of an outbound contact centre which uses large lists of phone numbers for the purposes of sales to consumers.

The SIP Trunking service is suitable for Automatic Diallers, but requires a high Call Attempts Per Second (CAPS) ratio to be used, and will require Premium-level channels.

Customers who are intending to use an Automatic Dialler should work with a Claranet Solution Architect to ensure that their SIP Trunking service will cope with the high volume of attempted calls.

2.3. SIP Trunk Service Hierarchy

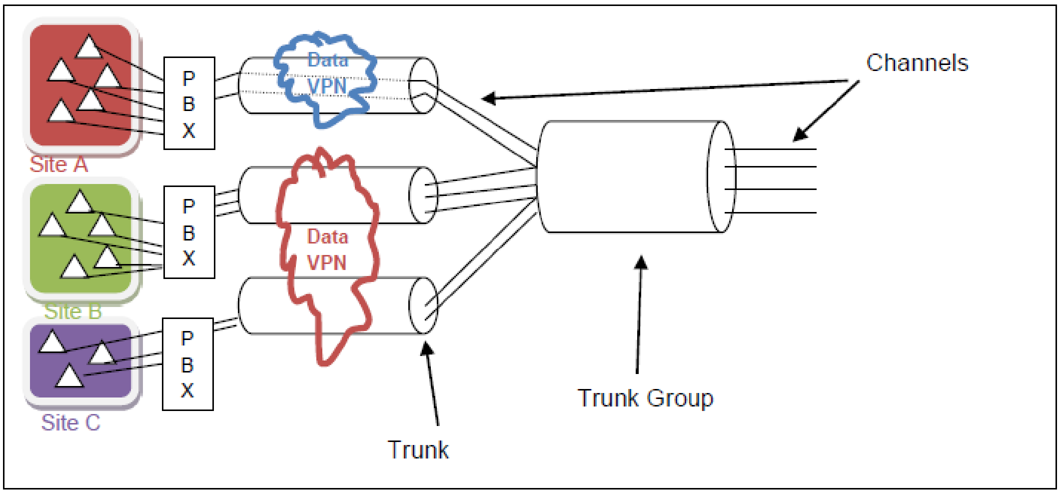
To enable flexibility, SIP Trunking uses a hierarchy of Channels, Trunks and Trunk Groups.

A Channel is the capacity needed to carry a telephone call and is an attribute of both a Trunk and a Trunk Group.

A Trunk is a logical connection between an IP PBX and the SIP Trunking platform, and will have a designated Channel capacity.

This capacity is governed by the trunk's CAC (Call Admission Control).

One or more Trunks can be grouped into a Trunk Group, which in turn also have their designated (aggregated) Channel Capacity, hence forming a hierarchical tree.



Multiple data VPNs can be used in the same Customer to provide the physical transport layer, then Trunks are created to logically define the connections between sites, and the whole sit within a Trunk Group. Please note trunks can only belong to one trunk group.

2.4. Differentiated Channel Types

In contrast to ISDN30 where it is a one-size-fits-all, with SIP Trunking there are three types of channels that can be assigned to a Trunk Group. These have been defined for specific traffic use. Please note each Trunk Group has to have the same type of channel – it is not possible to mix and match channel types within the same Trunk Group.

- Basic – this type of SIP trunk is suited for occasional use, typically smaller satellite offices or remote / home locations that don't produce a constant flow of telephony traffic.
- Standard – this SIP trunk provides the trunk throughput for normal business use, with peaks & pits of utilisation throughout the day. This is engineered in the backbone of the SIP trunking platform to offer a match to the present ISDN30 service
- Premium – these are the heavy-duty SIP trunks that are engineered inside the platform to support a near to constant use, day-in day-out, a service level superior to what is presently available with ISDN30 trunk telephone lines. These channels are capable of supporting diallers and high-volume outbound call centres.

2.5. Permitted Call Types

The SIP Trunking service will support the call types in the list below with a tick next to them, where * represents any number.

Calls preceded by 141 and 1470 will either release or withhold the presented CLI as normal.

Dialled Number	Digits	Allow	Emergency	Translate To	Comments
00xxxxx*	7-17	☑		+xxxxx*	International (00 + up to 15 digits).
01xxxxxxxx[x]	10-11	☑		+441xxxxxxxx[x]	National, Geographic
02xxxxxxxxx	11	☑		+442xxxxxxxxx	National, Geographic
03xxxxxxxxx	11	☑		+443xxxxxxxxx	Services charged at national rate.
04*		☒			Unallocated
05xxxxxxxxx	11	☑		+445xxxxxxxxx	National, Corporate and Location Independent ECS
0500xxxxxx	10	☑		+44500xxxxxx	Freephone
06*		☒			
07xxxxxxxxx	11	☑		+447xxxxxxxxx	National, Mobility
08xxxxxx[xxx]	8-11	☑		+448xxxxxx[xxx]	National, Special
09xxxxxxxxx	11	☑		+449xxxxxxxxx	National, Premium

Dialled Number	Digits	Allow	Emergency	Translate To	Comments
1*		☒			Do not route except as noted below.
100	3	☑		10070124	Assistance Operator
101	3	☑		1010678001001001	National Single Non-Emergency Number Rollout expected during 2011 and 2012.
111	3	☑		111	Access to NHS Non-Emergency Health Services Not nationally available until 2013. NICS translates to 111067900100111
112	3	☑	☑	99970124	EU Emergency Number , routed as 999
116xxx	6	☑		116xxx	EU Harmonised services, defined services follow.
116000	6	☑		116000	Hotline for Missing Children
116006	6	☑		116006	Helpline for victims of crime Not currently routed by NGS
116111	6	☑		116111	National Society for the prevention of Cruelty to Children.
116117	6	☑		116117	Nonemergency medical on-call service Not currently routed by NGS
116123	6	☑		116123	Samaritans, Emotional Support Helpline
118xxx	6	☑		118xxx	Directory Enquiries
123	3	☑		123	Speaking Clock
12*		☒			Indirect Access Codes
13*		☒			Indirect Access Codes
14*		☒			Indirect Access Codes
144		☑		+44800144144	BT Chargecard
1471		☒			Call Return
1475		☒			Call Return CLI Erasure
1477		☒			Malicious Call Trace
150		☑		+44800800150	PC Sales and Billing

