Business

# IP VOICE (SIP TRUNKING)

G-Cloud 15: Service Definition Document

Date: January 2026

Issue number: 1

# Contents

**Brand Statement**

Buyers entering into a Call-Off Contract will be contracting with Virgin Media Business Limited (“The Supplier”).

On the 1st August 2025 Virgin Media O2 Business and Daisy Group merged to create a new B2B telecoms powerhouse – O2 Daisy.

O2 Daisy is fully consolidated by Virgin Media O2 with a 70/30 ownership structure with Daisy Group. The corporate brands Virgin Media O2 Business, Virgin Media Business and O2 Business remain.

For contracting purposes, customers of these brands contract with Virgin Media Business Ltd. Depending on the services to be provided, affiliate companies may act as billing and collection agents, such as Telefonica UK.

# 1.0 Overview

“**SIP”** (“Session Initiation Protocol”) is a standard which has been adopted by communications hardware and software vendors because it enables them to standardise their technologies and interoperate with other products and services more efficiently.

“**SIP Trunking”** is an IP based voice solution based on the SIP standard. SIP Trunking enables Buyers to make secure IP based calls, usually originating from an IP phone or desk-based client (softphone) to the PSTN (Public Switched Telephone Network) and vice versa.

SIP Trunking offers a logical migration path from traditional TDM based voice services such as ISDN for Buyers who are not yet ready to move to a Cloud Unified Communications solution and wish to retain a PBX.

SIP Trunking enables Buyers to consolidate their voice and data over a single network connection e.g. a WAN (Wide Area Network) rather than having to deploy dedicated voice circuits. This converged approach typically leads to cost savings.

The Supplier’s SIP Trunking service can be provided over ‘any access’ with multiple access resilience configuration options available. An optional SIP Trunking Portal is available which provides Buyers with web-based access to a configuration and reporting portal.

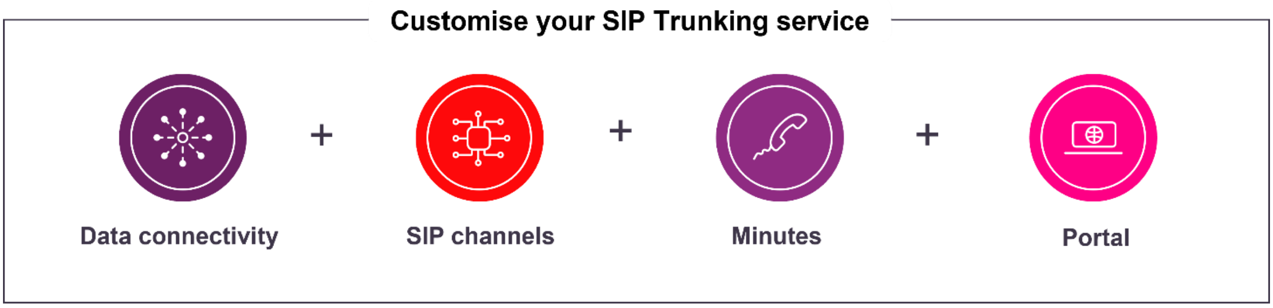
The Supplier can provide new Direct Dial In (DDI) for use with the SIP Trunking service or Buyers can port in number ranges from Other Licensed Operators (OLOs). The Supplier has Porting Agreements with a wide range of OLOs.

The Supplier’s highly resilient SIP Trunking service is provided from 2 x geographically diverse Data Centres where the core network elements, providing SIP Trunking capabilities are fully duplicated. The Supplier’s SIP network has been engineered to enable continuity of service even in the face of a disaster/force majeure event at one of the sites hosting the SIP core network. This geographic resilience means that the Supplier can guarantee 99.999% service availability in the core.

The SIP Trunking Service is highly flexible and can be scaled to a Buyer’s needs, including capacity that can be provided from 15 to 5,000 SIP Channels (or more on discussion with the Supplier).

The Supplier’s SIP Trunking service is monitored and managed from the Supplier’s National Operations Centre, located within the UK.

# 2.0 SIP Trunking Service




## 2.1 Data connectivity

The following data connectivity types can be used with the Supplier's SIP Trunking service:

- Supplier's IPVPN/SD-WAN service
- Supplier's Internet services
  - Smart Internet Access
  - DIA
  - Business Broadband
- 3rd Party Internet

Provision of Supplier data connectivity is separate from the provision of the Supplier's SIP Trunking service. Please refer to the applicable Supplier Service Definition for further details.

