SIP Trunk Service Description

Version	Date	Description
2.2	28th March 2013	Addition of international number presentation to CLI flexibility and asymmetric resilient services
2.3	1st July 2013	Addition of the CallGuard service.
2.4	1st August 2013	Addition of Predictive or Auto dialler restrictions
2.5	25th September 2013	Addition of CLI presentation options
2.6	1st October 2013	Dialler policy clarification
2.7	10th December 2013	Clarification on CLI presentation
2.8	1st August 2014	Branding and name changes
2.9	1st November 2014	Addition of the Active – Standby Resilience+ option

Contents

1.0 Introduction and Purpose

The purpose of this document is to detail our full service offering for the supply of our SIP trunking service – also known as IP Direct Connect (IPDC).

1.1 SIP Trunk Overview

SIP trunks provide VoIP connectivity for certified telephone systems (PBXs) allowing inbound and outbound telephony when connected to a customer's chosen IP PBX and a suitable IP connection. SIP trunks help transform an end users site from analogue technology to an IP based one, where the end users IP PBX system will be connected over Gamma Business Communication's IP based network supplied by our parent company, Gamma.

Our SIP trunking service uses SIP (Session Initiation Protocol) as the signalling method and offers both Public and Private Access to the service depending on the specific customer needs.

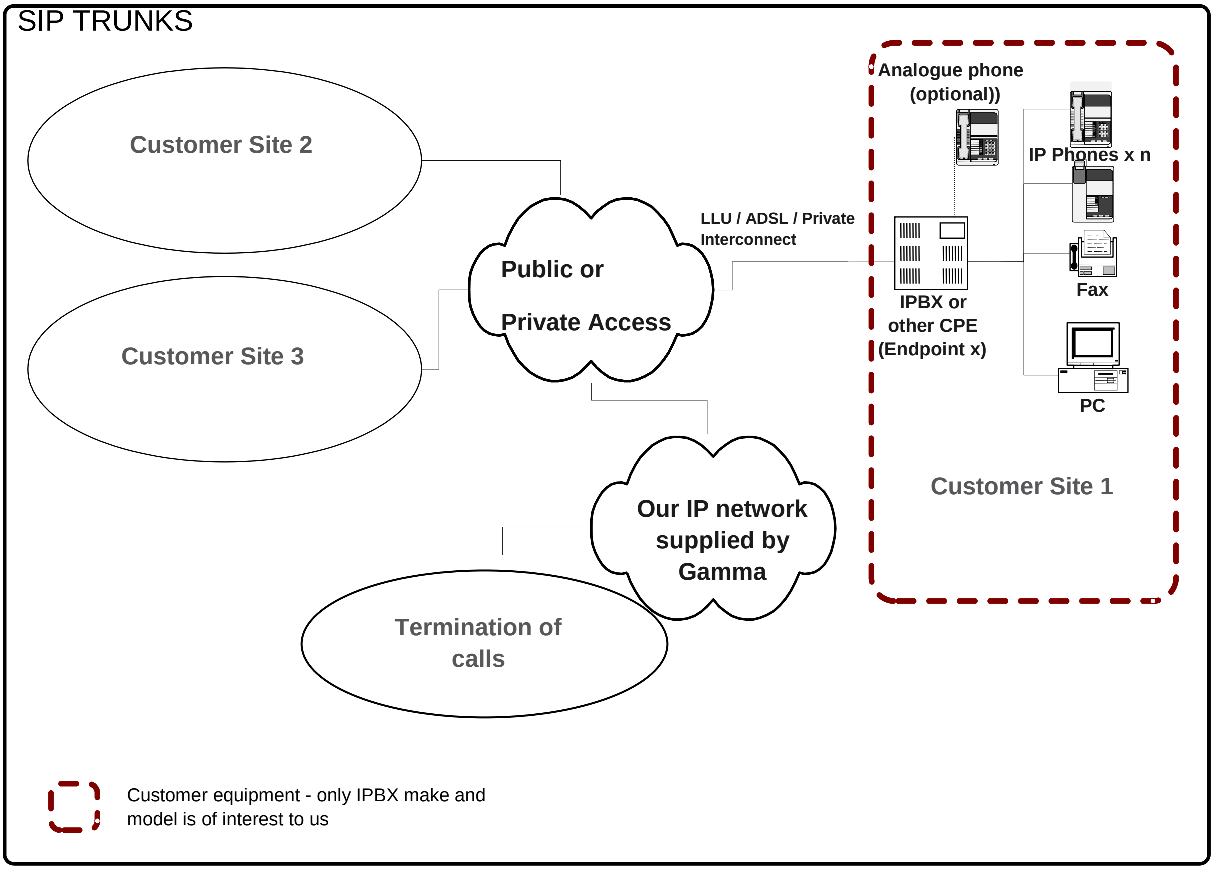
This will allow the end user to take advantage of benefits of a Voice over IP (VoIP) solution which includes:

- Lower call costs when compared to traditional connections
- Lower connection and rental costs when compared to traditional connections
- Free site to site IP voice calls
- Faster provisioning timescales than traditional connections
- Scalable to fit your needs without providing further physical connections
- Unlocks your PBX functionality over multiple sites
- Offers full emergency services support
- Provides disaster recovery support
- Complements a converged voice and data network with simplified architecture
- Access to all UK number ranges and comprehensive number porting agreements

2.0 Service Components

As part of the service offering, the end user will require the following service components and although these products can be sourced externally, by purchasing all components from Gamma Business Communications, you will gain the following benefits:

- Products and services designed with Voice Over IP connections in mind
- One provider for the complete solution simplifying the support model and your key contacts
- Ability for us to monitor all connections
- One bill for all aspects of the service



The service is supplied on a per site basis and the amount of channels requested per site will be provided but cannot be shared over multiple sites. You can however order any number of channels per site and can specify the connections that you require and as your system split requires.

Free site to site calls are only available between multiple IP connections with Gamma Business Communications and set up against the same customer account.

Access is provided to each site through a choice of xDSL, Ethernet or private interconnects as required and this can be provided by us or another party as you require.

For connection via the public internet, communication with our IP network must be made using a SIP proxy server such as a SIP trunk card within a PBX, and secured by either a fixed IP address or presentation from the PBX of a SIP username and password.

Where connections are made using our connectivity products, these service requirements are included as part of the provisioning service.

3.0 Access to the SIP Trunk Service

The SIP trunk service can be accessed by a number of connectivity methods, all of which are available from us direct. Alternatively it can be sourced by the customer along with any underlying analogue line requirements to support these services.

To ensure a stable and reliable service we would strongly advise our customers to select a solution from us which has been carefully designed with routing of voice calls in mind rather than the standard data connections available on the market place. If a customer should choose not to use one of our access products, we would ask you to ensure the service is of a business quality and capable of routing the required number of telephone calls subject to your PBX requirements

As a guide, 1 consecutive telephone call can take either 40 KBits or 100 Kbits depending on the quality of compression chosen. So a standard xDSL connection with an upstream bandwidth of 448 Kbits per second would only be able to support 4 telephone calls at the higher compression standard.

Great care should be taken when selecting your broadband provider to ensure the connection provided is of a stable speed and capable of meeting your requirements for the SIP trunk service.

Where a connectivity solution is chosen outside of our product range, support will only be available for the SIP trunk service and will not cover the resolution of faults associated to the connectivity solution and underlying service provided outside of our control.

The common connectivity types available on the SIP trunk service are:

- Gamma provided Access
- Public Internet – your choice of connectivity supplier
- Public Internet bypass - connecting directly to an ISP or access provider who routes traffic to our IP network
- Private Interconnect - connecting your equipment directly to our IP network at one of the “Meet Me” points in the UK

3.1 Gamma Provided Access

We can supply a number of connectivity offerings all dependant on the size of site being connected. These offerings have been built with Voice over IP connection in mind and focus on the key areas of latency, jitter and packet loss to ensure a stable and resilient service.

3.2 Public Internet

If you choose to connect via a 3rd party supplied public internet service, the service boundary for internet connected sites is the IP address of our IP network, specifically our equipment known as Session Border Controllers (SBCs) which control the establishment of SIP trunk connections with end user sites.

It is therefore your 3rd party supplier’s responsibility to ensure the traffic routed is transported through the public internet to the customer’s site. In this case, a static public IP address assigned by the 3rd party supplier will be configured as an endpoint.

3.3 Public Internet Bypass

As with Public Internet connections, it is the responsibility of your 3rd party supplier to ensure the traffic is transported to the customer's site successfully and our service boundary will be the IP address within our P network.

The bypass aspect of this service means that the 3rd party supplier you are working with already has a single hop agreement with our IP network. This peering arrangement means the traffic is not routed through the public internet but instead takes a single hop from your 3rd party supplier's network to ours reducing the aspects of a poor service such as packet loss or delay in data being sent.

3.4 Private Interconnect

We are currently able to provide private interconnection at the following peering points:

- Telicity 2, 8/9 Harbour Exchange Square
- Telehouse East, E14 2AA
- Redbus, 6/7 Harbour Exchange Square
- Telehouse North, E14 2AA
- Eplison Global Hubs, London
- City Lifeline, 80 Clifton St, London,
- Telecity Powergate, London
- Kilburn House at Telecity, Manchester

Connection types include copper, single or multi-mode fibre and bandwidths ranging from 10Mbps to 1Gbps. Interconnect port types include Layer 3 BGP, Layer 2 VLAN and Layer 2 802.1q VLAN. For improved connectivity resilience, you may like to consider diverse connections as opposed to single connections.