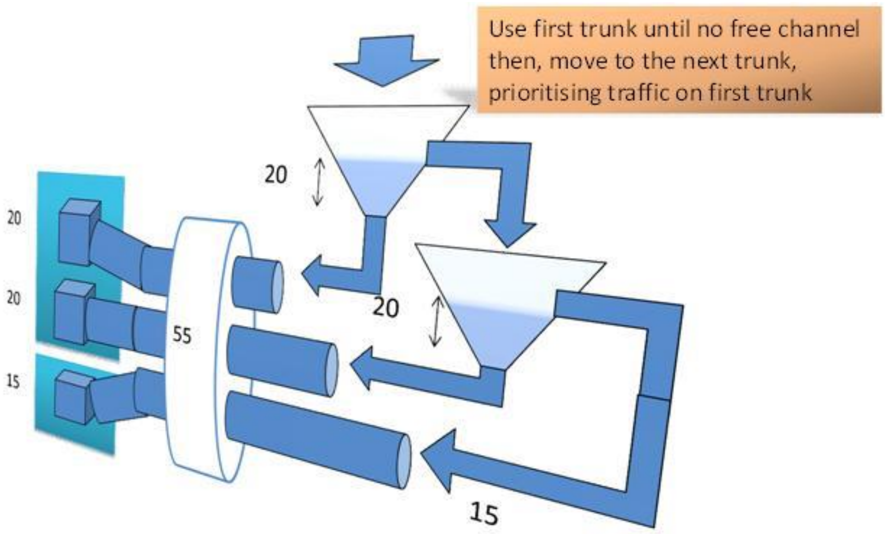
Dialled Number	Digits	Allow	Emergency	Translate To	Comments
151		☑		+44800800151	PC Fault Reporting
152		☑		+44800800152	BC Sales and Billing
153		☒			Message for International DQ
154		☑		+44800800154	BC Fault Reporting
155		☑		15570017	International Assistance Operator
1571		☒			Voicemail
16*		☒			Indirect Access Codes
17*		☒			Test Numbers
17070		☑		+442087599036	Line Test
18*		☒			Indirect Access Codes
18000	5	☑	☑	1800070124	BT TEXTDIRECT - TEXT EMERGENCY CALL
180010	8-22	☑			BT TEXTDIRECT - TEXT CALL PREFIX
18001112	8	☑	☑	1800070124	Emergency call remapped
1800150	9-23	☑			Access to TextDirect for textphone users who don't want to use relay
18001999	8	☑	☑	1800070124	Emergency call remapped
180020	8-22	☑			BT TEXTDIRECT - VOICE CALL PREFIX
192	3	☒			Message Advising National DQ Change
194	3	☒			RSC night service
195	3	☑		9116519570124	Blind and Disabled Directory Enquiries
198	3	☑		19870017	Access to 100 for disabled Customers
2*		☒			Do not route.
3*		☒			Do not route.
4*		☒			Do not route.
5*		☒			Do not route.

Dialled Number	Digits	Allow	Emergency	Translate To	Comments
6*		☒			Do not route.
7*		☒			Do not route.
8*		☒			Do not route.
88*		☒			Default prefix for private dial plan calls. Not currently routed in the UK.
9*		☒			Do not route except as noted below.
999	3	☑	☑	99970124	UK Emergency Number
Non-Numeric		☒			Do not route.
Others		☒			Do not route.

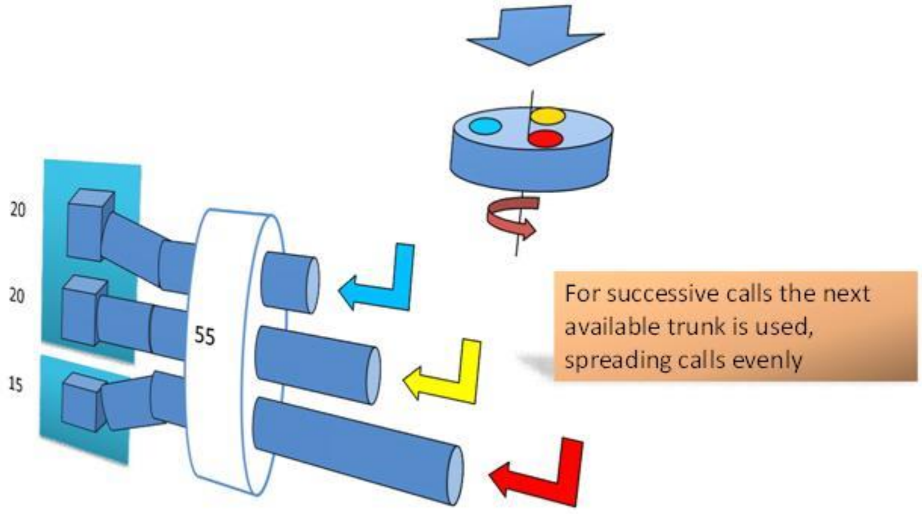
2.6. Call Routing Options

Trunks part of the SIP Trunking service can be configured to route calls in two ways, either Priority Based or Round Robin.

> Priority Based, also known as Overflow, call distribution means that the subsequent trunk in a trunk group is only used when the first trunk has reached the maximum call admission as specified for that trunk.



Round Robin, or load share, call distribution means that the calls are alternating between each of the trunks within that trunk group.



In specifying the role of each trunk in the trunk group, Customers can define in detail how telephone calls should arrive to their estate.

For clarity, trunks that are inactive or out of service will be skipped by the trunk hunting.

2.6.1. Call Diverts

Call diverts are configured on number ranges, and can be set up in advance to be triggered in the event of specific conditions. Options available are as follows:

Call diversion on Busy – Incoming calls to a Trunk that has reached its Call Admission Control limits will be delivered to an alternate destination number (1 number for the entire DDI range). This destination should be a valid UK reachable number.

Call diversion on Error – Where there is an error reaching the PBX terminating a Trunk, (e.g. if the connection to the PBX has failed) calls will be delivered to an alternate destination number (1 number for the entire DDI range). This destination should be a valid UK reachable number. SIP error codes 408, 500 and 503 from the PBX will trigger the call divert on error, as will 2 failed SIP Options messages.

Call diversion Unconditional – All incoming calls will be immediately delivered to an alternate destination number (1 number for the entire DDI range). This destination should be a valid UK reachable number.

2.6.2. Anonymous Call Reject

Incoming calls that have a Presentation number marked as “Anonymous” or withheld (using 141 prefix or using privacy headers) can be rejected at the platform level if required. This is a feature configured against specific number ranges.

2.6.3. Incoming Call Barring

Inbound Call barring can be configured on individual number ranges if required. When activated, all inbound calls to the number range will be barred.

2.6.4. Outbound Call Barring

Outbound call barring is configured on Trunks. Options available are as follows:

Permanent OCB All, including emergency – this will not allow Outgoing calls from this Trunk including emergency calls.

