1. Service Overview

We offer the SIP Trunking Service to connect your IP-enabled PBX/Gateway to the PSTN via high availability Session Border Controllers.

The Service is available from your sites using our WAN Service, within our PoPs or over the internet.

Additional information is included in your Order Form and Statement of Work.

2. Service Features

This section describes the features of the SIP Trunking Service.

2.1 SIP Trunking

The SIP Trunking Service provides inbound and outbound telephony connectivity to and from the PSTN using one or more SIP Trunks which comply to the sections of IETF RFC 3261 (https://www.ietf.org/rfc/rfc3261.txt) needed for the implementation of the Service.

We offer the following:

(a) **Standard SIP Trunk** - your PBX/Gateway is connected to a single high-availability Session Border Controller cluster; or

(b) **Resilient SIP Trunk** - your PBX/Gateway is connected using two SIP Trunks of equal capacity to high-availability Session Border Controller clusters located in two of our geographically diverse PoPs.

Your selection will be specified in the Order Form or, if it differs from our standard offering, in the SoW.

The dimension of the SIP Trunk is measured in SIP Channels, the maximum number of concurrent voice calls that can transit over the SIP Trunk, as stated in the Order Form. The number of SIP Channels on a Resilient SIP Trunk is equal to the number of one of the two SIP Trunks. For example, a standard or resilient 30 channel trunk will have 30 channels available at any time.

Call Admission Control will be configured to safeguard capacity in our network and to ensure the contracted number of SIP channels is not exceeded.

Inbound traffic from the PSTN can be presented to multiple PBXs or Gateways (endpoints) either in Active-Standby or using Round Robin. Conversely, we accept traffic destined towards the PSTN from you in either Active-Standby mode or using Round Robin with a resilient SIP Trunk build.

We support SIP options messages to proactively ascertain PBX/Gateway availability. If you select this option, we will automatically re-route SIP traffic in the event of failure of your PBX/Gateway where a secondary PBX/Gateway is available.

SIP refer messages which can be used as part of the call transfer process are not supported on our SIP Trunking Service, whereas SIP re-invites are supported.

2.2 Voice Encoding

The SIP Trunks will be configured to use one of three codecs below for voice media encoding:

Codec	Companding	Compression	Packetisation Time
G.711	A-law (preferred)	None	20ms
G.711	µ-law	None	20ms
G.729	N/A	ACELP	20ms

We support DTMF codes according to RFC2833 and fax according to ITU-T T.38.

2.3 On-Net SIP Trunking

The SIP Trunking Service is available “On-Net”, which means from your sites connected via our WAN Service. The PBX/Gateway on an On-Net Site must be connected to a LAN which is directly connected to our WAN Service CPE. On-Net Sites can be connected as below:

(a) WAN Services dedicated to the SIP Trunking Service only. We commit to end-to-end voice quality where you use ethernet services (fibre/copper lines or ports in our PoPs) and where the bandwidth available is at least 100kbit/s per SIP Channel. We cannot commit to end-to-end voice quality where for broadband services are used as the access service.

(b) WAN Services where SIP Trunking Service is shared for multiple traffic types. We commit to end-to-end voice quality where you use ethernet services (fibre/copper lines or ports in our points of presence), the QoS is enabled and the bandwidth available for SIP Trunking is at least 100kbit/s per SIP Channel. We cannot commit to end-to-end voice quality where broadband services are used as the Access Service, and we do not recommend sharing other traffic types over a broadband connection.

The QoS marking for SIP Trunking traffic is as follows:

(a) Signalling → DSCP 40

(b) Voice media → DSCP 46

2.4 Off-Net SIP Trunking

The SIP Trunking Service is also available as an off-net service, which means that the SIP Trunk will be connected over the internet to our off-net session border controllers.

Access to our off-net SBCs will be permitted from a specified public IP address per endpoint which will be agreed during Service Implementation. We cannot commit to end-to-end voice quality where the internet is used for part of the transport of the SIP Trunk.

2.5 ISDN ↔ SIP Converter

We will provide with ISDN ↔ SIP Converter(s) in the number indicated in the Order Form. Each converter has either 1 or 2 ISDN connections which comply with Q.931 provided over RJ45 G.703 interfaces. 75ohm coaxial and 120 ohm twisted pairs are supported with an optional balun. We will install the converters in racks on your site(s). Converters are available for both types of connection described in Section 2.3 above.