### 2.1.1 IPVPN/SD-WAN

The Buyer's site will be connected to the Supplier's SIP Trunking service using the same physical circuit as is used to connect the Buyer's IPVPN/SD-WAN service.

The Buyer can select to use a dedicated circuit for their SIP Trunking service (dedicated access) or layer the SIP Trunking service on top of an existing access circuit (shared access). With shared access the Supplier recommends the Buyer enables QoS for SIP traffic in their network to enable appropriate prioritisation within the Buyer's organisation where applicable.

The Supplier will configure IP routing so that SIP Trunking traffic originating from the Supplier's SIP platform will be forwarded over the Supplier's network towards the Buyer's enterprise Session Border Controllers (eSBCs). Both source and destination will use Public IPv4 addresses.

### 2.1.2 Internet

The Buyer's site(s) will be able to access the Supplier's SIP Trunking service using the Internet connection the Buyer has at their site(s).

The Buyer's eSBCs will register with the Supplier's SBCs using a SIP REGISTER message. This simplifies the provisioning process and removes the need for the Buyer to contact the Supplier should the Buyer's eSBC IP address change.

### 2.1.3 Bandwidth

It is important that the data connectivity being used with the Buyer's SIP Trunking service has been provisioned with sufficient bandwidth to cater for the number of SIP Channels with the chosen Codec.

The bandwidth required per SIP channel for the Codecs available with the SIP Trunking service are detailed in Table 1 below.

| Codec | Bandwidth (Kbs) |
|---|---|
| G.711 (A-Law) | 100 |
| G.711 (U-Law) | 100 |
| G.729 | 40 |
| G.722 | 50 |
| AMR | 45 |
| AMR-WB | 100 |

*Table 1:* ***Codec Bandwidth***

## 2.2 SIP Channels and SIP Trunks

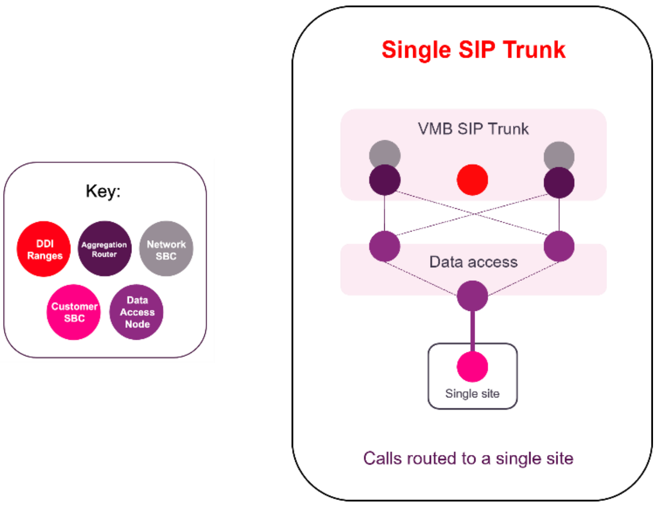
A SIP Channel allows one call to be made or received at any one time. The number of SIP Channels required will be based on the maximum number of simultaneous calls that will be made or received by the Buyer's PBX.

A SIP Trunk provides a logical connection from the Buyer's SBC to the SIP Platform over whichever data connectivity is deployed. SIP Trunks will be configured with the Buyer's required quantity of SIP Channels.

Multiple configuration options exist and can be deployed to suit the Buyer's requirements.