Permanent OCB all, excluding emergency – this will not allow any outgoing calls except those to emergency numbers.

Premium OCB – this will not allow outgoing calls to Premium Rate numbers.

International OCB – this will not allow outgoing calls to International numbers.

Operator controlled OCB – this will not allow outgoing Operator Controlled calls.

Emergency OCB – this will not allow calls to emergency numbers but allow all other calls

2.7. Number Management

The SIP Trunking service includes the ability to request new numbers and to import existing numbers. The numbering is managed centrally on the BT platform and is not restricted by geography – so for example a range of London numbers (0203x) could be applied to IP-PBXs which are physically in Birmingham, Manchester etc.

The service only supports UK Geographic numbers.

2.7.1. New Numbers

Claranet is able to request new geographic numbers for Customer organisations, in blocks of up to 100 numbers. We cannot confirm the number ranges allocated in advance of an order, but will confirm them with the Customer immediately before the order for the ranges is placed.

Please note that ranges are only reserved for 12 hours from the point of a request being made and that reservation cannot be extended. If the Customer is unable to confirm acceptance of the allocated ranges within the 12 hours then new ranges will be allocated and can be accepted or refused.

Please note we will not be able to request additional numbers that are directly contiguous with an older allocated range.

We are able to request numbers from any valid UK code area. Please note that new London numbers begin with 0203. The 0207 and 0208 prefixes are no longer available for new numbers.

All numbers must be associated with a site address assigned to the Trunk Group, for compliance with Emergency Services regulations. BT will use these details to activate the numbers and ensure the Emergency Services database is provided with accurate information in the event of an emergency.

New numbers from allocated ranges are available immediately and have no lead time for provision.

2.7.2. Number Porting

We are able to import Geographic numbers, either as a Single Line or as a Multi-Line Import. Numbers cannot be allocated to a Trunk until the port has completed.

Number porting is a process regulated by OFCOM to facilitate the porting of numbers between providers. However, there is no central register of numbers - numbers may be ported repeatedly and the porting history is held in multiple look-up tables across the different providers and is not centrally managed. As a result number porting can be both time consuming and prone to errors that Claranet cannot control or predict.

It is recommended therefore that Customers seeking to implement a service or site quickly use new numbers in the first instance and then port existing numbers as required at a later date.

2.7.2.1. Single Line

Single Line Porting typically caters for an individual line that terminates onto a socket where one number is provided, i.e. a PSTN line. Where a single line number terminates onto a Feature line service, for the purpose of porting, this is usually classed as a Multi-Line order. Please note this is a minimum lead time and the losing provider may take up to 22 days.

Installation Type	***Minimum Lead Time (Working Days)***
Single Line	10

2.7.2.2. Multi-Line

Multi-Line Porting caters for IP-PBX groups or single lines that terminate on equipment, i.e. ISDN or 11+ single lines at a single address. There are three types of DDI porting requests that come under Multi-Line requests:

1. Multi-Line (30 Lines or Less) – is where main billing numbers and associated numbers terminate on ML equipment, i.e. IP-PBX. In this scenario the numbers have not been built as a DDI range.
2. Multi Line Simple DDI (31 Lines or greater) – is where an entire block of numbers is to be ported over. This includes the main billing number and associated DDI's.
3. Multi-Line Complex DDI – as per Simple DDI but block is to be broken up, with some lines being ported, some being ceased and some remaining on a TDM service such as ISDN.

The above lead-times are based on BT having a porting agreement in place with the provider who is losing the number (the LCP). If this is not in place then please allow an extra 80 days for service establishment to be setup.

Operators with whom BT have a porting agreement:

Operator
AFFINITI INTERGRATED SOLUTIONS
AGGREGATED TELECOM LIMITED
CABLE & WIRELESS PLC/VODAFONE
CABLE & WIRELESS PLC
COLLOQUIUM LIMITED
COLT

Operator
COMMUNICATIONS NETWORK SERVICE
EASYNET GROUP PLC
ENERGIS
EUROBELL
GAMMA TELECOMMUNICATIONS LTD
GLOBAL CROSSING UK TELECOMS
GLOBAL ONE
HULL (KC COMMUNICATIONS)
INCLARITY LTD
INTECHNOLOGY LTD
JEDILLON GRANT LIMITED
MAGRATHEA TELECOMMUNICATIONS
NPLUSONE LTD
OPAL TELECOM LTD
PRIMUS TELECOM LTD
SMALLWORLD MEDIA COMMS LTD
SPITFIRE NETWORK SERVICES LTD
TEAMPHONE.COM LTD
TELEWEST LIMITED
TELSTRA EUROPE LTD
TG SUPPORT LIMITED
THUS PLC
VERIZON UK LTD

Operator
VIRGIN MEDIA LIMITED
VTL (UK) LIMITED
WAVECREST UK LTD
WIGHTCABLE 2005 LTD

Please note if you ask for a number to be ported from another Communication Provider to SIP Trunking, the existing telephone line for that number will cease (if there is one) as a result of the number being ported away from the service. Please ensure that any DSL services are moved before a porting request is made.

Please also note that porting a number away from a Communication Provider will not cease an existing contract with that Provider and cancellation charges may be payable. Customers are recommended to formally cancel their service with other Providers to ensure they are not liable for continuing costs.

As above please note these are minimum lead times.

Installation Type	***Minimum Lead Time (Working Days)***
Multi Line	10
Simple DDI	20
Complex	25

2.7.2.3. Number Porting Scenarios - Import

There are a number of porting scenarios that we use to import numbers, which can make a difference to the porting timescale.

Scenario 1	A BT telephone number (BT is the Range Holder) that is to be ported from the BT Public Switched Telephone Network (PSTN) or VoIP network to SIP Trunking.	Supported

Scenario 2	A BT telephone number, previously exported to another Communication Provider's network, is to be ported to SIP Trunking.	Supported
Scenario 3	Another Communication Provider's telephone number (other CP is the Range Holder), currently on their network, is to be ported to SIP Trunking.	Supported
Scenario 4	Another Communication Provider's telephone number, already imported to the BT PSTN, is to be ported to SIP Trunking.	Supported
Scenario 5	Another Communication Provider's User telephone number, currently on a different Communication Provider's network, is to be ported to SIP Trunking.	Supported
Scenario 6	A range of numbers in a block that belong to 2 or more different range holders or Current Network Operator, referred to as a Mixed Range Holder Port.	Supported
Scenario 7	Porting of Communication Provider's own number ranges hosted on BT's core platform to another BT SIP Trunking or IPX Communications Provider.	Supported

2.7.2.4. Number Porting Process - Import

Porting orders will typically be accepted or rejected within 3 working days and it is not uncommon for ports to be rejected several times before being accepted and a port date issued. The most common reject reasons are due to:

- Single Line request placed but the line turns out to be a Multi-Line request.
- Incorrect installation address.
- Additional numbers on the line that you are not aware of.
- Additional product on the line, e.g. Redcare.