We can assist in providing access to these sites however in all cases you are responsible for this connection and the costs of installation/delivery of the cables to our racks. From a support point of view this then becomes the service boundary.4.0 Service Specification

The SIP trunk product is a connectivity product to support the routing of incoming and outgoing calls from your IP PBX. As such all end user features will be subject to the specification of the IP PBX the customer has purchased and installed within their organisation and you should refer to this documentation for a full feature list.

As standard the SIP trunk service supports the following features:

- Calling Line Identification Presentation (CLIP)
- Calling Line Identification Restriction (CLIR)
- Call park, transfer and conferencing
- Standard call barring packages applied to your IP PBX termination (endpoint)
- Standard geographic and non-geographic numbers
- In band DTMF (RFC 2833)

For fax services we would strongly advise looking to move these numbers to our Fax to Email service to take advantage of the savings and efficiencies enabled by this product.

4.1 CLI Presentation

Our network provides a presentation number facility which supports the ability to present a geographic number in the UK number format. This will be supported as follows:

- One of our geographic numbers is allocated to your connection at order creation
- One of our geographic numbers is allocated to your connection as a subsequent change to the service
- A geographic number is ported to us from another Carrier's network

If the number to be presented meets the above criteria, it can be presented to the called party as the number dialling. If the number to be presented does not meet the above criteria, the number presented will be the default number for this circuit which was allocated at point of order as being the first number of the range requested.

For any calls made to the Emergency Services, we will automatically ensure the default number of the circuit is provided against which the 999 details of the connection would be registered.

4.1.1 Flexible CLI Presentation

As an optional service, customers can choose to present a number that is not registered or associated with their SIP trunk connection or endpoint. **Please note this service is only available for SIP trunk connections that authenticate via fixed IP addresses. SIP authentication is not supported for this option.**

The service supports the presentation of the following number formats:

- National Significant (1NNNNNNNNN, 2NNNNNNNNN, 3NNNNNNNNN, 7NNNNNNNNN, 8NNNNNNNNN) for UK numbers.
- National Significant with leading zero (01NNNNNNNNN, 02NNNNNNNNN, 03NNNNNNNNN, 07NNNNNNNNN, 08NNNNNNNNN) for UK numbers.
- E.164 (441NNNNNNNNN, 442NNNNNNNNN, 443NNNNNNNNN, 447NNNNNNNNN, 448NNNNNNNNN) for UK numbers.
- SIP E.164 - with leading plus - (+441NNNNNNNNN, +442NNNNNNNNN, +443NNNNNNNNN, +447NNNNNNNNN, +448NNNNNNNNN) for UK numbers.
- UK International (00441NNNNNNNNN, 00442NNNNNNNNN, 00443NNNNNNNNN, 00447NNNNNNNNN, 00448NNNNNNNNN) for UK numbers.
- SIP E.164 - with leading plus - (+CCNNNNNNNNNN) for non-UK numbers.
- UK International (00CCNNNNNNNNNN) for non-UK numbers.

Please note:

- For international numbers, successful presentation cannot be guaranteed across international networks as the transmission is entirely dependent on our connected carrier partners. For example, calls with international CLI Flexibility destined for UK landline termination will generally show the default network number and not the International number. As a result, we will not accept as faults, any situations where International numbers fail to be presented as the intended CLI.
- Calls to mobile carriers cannot guarantee consistent presentation of the intended CLI as successful presentation is entirely dependent on the carrier's use of these numbers and specific call flow. For instance, missed calls and voicemail notifications will often use the underlying network number rather than the intended CLI as the presented number.
- For the avoidance of doubt, we will not support the presentation of any other number types such as premium 09 numbers. If a number is found to connect to a revenue sharing number that generates

excessive or unexpected call charges, we reserve the right to suspend and/or withdraw the CLI Presentation service.

- Presentation of Mobile A-Number CLI types (7NNNNNNNNN, 07NNNNNNNNN, 447NNNNNNNNN, +447NNNNNNNNN and 00447NNNNNNNNN) excludes personal numbers.
- For calls made to the emergency, non-emergency and Operator Services (100, 101,111,112,116, 118,123,1800 and 999), only Gamma A-Number CLIs are accepted and must be in the National Significant (with or without a leading zero) or the SIP E.164 - with leading plus – format. Any other A-Number CLI or A-Number CLI format will be overwritten by the default CLI, which is the first number in the Gamma allocated DDI range.
- All A-Number CLIs will be "normalised" within the Gamma network to the SIP E.164 format for external presentation but Gamma are not responsible for subsequent carriers changing the presented A-Number CLI or its format.

To enable this service please ensure you discuss this with the Customer Development Manager and they will be able to supply the required documentation required for this option.