### 2.2.1 Single SIP Trunk

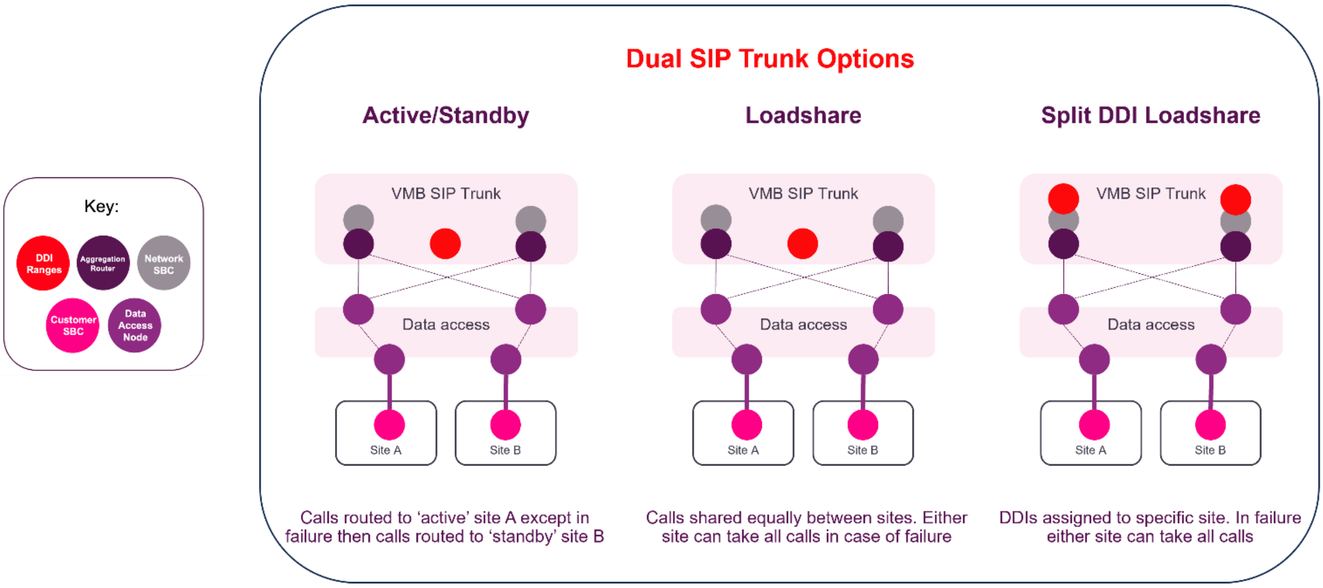
With a single SIP Trunk configuration, a single SIP Trunk is provided to a single site (eSBC).



## 2.2.2 Dual SIP Trunks

With the Supplier's Dual SIP Trunk configurations, a SIP Trunk is provided to two sites (or two eSBCs). Both SIP Trunks are engineered to take the full call load in the event of a failure.

Outbound calls can be sent from either of the SIP Trunks under normal conditions, under failover only the "live" SIP Trunk can be used.




### 2.2.2.1 Active/Standby

With the Active/Standby configuration calls are routed to the 'Active' site e.g. Site A but in the event of a failure will route to the 'Standby' site e.g. Site B.

### 2.2.2.2 Loadshare (Active/Active)

With the Loadshare configuration calls are loadshared across Site A and Site B using a round-robin methodology. If calls are unable to be routed to one of the sites e.g. Site A all calls will be routed to the other site e.g. Site B.

### 2.2.2.3 Split DDI Loadshare

With the Split DDI Loadshare configuration DDIs are assigned for delivery to Site A or Site B e.g. calls to London range DDIs are routed Site A (London) and calls to Manchester range DDIs are routed to Site B (Manchester). If calls are unable to be routed to one of the sites e.g. Site A calls will be routed to the other site e.g. Site B.

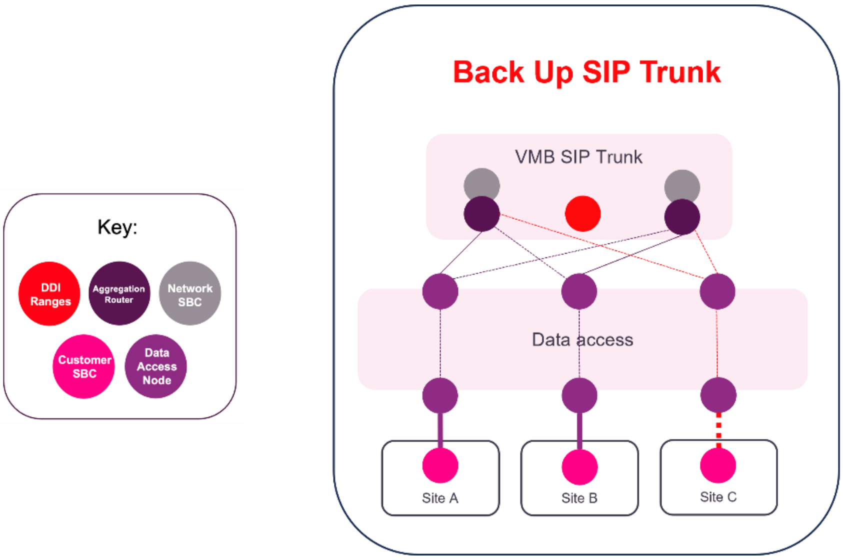
### 2.2.2.4 Supported Data Connectivity Types