There are a number of checks we will seek to carry out via the OFCOM and BT porting guidance tools to try and avoid these issues but as mentioned above the process is not 100% predictable. Any ports that are submitted with incorrect information will incur a rejection charge.

Claranet can, for a one-off charge, cancel or make changes to a port up to midday, 2 working days before the port date.

2.7.2.5. Number Porting Emergency Restore - Import

In the event of fault occurring during the porting process, it is possible to attempt to restore the port through liaison with Openreach and the LCP. Emergency restoration requests can be submitted up to **13.00 (1pm)** following the day of the port.

Please note it is not mandatory for the LCP to restore ports and there is no agreed lead-time for this process. A restoration order can take many days and may result in a loss of service for a period of time. We will use reasonable endeavours to ensure a restore happens but cannot guarantee full co-operation from the LCP. Other restrictions and limitations may apply on a case by case basis. If additional services, broadband, were on this line then we cannot guarantee that these will be reinstated.

2.7.2.6. Number Porting - Export

Number Export is the reverse of the import process whereby we need to export the number to another Communication Provider service, referred to as the Gaining CP (GCP). This process will be invoked if the Customer is leaving SIP Trunking service for an alternative voice service, and Claranet will be the Losing Communication Provider in this process.

BT is listed as the range holder for numbers used on SIP Trunking and all such port requests will come into BT as they are responsible for applying a routing prefix (Porting Prefix) to the numbers that are moving.

When an export request is made, the GCP will submit a request into BT Openreach which will then be passed to the BT support team to validate. Once BT accept the port they will advise Claranet of the pending export. The Customer should also submit a cancellation notice to Claranet for the number so we can process a billing cease within the terms of your contract with us.

2.7.3. Non-Geographic Numbers

Non-Geographic Numbers are ones that have a code that does not correspond to a part of the UK. Typically these numbers begin with 08 or 03.

The SIP Trunking service does not currently support Non-Geographic numbers on its core platform and we cannot request or port them into the service. This functionality will be delivered later in 2016.

However, Customers are able to source non-geographic numbers via specialist Inbound suppliers and have the non-geographic number directed to a geographic number that is on the platform.

The sequence would be:

Caller Dials 08x number -> call is directed to 3rd party inbound platform -> platform translates 08x number to geographic 01/02 number -> call is directed to the SIP Trunking platform and handled from there.

2.7.4. Outbound Call CLI Classifications

SIP Trunking enables all 5 types of outbound CLI to be used. These types are:

2.7.4.1. Type 1 CLI

A Presentation number that is applied at the network level for all calls generated from a Trunk. This is handled by configuring:

- Presentation Number Source = Default
- Presentation Number Default = The required presentation number
- Any Presentation numbers sent from the PBX are replaced with the Default by WSIPT

2.7.4.2. Type 2 CLI

Presentation numbers that can identify the extension number of the caller. Although the number will be generated by the PBX, the network provider is able to check authenticity.

This is handled by configuring:

- Presentation Number Source = From or PAID (PBX configured to match)
- PBX only sends Presentation numbers that are configured on the Trunk
- Presentation Number Default = Number to be used if PBX sends an invalid number

2.7.4.3. Type 3 CLI

Presentation numbers for callers that may be in a different geographical location to the PBX. With ISDN that may mean that caller's number isn't hosted on the Trunk generating the outbound call. The SIP Trunking service essentially treats Type 3 Presentation Numbers in the same way as Type 2 as SIP Trunking does not limit numbers on a Trunk to a local exchange area:

- Presentation Number Source = From or PAID (PBX configured to match)
- Screening List = all valid numbers or range(s) added
- PBX only sends Presentation numbers that are configured on the Trunk or in screening list
- Presentation Number Default = Number to be used if PBX sends an invalid number

There is a contractual responsibility on the Customer to ensure that numbers sent from the PBX are valid for the originating party (particularly for emergency calls).

2.7.4.4. Type 4 CLI

Where an inbound call is passed back out again retaining the original inbound caller's CLI. On the outbound leg the number is generated by the PBX although should be the same number as the number received on the inbound leg.

Customers wishing to use type 4 services will be asked to sign a contractual commitment that they will only submit CLIs that have been received from the public network. Unlike other types of presentation numbers, type 4 numbers may not always be diallable; this will depend on the nature of the number received on the inbound leg.

The SIP Trunking service does not technically support type 4 Presentation Numbers but can be achieved as follows:

- Presentation Number Source = From or PAID (PBX configured to match)

- Screening List = ranges added to cover all inbound caller Presentation number scenarios
- Presentation Number Default = Number to be used if PBX sends an invalid number

2.7.4.5. Type 5 CLI

Where Presentation CLIs need to be used that are not configured on a Trunk or Trunk Group but the user has permission to use it. Type 5 presentation numbers are generated by the PBX. A typical scenario is a call centre making calls on behalf of more than one client. The SIP Trunking service essentially treats Type 5 Presentation Numbers in the same way as Type 2.

- Presentation Number Source = From or PAID (PBX configured to match)
- Screening List = all valid numbers or range(s) added
- PBX only sends Presentation numbers that are configured on the Trunk or in screening list
- Presentation Number Default = Number to be used if PBX sends an invalid number

There is a contractual responsibility on the Customer to ensure that the numbers sent from the PBX are valid for the originating party.

2.8. Approved Equipment

BT Wholesale operate an Approved Equipment list that details the IP-PBX models that are approved to work with the SIP Trunking platform. This list is regularly updated and includes both IP-PBXs and Gateway devices from a variety of manufacturers, including Avaya, Cisco, Mitel, Siemens Openscape, Panasonic and others.

Please refer carefully to this list before attempting to purchase the SIP Trunking service as it will not work with non-approved equipment.

The Approved Equipment list also indicates if the IP-PBX has been tested with a specific ESBC or Gateway device, and also details any problems found during testing (e.g. fax doesn’t work).

Finally the Approved Equipment list also indicates the software version of the IP-PBX that has been tested. Customers may need to upgrade their IP-PBX to the correct software version to ensure compatibility.

If the Customer has IP-PBX equipment that is not on the Approved Equipment list they may be able to get around that problem by using an approved Gateway device (e.g. Audiocodes, OneAccess) as the point of connection. Claranet will not supply or configure any Gateway devices and Customers should work with their IP-PBX supplier to ensure that their equipment meets the Approved Equipment requirements.