2.6 Fraudulent Traffic

We undertake measures to identify potentially fraudulent traffic, and we will attempt to contact you in this case. We reserve the right to block all traffic, acting reasonably, where we identify potentially fraudulent traffic.

We cannot guarantee that we will identify all potentially fraudulent traffic, and we will charge you for all calls fully initiated from your telephony equipment or on the SIP Trunk, whether it is fraudulent or otherwise.

2.7 PSTN Numbering

Your SIP Trunking Service will be allocated new UK telephone numbers or we will reasonably endeavour to port your existing UK telephone numbers to the SIP Trunking Service.

If you select the option of having new PSTN numbers, we will provide you with new number ranges as described in the Order Form, within most UK geographic number localities and/or within the non-geographic ranges 03, 08 and 09.

If you require us to port your existing PSTN numbers, we will take reasonable endeavours to migrate your existing number(s) ranges to your SIP Trunking Service through porting, whether UK geographic numbers (01 and 02) or UK non-geographic numbers (03, 08 and 09). The lead time to complete porting and, also the options for any out of hours migration, depend upon the provider you currently use in addition to the size of the number range to be ported.

2.8 Call Rate Limits

The maximum call rate that the Service will support depends on the number of voice channels ordered as set out below.

SIP Channels	Calls per Second (CPS) Limit
50	3
100	4
200	5
300	6

SIP Channels	Calls per Second (CPS) Limit
400	7
600	9
800	11
1000 or above	12

The number of SIP channels will be specified in the Order Form or, if it differs from our standard offering, in the SoW.

2.9 Call Features

The SIP Trunking Service includes the following features:

(a) **Number Presentation** - this feature enables you to replace the outgoing Calling Line Identity alternative number that you specify. Number presentation is only available where the far-end carrier supports the feature.

(b) **Calling Line Identity Restriction** – this feature enables you to restrict the presentation of the CLI so the called party would not see it. Calling Line Identity Restriction is only available where the far-end carrier supports the feature.

(c) **Calling Line Identity Presentation** – this feature allows the called party to receive and display the calling party's identity before answering the call. The called party will only receive this information if the caller has not restricted the sending of their number and where far-end carriers support the feature.

3. Service Implementation

After order acceptance, we will implement the Service as follows:

Service Implementation	Us	You
We validate timescales with suppliers for WAN Services (see Managed WAN Service Description), if required.	●	
If required, we hold kick-off call or meeting to validate the DDI blocks/porting and, where applicable, to agree network set-up.	●	
We communicate your projected Ready for Service Date.	●	
We validate timescales with suppliers for WAN Services (see Managed WAN Service Description), if required.	●	
We will set up and configure SIP Trunking Service.	●	
For On-Net SIP Trunks, we will deliver new L3VPN or reconfigure existing L3VPN as necessary. For Off-Net SIP Trunks, we will activate the SIP Trunking Service. We will deliver and install converters, where required.	●	
You will configure the PBX/Gateway and test the new SIP Trunks.		●
We will confirm the Ready for Service Date and provide our Handover Pack.	●	

Service Implementation	Us	You
You will cut over from the existing service to the SIP Trunking Service.		●

All implementation work will be carried out during Business Hours. We will invoice additional charges for any work undertaken out of hours. We reserve the right to charge additional fees for any change in your requirements occurring after the implementation of the Service and outside the scope of the contract (as described in the MSA).

3.1 Ready for Service

The Ready for Service Date of the SIP Service is when we (acting reasonably and properly) notify you and will be the latter of:

(a) The date of the configuration of a Standard SIP Trunk or either trunk of a Resilient SIP Trunk on our network;

(b) Where applicable, the installation of connectivity to your site; or

(c) Where applicable, the delivery of CPE (e.g. router) to you for installation on your site.

This date may be specified to occur at a later date, if mutually agreed to be so. If you consider that the Service is not ready for service on the date notified, you must notify us within two (2) weeks of our notification, after which the Ready for Service date will be deemed to have occurred regardless of issues or defects of which you may later become aware and notify us about (which will then be dealt with as faults or service issues but will not change the Ready for Service Date).