Table 2 below details which data connectivity types are supported with our Dual SIP Trunk configurations including Hybrid Access which is a mixed data connectivity environment of IPVPN/SD-WAN (Active) and Internet (Standby).

| Dual SIP Trunk Configuration | Supported Connectivity |
|---|---|
| Active/Standby | IPVPN/SD-WAN<br>Internet<br>Hybrid Access |
| Loadshare (Active/Active) | IPVPN/SD-WAN<br>Internet |
| Split DDI Loadshare | IPVPN/SD-WAN |

*Table 2: **Supported Connectivity for Dual SIP Trunks***

## 2.2.3 Back Up SIP Trunk




A Back Up SIP Trunk can be added for Buyers who want additional resiliency. The Back Up SIP Trunk can only be added to a Dual SIP Trunk with a Loadshare (Active/Active) configuration which is delivered over IPVPN/SD-WAN.

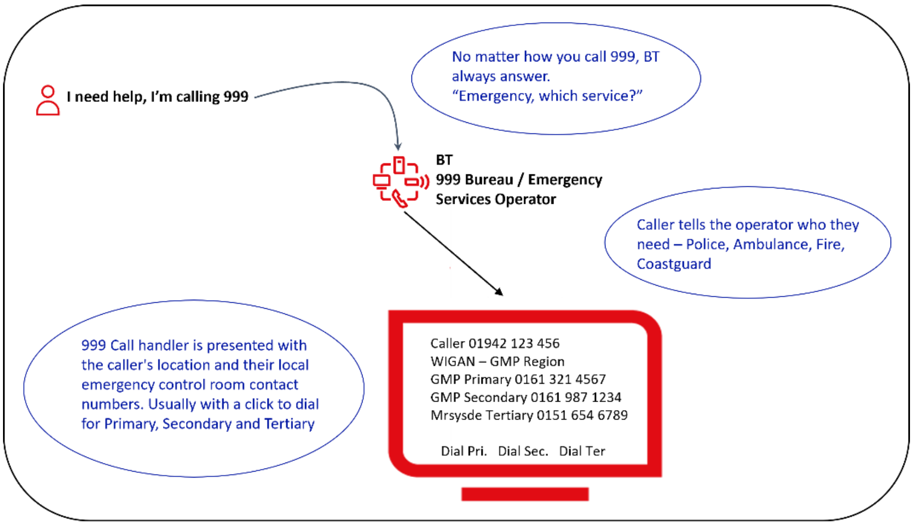
If calls cannot be delivered to Site A or Site B, calls would then automatically failover to the 3rd SIP Trunk/eSBC over an Internet connection.

This would require the Buyer to have a 3rd site and/or eSBC configured with Internet connectivity. In the case of a failure, the Buyer would need to re-route calls to go out of Site C instead of Site A and Site B.

The Back Up SIP Trunk can have the same number of SIP channels as the primary Loadshare SIP Trunks or less and can be added at point of order of the primary Loadshare SIP Trunks or can also be added in life with no downtime experienced on the existing SIP Trunk service.

## 2.2.4 Blue Light

The Supplier has designed a ‘Blue Light’ SIP Trunking service to meet the needs of Buyers who are providing 999 services. The diagram below explains what happens when a 999 call is made.




With the Blue Light SIP Trunking service, the Supplier provides the Buyer with primary and secondary numbers as follows:

**Primary Numbers (On Net SIP)**

- Numbers are from the Supplier’s SIP Trunking service
- Numbers are from a Blue Light block which skips network screening
- Numbers are always blue, never ported and always homed on the SIP Trunking platform
- Calls are prioritised through the network
- Call concurrency is not policed – the Supplier never drop calls even if the Buyer uses more than they buy

**Secondary (Backup) Numbers (Off Net SIP)**

- Numbers are from Off Net SIP (BT Wholesale)
- Numbers are on Premium (1:1 contention) channels

The Blue Light SIP Trunking service is provided over Supplier provided diverse and highly resilient data connectivity which includes:

- Minimum of two Buyer sites
- Minimum of three access circuits
- Redundant NTU & CPE (one per circuit)
- Oversized circuits (demand surge protection)
- Diverse POPs within the “last mile”
- Mixture of access networks (On Net & Off Net)

## 2.2.5 Other Configurations

The Supplier can provide other SIP Trunk configurations in addition to the standard configurations described above e.g. Loadshare across four sites/four eSBCs. The Buyer should contact the Supplier to discuss any non-standard configuration requirements.