While it is possible to request an IP-PBX or Gateway model that is not on the list be tested for approval against the platform by BT Wholesale, there are a number of restrictions that mean Claranet cannot guarantee this. Those restrictions include:

1. BT Wholesale control entry to the process and will only proceed if there is a good commercial business case for the effort involved. IP-PBXs with low market share are unlikely to be accepted as the return will not justify the effort.
2. Older versions of IP-PBXs already on the Approved Equipment List will not be accepted. Customers should work with their supplier to ensure they are up-to-date with their Manufacturer’s software versions.
3. Testing even when approved can take some months, and is not guaranteed to be approved.

Claranet can apply for testing on behalf of a SIP Trunking Customer with an IP-PBX that is not on the Approved Equipment List and will advise of acceptance and timescales.

2.9. Audio Codec's

SIP Trunking is a VOIP service, which implies that the audio conversation is digitally encoded for transport over the IP network, and this encoding is done using codec's. There are a currently a large number of codec's available in the market.

For on-net traffic between sites of an enterprise as well as for traffic between Customers on the network, the SIP platform will allow the end-points to negotiate the appropriate codec for the call directly and will not restrict the types of codec used, nor does the platform offer mediation/transcoding of the codec for these calls.

Therefore, for all Customers to be able to talk to all Customers on the platform, at the minimum the IP-PBX of the end Customer needs to support G.729a. It is recommended that IP-PBX is configured with both G.711 and G.729a otherwise a call from differing codec's where only one has been configured on the IP-PBX calls will fail.

For the calls handed from/towards the UK PSTN, there is a choice between the following codec's:-

2.9.1. G.711 a-Law & u-Law

This ITU codec dating back to 1972 is the encoding used in the national PSTN network as all voice conversations are already digitally transmitted. Using this codec therefore implies that the voice samples do not need any transcoding until they reach the distant telephone exchange at the far end. There are two slightly different versions; µ-law, which is used primarily in North America, and a-law, which is in use in most other countries outside North America.

G.711 u-law tends to give more resolution to higher range signals while G.711 a-law provides more quantization levels at lower signal levels

2.9.2. G.729 annex A

This codec gives a decent voice quality, but automatic voice response units may struggle with it. Given that the compression is a lossy compression, care needs to be taken not to have multiple transcoding hops using low bit rate codes in a call. Note that G.729 annex B is not supported.

2.9.3. AMR-NB

The Adaptive Multi-Rate (AMR or AMR-NB or GSM-AMR) audio codec is an audio compression format optimized for speech coding. AMR speech codec consists of a multi-rate narrowband speech codec that encodes narrowband (200–3400 Hz) signals at variable bit rates ranging from 4.75 to 12.2 kbit/s with toll quality speech starting at 7.4 kbit/s.

2.9.4. G722

G722 is an ITU-T standard 7 kHz Wideband audio codec operating at 48, 56 and 64 kbit/s. It was approved by ITU-T in November 1988. Technology of the codec is based on sub-band ADPCM (SB-ADPCM).

G722 provides improved speech quality due to a wider speech bandwidth of 50–7000 Hz compared to narrowband speech coders like G.711 which in general are optimized for POTS wireline quality of 300–3400 Hz. G.722 sample audio data at a rate of 16 kHz (using 14 bits), double that of traditional telephony interfaces, which results in superior audio quality and clarity.

2.9.5. G722.2

G722.2 is another ITU-T 7 kHz wideband codecs include which is not a variant of G.722 and it uses different patented compression technologies. G.722.2, also known as AMR-WB ("Adaptive Multirate Wideband") is based on ACELP and offers even lower bit-rate compressions (6.6 kbit/s to 23.85 kbit/s),as well as the ability to quickly adapt to varying compressions as the network topography mutates. In the latter case, bandwidth is automatically conserved when network congestion is high. When congestion returns to a normal level, a lower-compression, higher-quality bitrate is restored.

Please see the table in the Design Guidance section below to understand how codec selection influences bandwidth consumption.

2.10.Fax Support

The SIP Trunking platform supports Group 3 fax transmission using T.38 and G.711 up-speed when used with compatible CPE. Due to the wide variety of proprietary extensions to fax standards, BT does not guarantee interworking between all fax terminals and Customers should be aware that T.38 as a standard does not support all extensions used by G3 fax machines, such as the v.34bis transmission speeds used by Super Group 3 fax terminals.

The SIP Trunking service supports the following in dealing with fax (and expects compatible CPE to also support these)

- T.38 Fax Relay, as per ITU T.38 Annex D standard
- G.711 pass-through, in which case the platform expects from the CPE support for G.711 pass through of fax modem signals, with the ability to disable echo cancellation and dynamic jitter buffers on a per call basis.
- G.711 up speed, in which case the platform expects the CPE to have the ability to up speed from G.729 to G.711 if a fax tone is detected, using a SIP re-INVITE mechanism.

2.11.PDQ Machines support

Unfortunately, these devices have problems and may not work with any consistency over a VoIP connection. This is due to the wide variety of machines and different implementations of them. The codec's used by ATAs have been designed to compress voice, not the analogue signals sent and received by modems.

We use G.711 pass through for these calls, where the call is carried in a VoIP call encoded as audio. This is sensitive to network packet loss, jitter and clock synchronization. When using voice high-compression encoding techniques such as, but not limited to, G.729, some tonal signals may not get correctly transported across the packet network. It is always advisable to set the machines to G711.

Claranet will not guarantee any PDQ machines will work.

2.12.Integrating an alarm system with VoIP

Alarm companies are currently coming to grips with the realization that they will need to make their services work with VoIP. Some companies can get their equipment to make the necessary communications and line seizure, but some have not and will not support the service over VoIP. Please check with your Alarm System supplier.

3. Designing The SIP Trunking Solution

Claranet will work with the Customer to design the voice solution on two levels:

Statement of Works (SoW)

This will only be required in circumstances where a larger network order is also required to deliver the additional IP bandwidth required to supply the SIP Trunking service.

Configuration Data Capture Form (DCF)

This is a specific document that captures the setup of the SIP Trunking service that Claranet will configure for the Customer. It will detail all the Channels, Trunks and Trunk Groups, what numbers are associated with each and how any routing options will function.

This document will be supplied to the Customer early in our engagement with you. The form is available on the Claranet Online Service Catalogue.