Note: for Resilient Access, the Ready for Service Date of one Access Service may be different from Ready for Service Date of the other Access Service.

4. Billing

The Service will be billed as Set-Up Fees, Recurring Fees and Usage Fees as described in the Billing Guide.

The Usage Fees will be charged monthly in arrears based on your Rate Card. This is a separate document that we will provide you and which defines the charge per destination calculated based on the number of seconds of usage. The charge is rounded up to the nearest penny. We may update the Rate Card provided that we give you at least four (4) weeks of notice of the new charges per destination.

4.1 Rate Cards and Call Packages

SIP Trunking Services are provided with a rate card or call package. All traffic received on the SIP Trunking Service will be charged to you according to the rate card and any call package which applies, as specified in the Order Form or, if it differs from our standard offering, in the SoW. In the event that the call package allowance is exceeded, the rate card will apply.

The rate card applies to outbound traffic and inbound traffic to non-geographic numbers. Calls are charged (or revenue share is granted) based on the rate card, on a price per minute and/or price per call basis. Prices are set for daytime, evening and weekend times.

Call packages exist for SIP Trunking Services for calls to UK Geographic Destinations (numbers starting with 01 and 02), UK National Non-Geographic (numbers starting with 03) and to the 4 major mobile providers (EE, O2, Three and Vodafone). Thereafter, the rate card applies.

5. Service Operations

Please refer to our Operations Manual for further information on incident management, requests for change and information requests.

6. Dependencies

We will provide the SIP Trunking Service subject to the following dependencies:

6.1 Client Obligations

(a) You will designate a project lead and technical contacts with the appropriate level of expertise to support the successful implementation and operation of the Service;

(b) You will provide us with all of the information needed to complete the configuration of the Service;

(c) You will manage any of your own third parties needed during Service Implementation or Service Operations;

(d) During the installation of the Service, it is your responsibility to show our installation engineer (or our third-party engineer) to the installation point;

(e) To ensure the smoothest possible porting process for any existing telephone numbers that you would like to transfer to your Service, you will provide us with a Letter of Authority containing accurate information within five (5) days of your order;

(f) For all Calling Line Identifiers ported or provided on the Service, you will provide us with name of company and suffix (plc, Ltd, etc.) so that we may submit this information to the Emergency Services.

(g) You will provide comms room space in your site(s) for the CPE;

(h) You will provide IP addressing for delivery of the SIP Trunking Service;

(i) You will ensure that your PBX/Gateway is configured and ready for the integration of the SIP Trunks before the cut-over date, including appropriate differentiated service code point marking for QoS;

(j) The use of the Service must be in line with OFCOM regulations and best practices. We reserve the right to disconnect the Service if we believe these regulations and best practices are not being followed; and

(k) If you require us to port your existing numbers, you will provide us with a letter of authority.

6.2 Service Dependencies

(a) Number presentation and CLIR features are only available where supported by the far-end carrier;

(b) The CLIP feature is only available where supported by the far-end carrier and the caller has not restricted the sending of their number;

(c) Your PBX or telephony equipment connected to the SIP Trunking Service must present a valid CLI as follows:

(i) The format of the CLI follows the E.164 standard;

(ii) A valid CLI that is attributed/owned by you in the UK 01, 02, 03 or 08 number range must be presented as a P-Asserted-Identity following RFC 3325, or where ISDN <> SIP Converters are used, the ISDN Network Number will be used;

(d) Traffic to the European Economic Area may be blocked by us, or our underlying carriers, where a P-Asserted-Identity is not a valid number from a European Economic Area country; and

(e) This document describes how we will support sites in the UK. For sites outside the UK, we will provide a separate SoW to describe the solution and our services.

7. Definitions and Acronyms

7.1 Definitions

In this document “we” or “us” refers to the Supplier, and “you” refers to the Client.

The terms listed have the following meanings:

Term	Meaning
Active-Standby	Method of routing calls to multiple endpoints where one endpoint is prioritised over the other.
Balun	Device converting a balanced line to unbalanced and vice versa.
End-Point	SIP enabled network device such as a phone system, gateway or session border controller.
IP-Enabled	A device that supports internet protocol.
ISDN ↔ SIP Converter	A device that converts ISDN to SIP.
P-Asserted-Identity	Privacy Asserted Identity - Ref: IETF RFC 3325.
PBX/Gateway	Private branch exchange (phone system) or system that supports the SIP protocol.
PoP	Point of presence where network infrastructure is located.