Once confirmed the customer will need to configure their PBX/Gateway to pass Presentation CLI in From Field and Gamma Provided E164 in the “P-Asserted-Identity” field (as defined in RFC3325) in the header of the SIP packets sent, and such field must be populated with a CLI provided by us in E.164 format associated to our SIP trunk end point.

It is expected that the end user will ensure test calls are made and the PA-ID is configured correctly. This should take place within the first 20 days of service otherwise we reserve the right to withdraw this service.

4.2 Call Park, Transfer and Conferencing

The service provides the features listed below in the majority of cases but are not guaranteed on every platform connected to SIP trunk because of vendor interoperability issues:

- Call parking
- Call transfer
- Conferencing

These features are supported via the SIP re-invite mechanism.

4.3 Call Barring

By default, calls to international numbers will be barred. Customers are able to modify their profiles to allow access to international numbers. Similarly, customers can modify their barring profile to allow or restrict all premium rate 09 numbers.

4.4 Call Divert on Failure

An optional service of call divert on failure of the SIP trunk connection is available which will divert calls to another number on the failure of the customer’s connection. The failure may or may not be related to our infrastructure and where a 3rd party supplier is being used, the end user is responsible for managing this through to resolution.

In order to implement this service, the relevant failover numbers should be included on the order form. There is a choice between diverting all DDIs to a single number and diverting each DDI to a different number. Please note that international numbers are not allowed as divert destinations.

It should be noted that it is the customer's responsibility to give us a number that will be suitable to use in case of service failure which may occur at any time.

For calls diverted during this time, a call charge and additional service charge will be levied to you to cover this service offering.

4.5 SIP Trunk CallGuard

CallGuard is an optional service that can be applied to all SIP trunk end points to protect the customer against unusual call usage. The service enables customer call usage to be monitored 24/7 and for suspected fraudulent or unusual call activity to be identified. If detected, we will automatically apply a bar to the relevant end point which can be removed upon customer request.

CallGuard is based upon an approximate daily limit of outbound calls made across a particular end point. This is a guide only and the automatic bar will be applied once unusual activity is detected rather than when an actual set limit is reached.

Please note that CallGuard is applied to outbound traffic passing on the Gamma network only and if calls are routed over any other network for any reason, these calls will not be protected and the customer will remain liable for these calls. Calls to the emergency services 999, 112 and 18000 will be unaffected and if an end point is barred then inbound calls are not affected.

Please refer to the CallGuard Service Description for more information.

4.6 Call Admission Control

The maximum call limit of an endpoint will define its capacity for routing calls in the network and this is known as call admission control (CAC). Customer SIP trunk solutions will allow for a certain number of concurrent calls on their endpoint. As each customer endpoint will have 2 ports, one for outgoing and one for incoming calls, the CAC limit will be allocated to both ports to allow maximum flexibility. Thus we will support any combination of incoming or outgoing calls provided the total number of calls does not exceed the total channel allocation i.e. CAC limit.

- Maximum Total Calls – specifies the overall number of calls the endpoint will support, both inbound (ingress) and outbound (egress)
- Maximum Ingress Calls – specifies the maximum calls that may be placed from that endpoint to our network
- Maximum Egress Calls – specifies the maximum number of calls that may be placed to that endpoint by our network.

For example, if the channel limits is 100 concurrent calls and there are 70 ingress calls, the maximum no of egress calls allowed will be 30.

4.7 Maximum Calls per Second

For security reasons, we will also set limits for the maximum calls per second. If this constraint is reached, then we will log and reject calls with a SIP response 486.

The limit is set at 2 calls per second for SIP trunks with up to 30 channels and 5 calls per second for SIP trunks built with a resilient design (i.e. active/standby or load share).

4.8 Calling Plans and Number Formats

Calling plans for both incoming and outgoing calls are allocated to each customer's connection and it is a requirement of the SIP trunk product that the calling party number (A-number) issued from the IP PBX is validated to confirm the format and ensure the number is owned by us so that we can ensure accurate 999 information is passed through to the Emergency Services.

4.8.1 Presentation CLI (A-Numbers)

A-numbers should be presented by the customer premises equipment (CPE) in the UK national format without a leading zero. The A-number is checked against a database on our network of geographic numbers that are allocated to the SIP trunk site connection and if it is not in our range, it will be overwritten with a default number (CLI) which is the first number in the allocated DDI range.

If the CPE connected to our network presents a geographic number, we will pass these details as the A-Number CLI into the PSTN or mobile networks.

This outbound presentation will be supported by default if the number presented is as follows:

- A-numbers presented by the Customer premises Equipment (CPE) in the SIP FROM or P-Asserted-ID fields as E.164 format
- It is one of our Geographic Numbers that is allocated to the endpoint at order creation
- is one of our Geographic Numbers that is allocated to the endpoint at a later date
- A Geographic number that is ported from another Carrier to our network

If the presented number meets these criteria, the A-Number CLI will be presented to the called party and that CLI information will be presented on your monthly management information.