The responsibility matrix for the Customer solution design is:

Activity	Claranet	Customer
Customer Pre-Work		
Identify list of numbers to be ported.		x
Identify the Losing Communication Provider		x
Confirm Customer Equipment is on Approved List	x	x
Customer Solution Design		
Document network design including any bandwidth upgrades	x	x
Document trunk, channel and number requirements in DCF	x	x
Agree SOLUTION DESIGN DOCUMENT if required.	x	x

3.1. Bandwidth Guidelines

This section covers guidance for Solution Architects and Customers on how to assign the correct amount of IP bandwidth for the SIP Trunking service.

SIP Trunking consumes IP bandwidth for each concurrent call that the IP-PBX is handling in or out of the Customer network. It is therefore critical that the Customer sites that have the IP-PBXs on them are provisioned with sufficient bandwidth for the number of channels required to service the site.

Where possible, Quality of Service configurations should also be deployed to ensure that the voice calls are not negatively affected by restricted or congested bandwidth at the Customer site. This is particularly necessary where there are many users at a site. Please note that QoS or Q-in-Q must be ordered separately to the SIP Trunk order.

The Customer’s IP-PBX will be configured to use a specific supported codec (detailed above) and this will define how much bandwidth each concurrent call channel will consume.

It is important to note that the call will use both upstream and downstream bandwidth in the same amount, so it is recommended to use Ethernet connections wherever possible to ensure the site has synchronous bandwidth. Asynchronous services such as ADSL or FTTC VDSL will have problems if their limitations on upstream bandwidth are exceeded.

Claranet Connectivity, either MPLS or Internet-facing and using Q-in-Q will provide the best connectivity for SIP Trunking. Internet connections without Q-in-Q will introduce a significant element of risk as without protection for the voice traffic, calls may drop or suffer quality issues which cannot be solved.

Please refer to the Quality of Service and Elevated Traffic guidelines in the Networks portfolio for detailed guidance on setting up MPLS networks for SIP Trunking.

Claranet Product Management have agreed the following guide, based on the number of concurrent calls from a site using the G711 codec:

Concurrent Call- Range	Sync Bandwidth Required	Minimum Connection	Recommended Connection
0-9	1Mbps	Dedicated ADSL2+ with Elevated Traffic	FTTC or EFM
10-35	3.5Mbps	FTTC @ 20Mbps	EFM or Fibre Ethernet @ 10Mbps with Q-in-Q
35+	3.5Mbps+	EFM @ 10Mbps	Ethernet @ > 10mbps + QoS

Claranet does not recommend under provisioning or use of alternative audio codecs to fit more users or concurrent calls into a site – the degradation in call quality will result in failed calls and trouble tickets that cannot be resolved. Bandwidth upgrade should always be preferred.

The figures above assume maximum throughput on all connections and no contending applications, and represent a ‘best case’ position. If there is any doubt, Customers should work with Claranet Solution Architects to ensure that the whole picture in terms of bandwidth consumption by all applications is understood and the SIP Trunking is incorporated into their network requirements.

Other Codec’s use different bandwidth amounts and care should be taken in calculating the amount of bandwidth allocated to the SIP Trunk, please see below:

Codec	Bandwidth per Channel Including Ethernet Headers
G.722	102 kbps
G.722.2 (AMR-WB)	43 kbps
G.711a	102 kbps
G.711u	102 kbps
G.729	43 kbps
G.729A	43 kbps
AMR-NB	48 kbps

Any orders for new or upgraded Customer site connectivity should be submitted separately ahead of the SIP Trunking service order and should be completed and enabled before the SIP Trunking service is deployed.

Consideration should also be given at this stage as to whether additional capacity is going to be needed in the immediate future (i.e. within 3 months of the planned rollout) as bandwidth increases may also need to be planned in.

3.1.1. Firewall Settings

We recommend that SIP ALG is turned off. Not all firewall configurations need ports to be opened. If the Customer is running inside to outside rules then ports should be opened to allow the SIP protocols out. There should be no reason for the Customer to open ports inbound on the firewall. See table below with the TCP/UDP ports required for SIP Trunking operation:

Protocol	BT SBC IP Address	Customer SBC/PBX IP Address	Source Ports	Destination Ports
SIP (Signalling)	Confirmed at order	Configured on portal	UDP 5060 or TCP 1024-65535	UDP 5060 or TCP 5060
RTP (Media)	Confirmed at order	Configured on portal	UDP 32768 - 65535	UDP 32768 - 65535

3.1.2. Number Requirements

It is critical to get an accurate gauge of the volume of numbers that the Customer requires and it is a good idea to over-provision numbers at the point of the original order so that contiguous numbers can be provided for new users as the Customer grows.

Several of the IP-PBX features such as hunt groups may require additional phone numbers which should be identified early and included in the initial Customer order:

3.2. Pricing and Billing

Claranet will provide a commercial quote that will include the following elements of the SIP Trunking service.

3.2.1. SIP Trunk Groups

This is the logical group of channels for each IP-PBX. There is no charge or term for these.

3.2.2. SIP Trunk Channels

These are charged at a per-channel, per-month rate which varies depending on commercial model and which type of channel is provided.

3.2.3. Call Usage & Telephone Numbering

Call usage is billed each month, and can be billing can be tracked via the Claranet Online portal. A detailed destination rate-card is available on request.

The billing data on the Claranet Online portal is updated each month.

New and ported numbers are billed as one-off costs either per number or if porting in a block, per block of numbers.

3.2.4. Billing

The invoice will not always reflect accurately the distribution of channels and numbers across the Customer estate – it is not an inventory.

Billing will be triggered for the different line items as we progress through the order, so billing for user licenses will be triggered before completion of number porting.

3.2.5. Processing your order

Once we have received a valid order including an approved Statement of Works (if required) and the first section of the Data Capture Form completed, we will process your order and assign a Project Owner to progress the completion of the Data Capture Form.

If the project does not progress as expected due to other issues (e.g. LAN or WAN upgrades, personnel changes within the Customer organisation), after 30 days without activity the order will go on hold and other orders may be prioritised, in order to maintain efficiency of order management and delivery within Claranet.

3.2.6. Professional Services

Claranet may assess some orders for additional Professional Services charges if the work involved is particularly complex or lengthy.

4. Implementing Your Service

Based on the information in the order and Data Capture Form, Claranet Project Management will work with the Customer to plan the deployment of the SIP Trunking service. This will include:

1. Arranging number porting, in batches, bulk or individual numbers.
2. Confirming IP addressing for SIP Trunk configuration on the IP-PBX.
3. Ordering numbers and channels.