# 2.3 SIP Trunking Features

## 2.3.1 Calling Line Identity (CLI)

SIP Trunking supports both Network CLI and Presentation CLI which can be set independently for outbound calls.

The Network CLI is used within carrier networks and is the number used by the Emergency Services if a call is made. It should therefore always be a UK geographic number.

### 2.3.1.1 Calling Line Identity Presentation (CLIP)

A Presentation Number is a number nominated or provided by the caller that is presented to the receiving party when a call is made. It can identify that caller and can be used to make a return or subsequent call.

#### Presentation Numbers

The Supplier supports four of the five available Presentation Number types with the SIP Trunking service:

- **Type 1** - generated by the Supplier network. The number is stored in the network and applied to an outgoing call at the local exchange by the provider.
- **Type 2** – this identifies a caller's extension number behind a DDI switchboard. Although the number or partial number is generated by the Buyer's own equipment, the network provider is able to check that it falls within the range and length allocated to a particular subscriber.
- **Type 3** – limited to the far-end breakout scenario where a call's access to the public network may be different to where it was originated (e.g. where there are several transitions between several service providers). The number is generated by the Buyer's equipment but isn't capable of being subjected to network verification procedures.
- **Type 5** – Presentation numbers that identify separate groups of callers behind a private network switch wishing to send different outgoing CLIs. A typical scenario is a call centre making calls on behalf of more than one client.

### 2.3.1.2 Calling Line Identity Restriction (CLIR)

Restricting or withholding the presentation CLI is typically carried out on the PBX but the Supplier's SIP Trunking service can restrict the presentation CLI for all calls if required.

### 2.3.1.3 CLI Formats

#### Incoming Calls

For incoming calls SIP Trunking calls can be configured to send the following format options depending on the chosen data connectivity (see Table 3 below):

- Global = +442031234567
- National Dialable = 02031234567

| Data Connectivity | CLI Format |
|---|---|
| IPVPN/SD-WAN | Global<br>National Dialable |
| Internet | Global |
| Hybrid Access | Global |

*Table 3: **CLI Format Options***

**Outgoing Calls**

For outgoing calls, it is recommended that the Buyer configures the CLI format to match the incoming call CLI format.

### 2.3.2 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 which can configured by the Buyer and requires the SIP Trunking Portal Core Feature package.

### 2.3.3 Incoming Call Barring

Inbound call barring can be configured by the Supplier on individual numbers or ranges if required. When activated, all inbound calls to the number range will be barred. Standard MAC (Moves, Adds and Changes) charges apply.

### 2.3.4 Outbound Call Barring

Outbound call barring is typically applied from the Buyer's PBX but can be configured on the SIP Trunking service. Options available are as follows:

- Mobile – not allow outgoing calls to mobile numbers
- International – not allow outgoing calls to international numbers.
- Premium – not allow outgoing calls to premium rate (09) numbers.
- Directory Enquiries – not allow calls to 118 numbers

### 2.3.5 Divert on Failure DDI

One DDI can be configured to automatically divert to an alternative number should it not be possible to terminate calls to the chosen DDI due to a 'local' failure for example, an issue with the Buyer's connectivity or eSBC.

This would typically be the Buyer's switchboard number. Further divert options are available with the SIP Trunking Portal (see 2.6 for further details).

### 2.3.6 Emergency Calling

The Supplier's SIP Trunking service offers access to Emergency Services in compliance with Ofcom General Condition 4.

When an emergency call is made from a SIP endpoint, information is presented to the operator to flag it as a VoIP call. This flag ensures the operator asks the caller to confirm their location as the CLI presented may not represent the actual location.

### 2.3.7 Long call duration

Calls made on the SIP Trunking platform will remain up for a maximum duration of 24 hours at which point the call will be dropped.

## 2.3.8 DTMF

In band (or RFC2833) and out of band DTMF are supported on the SIP Trunking service.

## 2.3.9 Encryption

The SIP Trunking service supports encryption of both signalling and media streams as follows:

- IPVPN/SDWAN – the standard offering is non-encrypted. Optional support for encryption
- Internet – the standard offering is encrypted using TLS and SRTP

## 2.3.10 Supported Codecs

SIP is a VoIP service, which means that a conversation is digitally encoded for transport over the IP network and this encoding is done using a Codec.