4.8.2 Network CLI (A-Number)

Every endpoint must have at least one CLI from our allocated range i.e. a non-ported in number. This default number is known as the Network CLI and will be presented in the case of emergency calls and other call scenarios where the presented number is invalid. Physical address information must be associated with a network CLI and it is the customer responsibility to ensure this address information remains current.

We support both Network and Presentation CLIs. For calls from the customer to us where the call terminates on the PSTN, the SIP FROM field is mapped to the presentation CLI and the SIP P-Asserted-ID or Remote-Party-ID is mapped to the network CLI.

4.8.3 B-Numbers

B-numbers are the called party number from the customer’s premises and should be sent to us in the following format:

- NSN (national significant number), CC (country code)
- UK National Significant with a leading 0 (0&NSN)
- UK dialled E.164 with leading ‘+’ (+44&NSN)
- UK dialled E.164 without leading ‘+’ (44&NSN)
- UK dialled International (0044&NSN)
- UK International (00&CC&NSN)
- UK International E.164 with leading + (+CC&NSN)

Service and emergency calls (short codes) must not be prefixed with a leading 0, 00, + or CC (country code).

As a default configuration, B-numbers will be presented to the customer as UK National Significant with a leading 0, (0+NSN); however this is flexible and customers can request that this be replaced with UK National Significant without a leading 0 (NSN) or E.164 with leading '+' (+44&NSN), a prefix may also be inserted.

4.9 Customer Premises Equipment Hardware

The supply, provisioning and support of the hardware at the customer's site are the customer's responsibility and should be sourced from a well-known manufacturer. Our solution can support all devices that conform to the SIP v1 and v2 RFCs.

To ensure compatibility of equipment and ease of installation, we are continually undertaking conformance testing with equipment vendors and if you require a list or further details specific to a manufacturer please contact your Customer Development Manager.

Please note if connection is required for a device that has not previously been connected to our network, we will first work with you requesting connection to connect the equipment to a lab machine. This testing will ensure that the device is fully compatible with our network prior to an implementation being undertaken for any end user customer

4.10 Network Security

The SIP trunk service offers connectivity to end user devices/CPE via IP Authentication and as such, the service will only accept traffic from genuine SIP trunking endpoints that have been registered on the service.

It is the customer's responsibility to ensure that calls emanating from their endpoint are legitimate and that all practical steps have been taken to avoid fraudulent activity. This would include secure access to their network by means of a Firewall or a Session Border Controller.

4.11 Numbering

For every SIP trunk service connected at least one geographic number must be ordered to enable us to test the service prior to any additional number porting taking place.

We are able to supply 99% of the UK's area codes however it should be noted that in specific locations such as the Channel Islands, we are not able to supply numbers due to technical reasons when connecting to the local service provider.

All connection and numbers provided will be included in the pricing provided to you.

4.12 Call Termination

All common call termination types are available through this service, i.e. local, national, mobile, NGN, premium, international 118 etc.

4.13 Emergency Services

Our SIP trunk service is a VoIP service as defined by Ofcom and can be used to support Emergency Services calls. Once the service is fully operational, 999/112 public emergency call services can be accessed and will be routed to the national emergency call handling agents. The number presented will always be the site number indicated as a VoIP service type from Gamma, so that the emergency services operator will check the address details with the caller.

However, as a VoIP service, SIP trunk does NOT operate identically to fixed line 999 or 112 public emergency call services and connection to such services may not be possible, in the following circumstances:

- During a service outage where the customer loses connectivity, for example owing to a power outage or the failure of routing equipment
- If the customer's account has been suspended by us for non-payment

In such circumstances the customer should use the analogue line which provides the connectivity and retain an analogue handset in case of emergency.

Also, it may or may not be possible for the emergency personnel to identify the actual end customer's location and phone number when they dial 999/112. They will need to state their physical location and contact phone number, as well as the nature of their emergency, promptly and clearly, as emergency personnel may NOT have this information due to the VoIP nature of this service.

The customer should ensure that they are aware of the above service limitations related to the Emergency Services support in line with the Ofcom Code of Practice related to VoIP services.

4.14 Access to Operator Services

Fully supported by our SIP trunk product.

4.15 Number Portability

Fully supported by our SIP trunk product.

4.16 Service Mobility

Not applicable to the SIP trunk service as it is a per site service.

4.17 Fax and DTMF

The following fax methods are supported

- In-band fax and modem transport using G.711 A-law codec.
- Renegotiation to T.38 (subject to interoperability testing)

Note: In-band with codec up-speed to G.711 is not supported.