These actions will lead to a planned ‘go-live’ event for the Customer when they will start using the new SIP Trunking service. A ‘go-live’ document will be constructed that the Customer will sign off when ‘go-live’ is completed which will confirm we have delivered the service to the Customer to their satisfaction; the site and service will then move into in-life support.

4.1. Implementation Responsibility Matrix

Pre-SIP Deployment Orders	Claranet	Customer
Book and order any network requirements – new connections or upgrades to existing.	x	x
SIP Trunking Order Placement		
Order Trunk and Channels	x	
Request any new numbers to be made available	x	
Plan number porting requests	x	x
Advise Trunk IP addresses and number assignments	x	x
Configure IP-PBX to use new trunk		x
Go Live		
Confirm IP address for new trunk is live	x	
Configure IP-PBX/SBC/Gateway to point to new IP address.		x
Test user experience on site and ensure calls working as expected.		x
Sign off site		x
Initialise billing	x	
Close Project & Handover to in-life	x	

4.2. Implementation Timescales

There are a number of factors in the implementation of a SIP Trunking service and Claranet is not in control of aspects such as IP-PBX or Gateway configuration tasks.

From the point of order to full live deployment can take three to four weeks depending on scale and complexity. Porting of numbers in particular can add significantly to deployment timescales.

4.3. Implementation Key Milestones

The list of steps below is not exhaustive and not necessarily consecutive (steps 3 and 4 are usually completed while the porting request is in flight), but indicates the key milestones in a SIP Trunking delivery and how long they usually take to complete.

	Detail	Duration
1	Complete Data Capture Form and sign off with Customer.	2-3 Working Days
2	Order to BT for network interconnect VLAN and any porting requests.	Single line:10 Working Days (minimum); multi line: 30 days (minimum) from the point of acceptance of porting order by BT.
3	Submit order to BT for etherflow connection.	1 Working Day
4	Complete & test Claranet network config and apply QoS	2 Working Days.
5	SIP Trunk built and tested against Customer IP PBX, DDI numbers applied to Trunk.	4 Working Days
6	SIP Trunk handed over complete to Customer	1 Working Day.

4.4. Signing off a site

Once Claranet have set up all Trunks and Channels in line with the initial configuration document, and all number provision and porting completed, the Customer will be asked to confirm that a site is complete and can be 'signed off'.

The Customer will then have 5 working days to raise any issues, after which the service will be designated as delivered and passed into in-life service.

Where a Customer has multiple sites we will seek to collate all the signoffs into a single document to avoid repetition.

5. In Life Service

Claranet SIP Trunking a fully managed service – responsibility for implementing any in-life changes to the SIP Service rests with the Claranet. We will not however offer any support on the configuration of the IP-PBX or Gateway device.

5.1. In Life Changes Responsibility Matrix

In-Life Administration Tasks	Claranet	Customer
Add new or ported numbers	x	
Add new channels	x	
Change call routing rules	X	
Create new trunk site and roll out new site (new project)	x	X
Request any new numbers and any number porting	x	X
Configure new numbers for sites and users (PBX)		X

5.2. Chargeable and Non-Chargeable Changes

5.2.1. Chargeable Changes

Changes that require alteration to billing (e.g. new number ranges, additional channels or trunks) will require an order to be raised with your Account Team, and will be actioned by Claranet Delivery in the same way a new order would.

5.2.2. Non-Chargeable Changes

Changes to the configuration of the SIP Trunk service that do not require a billing change (e.g. change of divert or call barring rules) will not require an order to be raised. These should be raised as change requests via Claranet Support.

Claranet may assess that a change requires a Professional Services charge due to its complexity and will advise accordingly.

5.3. Tickets and Response Times

Claranet's Service Desk is available to respond to queries 24x7x365. Details of how to log request tickets, incidents and general Service Desk contact information can be found in the Service Operations guide.

Claranet grade service tickets on a severity scale, which dictates our response time. Details of this can also be found in the Service Operations guide.

5.3.1. Raising a Ticket

Customers will be advised during the Delivery process of their process for raising tickets,

Once a ticket is raised, Claranet Support will triage the issue and ascertain if there is need to raise a further ticket on our suppliers, either of the site connectivity or of the SIP Trunking core platform itself.

5.3.2. Platform Partner Ticket Management

Please note we will not raise a ticket with our platform partner until we have ascertained that there is a problem with that element of the service – our first actions will be to work through a troubleshooting process to find out what element of the end-to-end service is at fault, and whether it is something we can resolve.

If we ascertain that there is a problem with the core platform, we will raise a ticket with our supplier. Their agreed response times to us are as follows:

BT Severity Grade	Description	BT Response Time	BT Target Restoration
P1	Critical Outage	2 Clock Hrs, 24x7x365	4 Clock Hours
P2	Major Impact	4 Working Hrs, Mon-Fri 8am-6pm	8 Working Hours
P3	Minor Impact	8 Working Hrs, Mon-Fri 8am-6pm	24 Working Hours
P4	Information Only	5 Working Days, Mon-Fri 8am-6pm excl Bank Holidays	n/a

In practice this means that serious P1 outages will be responded to within 2 clock hours, 24x7x365, with target restoration in 4 clock hours. This would include total or partial platform failure or failure of a key platform element such as PSTN interconnect or internet connectivity to the platform. Individual site failures are likely to be classed as P2, individual channels as P3.

Please see the matrix below to see how tickets of different types will be handled within these matrix models.

5.3.3. Telecoms Fraud

The SIP Trunking service benefits from a strong anti-fraud system operated by BT as part of the overall service. This monitors usage volumes and patterns 24/7/365 to look for unusual or suspicious call activity. BT will act to shut down specific trunks or trunk groups if they believe fraudulent call activity is taking place.

However, this does not protect Customer IP-PBXs from intrusion and fraudulent use, and Claranet are not liable for the costs of calls made from Customer's IP-PBXs before BT act to shut down the trunks. We recommend that Customers work with their IP-PBX supplier to protect their platform from fraudulent use.

Customers whose IP-PBXs are subject to fraudulent use will be invoiced for the cost of the calls made.