Term	Meaning
RFC2833	IETF standard for dual tone multi-frequency digits.
RJ45 G.703 Interfaces	Specific interface type commonly used with phone systems.
Round Robin	Method of routing calls to multiple endpoints and calls are evenly distributed.
Session Border Controllers	SIP based back to back user agent used to terminate SIP trunks.
SIP Channel	Channel providing the capability for a single two-way communication over a SIP trunk.
SIP Trunk	Logical trunk enabling a SIP end point or phone system to send and receive calls over an IP network.
Statement of Work	A document which defines project-specific or client-specific activities, deliverables and/or timelines for providing services to that client.

7.2 Acronyms

Acronym	Meaning
CLI	Calling Line Identity
CLIP	Calling Line Identity Presentation
CLIR	Calling Line Identity Restriction
CPE	Customer-Provided Equipment
DTMF	Dual Tone Multi-Frequency
ISDN	Integrated Services Digital Network
L3VPN	Layer 3 Virtual Private Networks
LAN	Local Area Network
NGN	Next Generation Network
OFCOM	Office of Communications
Off-Net SBCs	Off-Net Session Border Controllers
PBX	Private Branch Exchange
PSTN	Public Switched Telephony Network
PoP	Point of Presence
Q.931	ITU standard ISDN Connection Control Signalling Protocol
QoS	Quality of Service
SIP	Session Initiation Protocol
SoW	Statement of Work

8. Operational Targets

Product		P1	P2	P3	P4
SIP Trunking Standard Type 3	Response	30 mins	30 mins	1 hour	Next Business Day
	Fix	6 hours	12 hours	24 hours	3 Business Days
	Hours of Support	24x7	09:00 - 17:30, Business Days	09:00 - 17:30, Business Days	09:00 - 17:30, Business Days
	Description	• Complete loss of service of the SIP Trunk; • All incoming and outgoing calls failing.	• Issues with specific destinations; • Performance issues or failures on 50% of calls.	• Service Available with non-service affecting issue.	• Service Requests; • Non-Service affecting work and reports; • A required change or configuration alteration to a service.
SIP Trunking Resilient Type 1	Response	30 mins	30 mins	1 hour	Next business day
	Fix	4 hours	6 hours	8 hours	3 business days
	Hours of Support	24x7	09:00 - 17:30, Business Days	09:00 - 17:30, Business Days	09:00 - 17:30, Business Days
	Description	• Complete loss of service of the SIP Trunk; • All incoming and outgoing calls failing.	• Issue with specific destinations; • Loss of service of either SIP trunk; • Performance issues or failures on 50% of calls.	• Service Available with non-service affecting issue.	• Service Requests; • Non-Service affecting work and reports; • A required change or configuration alteration to a service.

The target SIP Trunk implementation time is five (5) Business Days, exclusive of any time reasonably beyond the control of the Supplier. Please note that this target does not cover L3VPN (Managed WAN) implementation.

The target delivery time for Porting is highly dependent on the complexity of the migration and on third parties' delivery time. The implementation target is typically between ten (10) and forty (40) Business Days from Order Acceptance.

9. Service Levels and Service Credits

The Service will be delivered at the target level set out below for the percentage level described, against which service credits will be paid (as set out in the Billing Guide).

Service	Measurement	Period	Target	Service Level	Service Credit
SIP Trunking (Standard SIP Trunks)	**Availability** - the SIP Trunking service is available from either SIP Trunk. The Availability of the Service is maintained whilst there is no active P1 Incident.	Calendar Month	100%	99.35% - 98.85% Less than 98.85%	5% of monthly value of Recurring Fees 10% of monthly value of Recurring Fees
SIP Trunking (Resilient SIP Trunks)				99.89% - 99.35% 99.35% - 98.85% Less than 98.85%	5% of monthly value of Recurring Fees 10% of monthly value of Recurring Fees 15% of monthly value of Recurring Fees