The following methods will be supported to transport DTMF tones:

- RFC2833 is the preferred method for the transport of DTMF tones. Support of RFC 2833 is dependent on successful codec negotiation and requires the payload type 101 to be assigned. RFC2833 will be used with both G.711.and G.729 codecs.
- In band (G.711 codec is preferred. With G.729 detection of the tones at the far end will not always be guaranteed)

4.18 Signalling and encoding

Our SIP trunking provides SIP signalling as a method for Communication Providers to inter-connect with our VoIP network, supporting VoIP to VoIP calling as well as calls to/from the PSTN. In addition, our SIP trunking supports the transport of SIP signalling messages using UDP. SIP messages sent using TCP, TLS, SCTP and IPSEC are not supported at present.

SIP (V1 and V2) are supported between our Session Border Controller (SBC) and the customer's CPE. Please note that this service does not support SIP-I, SIP-D and SIP-T.

Voice encoding can be G.711, or G.729, with both μ-law and A-law supported. Currently the most common sample period is 20ms, although it is likely that 10ms, which introduces less latency, will become the standard in the future. Both sample periods are supported by our network. It is important to note that we can only control the sample period on the outbound direction of calls made by our customers.

4.18.1 Bandwidth Requirements

The table below gives an estimate of the bandwidth requirements for VoIP calls using G.711 and G.729 with corresponding sampling periods. Note: this represents a best case, un-contended scenario.

The maximum number of concurrent channels = the available bandwidth /total bandwidth, so if for example a customer has a 512 kbps upload line-speed, assuming no contention, using G.729 with a sample period of 20 ms there will be 512/40 ≈ 10 usable concurrent channels available.

Codec	Sample Period	Approx. bandwidth required
G.711	10ms	130 Kbps
	20ms	100 Kbps
G.729	10ms	70 Kbps
	20ms	40 Kbps

Generally G.729 is the preferred VoIP codec (the algorithm that encodes and decodes analogue voice to and from digital) when access is via ADSL, owing to its efficient use of bandwidth whilst still providing good audio quality.

4.19 Dialler Traffic Policy

In order to remove any ambiguity regarding our Dialler Policy, we do not accept any dialler traffic onto our SIP trunking service and will take efforts to remove such traffic if/when detected. This position is based on the operational need to safeguard our network and ensure that optimum service levels remain available to all of our customers across our SIP Trunking services. In addition, the following Ofcom guidelines http://stakeholders.ofcom.org.uk/consultations/silent-calls/statement/ regarding nuisance, silent and abandoned calls would be considered by us to be a regulatory requirement to be adhered to by any signatory to our contracts.

For clarity, Gamma considers acceptable traffic to have an Average Call Hold Time (ACHT) of greater than 50 seconds and an Answer Seize Ratio (ASR) of greater than 60%. We monitor all SIP traffic and will respond to endpoints that exhibit 'dialler like' call patterns and we reserve the right to suspend the endpoint if such activity is detected.

4.20 Service Limitations

G711 pass-through is supported for the provision of fax services, however we strongly recommend that customers convert their fax machines to our Fax to Email service. If the customer chooses to route fax calls through the SIP trunk service, it should be noted that this is not a fully supported service.

Our service does not allow users to withhold their number on a per extension basis i.e. similar to the 141 service.

4.2.1 Resilience+ Limitations

Resilience+ is limited to dual endpoint implementations and if there are more than two sites, please involve the IPT Pre-sales team.

Each implementation will need to have the correct number of channels to handle automatic failover, for example, in a 100 channel deployment (say 70/30) you would need to actually have 200 channels (split 100/100) to handle the full 100 channel capacity requirements in the event of an emergency. Otherwise the customer might see some impact on voice quality when the Call Admission Control (CAC) limit is reached.

5.0 Designing and Delivering SIP Trunking

The successful order and installation of an SIP trunk solution relies on a detailed understanding of the customer's requirements in terms of access and resilience and also an understanding of the design options available within the SIP trunk service itself.

Our SIP trunk service can accommodate a number of more complex builds that include both dual and multiple endpoints, offering higher degrees of resilience.

5.1 Design Options

We offer a number of design options within our SIP trunk service;

- **Single Site** – A single site working off a single Session Border Controller (SBC) HA Cluster
- **Multiple Sites** - (two or more), working off two SBC pairs
- **Multiple Sites -** (two or more), working off multiple SBC pairs

Please note that for multiple SBC and to provide geographic diversity, one SBC pair is usually located in London and the other in Manchester.

Multiple sites also allow for a load balancing option:

- **100:0** – Active/Standby load balancing. All calls will be presented to the resilient site. In the event of failure, either the private interconnect or the PBX, calls will be routed to the secondary site
- **50:50** – Calls will be evenly distributed between sites. In the case of more than 2 sites, calls can be distributed 33.3%:33.3%:33.3% or 25%:25%:25%:25% etc.

All multiple site and resilient designs (i.e. other than a single site working off a single SBC HA Cluster) are created with the assistance of our Pre-Sales Technical team.