Customer Incident	Symptoms	Claranet Priority Level	Claranet Actions	Likely causes
Site/Multiple Site Total Failure	All users affected	P1	Check MPLS network is live.	MPLS network connection failure
	No calls in or out.		Check core voice service (other Customers reporting, Claranet internal deployment)	Interconnect failure
			Raise ticket with BT if network is up and service is down.	Core platform failure.
Individual call failure	Multiple calls affected but not total failure.	P1	Check MPLS network is functioning correctly.	MPLS network bandwidth exceeded
	Calls failing to connect or dropping.		Check core voice service (other Customers reporting, Claranet internal deployment)	IP-PBX configuration issue.
			Check IP routing - any bandwidth restrictions	
			Check users are setup correctly - any changes made by Customer?	
Inbound Call Routing Failure	Inbound calls going to wrong site or failing.	P3	Check Inbound call routing rules.	MPLS network bandwidth exceeded
			Check connectivity is sufficient for call volume.	Inbound Call Routing incorrectly set.
Poor Call Quality	Calls connect but are of poor voice quality	P3	Check core voice service (other Customers reporting, Claranet internal deployment)	SIP Platform problem
	One-Way speech		Check IP routing - any bandwidth restrictions	Dropping packets across network
	Delay/echo/distortion		Raise ticket with BT if network is OK and user experience is failing	

5.4. Planned Outages and BT Terms

Planned outages will be communicated to Customers as soon as possible when we are notified by our platform partner. Some may be emergency notifications and therefore we cannot give a minimum notice period.

Customers should always provide Claranet with a named contact and email address that we can send notifications to so that they will be picked up and recorded.

These terms are passed through to our Customers from our Platform Partner:

BT are able:

- to interrupt the Service for operational reasons (including planned maintenance) where it is reasonable for BT to do so or because of an Emergency. BT agrees to restore the interrupted Service as quickly as reasonably possible and, BT will give Claranet as much notice as possible, unless due to an Emergency it is impracticable to do so.
- to take action to protect the Service if a Customer or end user is using the Service in a manner that is damaging to the Service. This may involve BT taking actions to block or restrict end user equipment from accessing the Service. BT will inform Claranet of any action taken pursuant to this Paragraph as soon as reasonably practicable.

5.5. Service Credits

The purpose of the Service Credit model is to offer Customers assurance that events that impact their ability to make & take calls (a critical business requirement) are treated with priority, and that Claranet continues to take action until the service is restored.

The complex nature the IP supply chain between the Customer endpoint and the core platform means that we cannot offer an end-to-end service credit agreement, but instead offer a Core Platform service credit model.

5.5.1. Service Credit Calculations

Claranet SIP Trunking has a Core Platform availability KPI of 99.95%.

Based on 730 hours or 43,800 minutes in each calendar month, this means that the Core Platform should be available for:

729.63 hours or 43,778.1 minutes in each calendar month.

Availability of 99.95% indicates a total of 21.9 minutes of Core Platform outage in each calendar month before a Service Credit is payable.

The scale of credit payment would be:

Availability in a month	Service Credit Percentage
99.95% – 98.50%	15%
98.50% – 97.00%	30%
97.00% – 95.00%	50%
Less than 95.00%	50%

Outages are measured from the point that Claranet responds to a ticket that the Customer submits either via Claranet Online of via telephone to the Claranet Service Desk and to the point where Claranet advise that Customer service has been restored.

<u>Example</u>

A Core Platform outage lasting 120 minutes for 50 Channels.

120 minutes – 21.9 minutes = 98.1 minutes of Service Credit applicable outage or 99.77%

Availability bracket 99.95%-98.50% = Service Credit 15%

Standard Monthly Price of Standard Channel = £5.50 per channel.

15% of Standard Monthly Price of Standard Channel = £0.82 per channel.

Total Service Credit Payable:

50 x £0.82 = £41.25

5.5.2. Exclusions

- Usage costs are not subject to Service Credits.

Additionally, Claranet will have no liability for Service Credits due to or as a result of any of the following reasons:

- Non-availability of Claranet connectivity services (including any associated CPE such as EDDs and routers) – only the service level agreement for the connectivity service shall apply.
- Non-availability of non-Claranet connectivity including CPE – Customers should pursue their ISP for financial recompense.
- Non-availability of internet access or Claranet connectivity due to cyber-attack.
- Issues related to Number Porting, Number Migration or New DDI provisioning other than where Claranet is at fault.
- Any planned or emergency maintenance of the Core Platform, whether notification has been issued or not.
- The failure of any CPE that the Customer is responsible for managing that impacts the service e.g. local network equipment, network firewalls etc.
- Customer failure to follow and comply with any training or reasonable instructions given by Claranet regarding the Service.
- The use of the Service for a purpose for which it was not designed or specified for.
- Any Force Majeure Event.
- Suspension of service in accordance with the terms of the contract (e.g. for fraudulent use).
- Customer default or delay, or any negligent, wilful or reckless act, fault or omission by the Customer, or any of their representatives, employees, agents or sub-contractors.
- Access issues and delays along the route of the Services across Claranet or other core IP network, or at the Customer Sites e.g. problems with the LAN.

5.5.3. Service Credit Request Process

- Customers must apply in writing to their Account and/or Service Manager for Service Credits.
- Customers must apply within 30 days of the claimed outage.
- Customers must present evidence (ticket numbers and timings) to show the period of time that they are claiming as an outage.
- Claranet should review this evidence in light of the exclusions above to see if the claim is valid.
- Assuming claim is valid, Service Credit should be applied at the next available billing run.

5.6. Leaving the SIP Trunking Service

The SIP Trunking service can be exited reasonably easily either in part or in whole as a Customer requires.

5.6.1. Removing Individual Channels

The total number of channels that the Customer has can be amended on a monthly basis if required, and will require an order to be raised to change the service as it is commercially affecting.

5.6.2. Removing Trunks & Trunk Groups

Cancelling a trunk will remove all of the channels within that trunk, and cancelling a Trunk Group will remove all the Trunks and Channels within that Trunk Group. The numbers associated with that Trunk Group will also be cancelled.

If the Customer is seeking a reconfiguration of the service rather than a cancellation (e.g. to move the main IP-PBX to a new location or change which is the primary site), this should be approached as a change rather than a cancellation.

5.6.3. Migrating numbers away from SIP Trunking

It is possible to migrate numbers away from the service if required. In this action, Claranet are the Losing Communications Provider (LCP) and the porting process will be led by the Gaining Communications Provider (GCP).

The Customer will need to provide a Letter of Authority to the GCP to trigger the port.

Porting the numbers away from the service will stop inbound calls being routed to the Claranet SIP Trunking service, but will not cancel the service or prevent outbound traffic from the SIP Trunk, for which the Customer will still be charged. Customers wishing to cancel their service should do so formally via the cancellation team.

5.6.4. Exiting the Service

If a Customer wishes to leave the service at the end of their contract they are able to do so by completing the following actions:

1. Porting their telephone numbers out of the service. The gaining provider may charge for this activity.
2. Cancelling all Trunk Groups via a Cancellation notification to Claranet.
3. Removing any voice VLANs or VRFs from their network equipment that point to the IP addresses of the Claranet SIP Trunking service.

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