There are two resilient design templates, Active/Standby and Loadshare.

5.2 Active/Standby

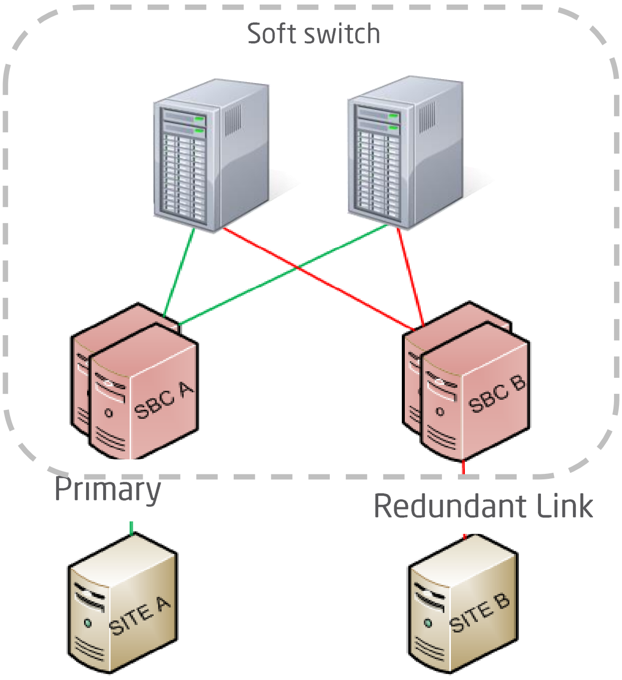
Our soft switch nodes are configured to route to SBC A as a first choice with second choice to SBC B if site A is un-available.

In normal operation, the primary link contains all the active calls; the redundant link can either have the same number of channels or a different number of channels allocated. Under failure conditions, all calls are presented to this redundant link.

- Resilient Active – Standby symmetric option – all DDIs are on the active endpoint and if that fails then they will failover to the active standby. The same number of channels is available on both endpoints.
- Resilient Active – Standby asymmetric option -– all DDIs are on the active endpoint and if that fails then they will failover to the active standby. A different number of channels are available on the two endpoints.

- Resilient Active – Standby Resilience+ option -– the customer is able to select which DDIs are allocated to a particular end point and a different number of channels are available on the two endpoints.

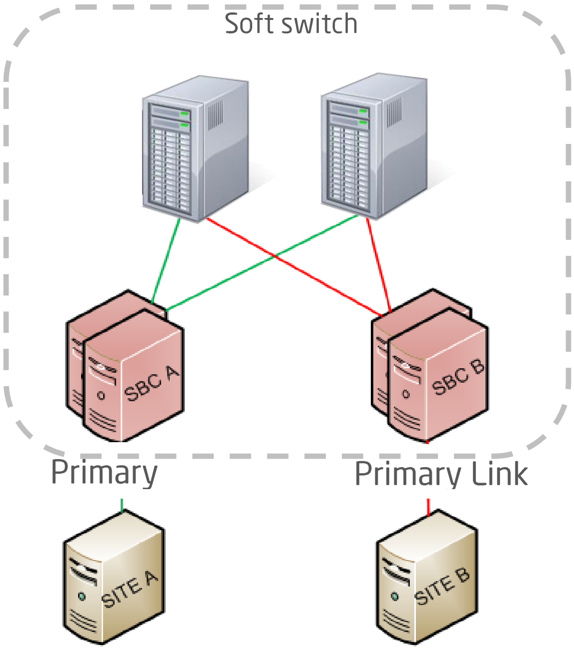
The primary link is continually monitored with traffic returning to this link when its availability is resumed.



5.3 Loadshare

Our soft switch nodes are configured to load share traffic across both SBC nodes with option to overflow/re-route to other either endpoint under channel exhaustion or endpoint failure conditions.

Note: If an SBC or customer connectivity fails then channel allocation will equal 50% of total.



6.0 Provisioning Process

In order to provide this service to our customers, we would establish a customer account for you against which the orders can be placed. Once this is established, please speak with your Customer Development Manager who will be able to discuss your requirements, the type of IP PBX being used and then progress an order accordingly.

As part of this service provision, you will need to also consider the following and add to your order as required:

- A single analogue line if access is dependent on this connection
- A business ready xDSL or other connectivity option.
- Identification of any numbers which are to be ported to this service
- Area code of numbers to be provided if taking new numbers with this connection
- Type of IP PBX which is being added to this connection

The lead time to provide an order without additional orders for porting or access being required is 5 working days.

6.1 Configuration of an Endpoint (IP PBX)

The configuration of an endpoint, traditionally an IP PBX, would be the responsibility of the customer's IP PBX supplier from whom this equipment was purchased.

In order to ensure they configure this correctly, please refer to Annex B for connectivity settings.

6.2 Testing

Once an SIP trunk service has been connected, a series of tests will be carried out subject to our standard test plan to confirm the following where applicable:

- SIP trunk calls to fixed line and mobile pre-validated destinations
- Calls to the SIP trunk from fixed line and mobile pre validated destinations
- Clear of calls in your IP PBX system once made
- SIP trunk calls with no answer to ensure ringing stops
- Invalid number calls to ensure correct tone and messaging
- Incoming calls receiving busy tone
- Incorrect numbers being dialled
- Number Presentation, both incoming and outgoing
- CLI Restriction, both incoming and outgoing
- Fax call from PSTN line if requested
- Fax call through SIP trunk if requested
- Call barring on outgoing

7.0 Support

Support is provided to you by our dedicated support teams and your Customer Development Manager as required for fault and provisioning queries. These teams will be available to Monday – Friday between the hours of 0900 and 1800 and can be contacted on 0333 014 0333.

7.1 Service Boundaries

As discussed earlier in this document depending on how the SIP trunk service is connected, the customer will need to ensure any queries or faults are pointed to the right supplier to progress.

In a scenario where we do not supply the underlying connectivity access, this should be checked using the suppliers fault guides and if required by attaching a computer and checking for internet connection.

8.0 Invoicing and Billing

The SIP trunk service is billed at a fixed rate per month.

Description	**Billing Description**
Connection	SIP trunk per site connection
Rental	SIP trunk per site rental
Calls	SIP trunk per site calls

Contract terms and conditions for the SIP trunk service are available online or are available from your Customer Development Manager.

8.1 Call Data

As part of the SIP trunk offering, all calls from the service will be billed as per your agreed tariff and be summarised your monthly invoice.

8.2 Invoicing

All chargeable rentals will be added to your monthly invoice, and will be issued and collected in line with the agreed terms. A summary view of both fixed charges and calls made will be provided

8.3 Billing queries

In the event there is a dispute with charges around the SIP trunk service, this should be raised directly to your Customer Development Manager for investigation.

Appendix A – SIP Trunk FAQs

Q1. What bandwidth is required at the customer site?

There are a number of factors that determine the amount of bandwidth required at a customer site i.e. the number of concurrent calls, the codec employed, and the packetisation of the data. For guidance on the maximum number of calls recommended on different access types please contact your Customer Development Manager who can discuss your exact requirements and advise on the best offering.

Q2. Can the access be shared with internet access?

We do not recommend that the broadband connection utilised for SIP trunks is shared with any other application. A dedicated connection for voice will ensure the maximum service quality and number of concurrent voice calls that can be supported.

The maximum number of calls that can be supported from an access connection should be determined with your ISP and is dependent on the throughput that can be supported and contention of the service.

Q3. What codecs and packetisation are supported?

G711 and G729 codecs are supported .Currently the most common packetisation is 20ms though it is likely that 10ms will become the standard in the future. The SIP trunk service supports both.

Q4. Does the service support CLIP and CLIR?

Yes

Q5. Can the number of concurrent calls be changed?

Yes the number of concurrent calls can be changed, however additional charges may be applied.

Q6. What signalling protocols are supported?

SIP v1 and v2.

Q7. How do I register the SIP phone end users?

The on-site customer PBX provides the user registrar. Our SIP trunk solution will provide registrar support for the IP PBX or concentration device on site, not for the individual IP Phones

Q8. I have a dynamic IP address, how do I configure my equipment?

At present we can only accept static IP addresses or alternatively we can use the SIP registrar functionality.

Q9. What customer equipment is supported?

Our SIP trunk solution can support all devices that conform to the SIP v1 and v2 RFCs. To ensure compatibility of equipment and ease of installation, we are continually undertaking conformance testing with equipment vendors. For the devices that have been tested please contact your Customer Development Manager.

Q10. Do you support call parking, call hold and conferencing?

These features are supported for most manufacturers however we cannot guarantee inter-working with all vendors due to differing vendor implementations.

Q11. Is fax supported?

G711 pass-through is supported for the provision of fax services. Please refer to your equipment vendor’s specification for the correct configuration of this facility.

Appendix B – SIP Trunk Connectivity Information

The following information defines the configuration details that relate to our SIP trunk solution. The details should be used in the configuration of the IP PBX or other connectivity device that is planned to be connected.

SIP signalling gateway address: 83.245.6.81

The SIP header requirements in the INVITE packet originated from the CPE should be set as follows:

- SDP payload must be present
- From: header must contain public facing interface and originating CLI
- To: header must contain SIP trunk SIP gateway address and the called number without the leading zero

Numbers are validated against our assigned number ranges for each connection

The following firewall ports must be opened up on the customer premises to allow access:

- TCP and UDP port 5060 egress/ingress to IP Address 83.245.6.81
- UDP all ports above 6000 – 40000 egress/ingress to IP Address 83.245.6.82