The SIP Trunking service supports the two most commonly used Codecs as described below as well as G.722, AMR and AMR-WB:

- **G.711** (A-Law and U-Law) is a Codec used for voice compression and is comparable to PSTN quality calls. It uses approximately100kbps of bandwidth to carry one simultaneous call. Typically, a G711 call will provide a Mean Opinion Score of 4.1 and above.
- **G.729** is a Codec used for voice compression and is comparable to ISDN quality calls. It uses approximately 24-35kbps of bandwidth to carry one simultaneous call. Typically, a G729 call will provide a Mean Opinion Score of 3.7 and above.

As standard Codecs will be configured with a payload size of 20ms but 10ms is available on request to the Supplier.

**NB:** If the Buyer will use an Internet data connectivity service with the SIP Trunking service, then G.711 must be selected if the Buyer needs to be able to send and receive faxes.

## 2.3.11 Fax, PDQ and Alarm Support

### Fax Support

The SIP Trunking service supports fax transmission as follows:

- IPVPN/SDWAN – T38
- Internet – G.711 passthrough fax using SRTP

Due to the wide variety of proprietary extensions to fax standards, the Supplier cannot guarantee interworking between all fax terminals.

### PDQ Machines Support

Credit Card (Process Data Quickly) machines used over SIP connections unfortunately have problems and may not work with any consistency. This is due to the wide variety of machines and different implementations and the codecs used by ATAs have been designed to compress voice, not the analogue signals sent and received by modems.

The Supplier cannot guarantee that PDQ machines will work on the SIP Trunking service.

### Alarm systems

Many alarm companies are now developing services to work with VoIP but many do not work correctly. Consequently, the Supplier cannot not guarantee the SIP Trunking service will inter working with these deployments.

## 2.3.12 Numbering

The Buyer is responsible for providing the required geographic numbers (see New Numbers and Ported Numbers below).

### 2.3.12.1 New Numbers

New geographic numbers can be requested as part of an order for SIP Trunking or in-life.

The Supplier will manage the assignment and allocation of new geographic numbers from the Supplier's own number ranges. The Supplier will allocate number ranges according to the Buyer requirement, accounting for number types, dial code and volume of DDIs required.

Please note 0203 has now replaced 0207 and 0208 for new London number requests.

### 2.3.12.2 Ported Numbers

Geographic numbers can be ported from Other Licensed Operators (OLOs) to the SIP Trunking Service. Both Single Line and Multi-Line porting are supported. Once a number port request has been placed and accepted by the Losing Communications Provider (LCP), these numbers can be added to a SIP Trunking Service.

All porting requests must contain clear and accurate information. Incomplete or unclear information may lead to delays and additional charges. The Supplier will perform number porting using the information supplied by the Buyer and in accordance with all relevant legislation and regulations (including codes of practice).

## 2.3.13 Billing

The Supplier can provide the following types of billing with the SIP Trunking service:

- Main Billing Number
- DDI Billing

**NB** only Type 2 Number Presentation can be supported with DDI Billing.

## 2.4 SIP Authorised Equipment, Equipment Configuration and Onboarding

### 2.4.1 SIP Authorised Equipment

It is the Supplier's default position to always deliver the SIP Trunking service into an eSBC (the SIP Endpoint), as opposed to a PBX directly. The eSBC acts as a security boundary between the Buyer's PBX and the SIP Trunking service.

Depending on the PBX manufacturer, the manufacturer may stipulate the use of a particular SBC with their PBX (i.e. their own).

The Buyer must only connect "SIP Authorised Equipment" to the SIP Trunking service which is an eSBC that **meets the minimum specification** detailed in Table 4 below, and which has successfully passed the onboarding tests.

| eSBC Requirement | IPVPN/SDWAN | Public Internet |
|---|---|---|
| RFC 2833/4733 DTMF | X | X |
| RFC 3261 | X | X |
| UDP | X | |
| RTP | X | |
| TLS | | X |
| SRTP | | X |
| Registration via SRV Domains | | X |

*Table 4:* ***eSBC Minimum Specification***

The Supplier also runs an Interop program for testing the SIP Trunking Service with key SBCs which enables a shorter onboarding process. Please contact the Supplier for further details.

The Buyer may modify or change the SIP Authorised Equipment connected to the SIP Trunking Service provided the Supplier has agreed the modification or change.

### 2.4.2 Equipment Configuration

The Buyer will be responsible for configuring their PBX, SIP Clients and other equipment within the Buyers's local network. The Buyer must configure their eSBCs in accordance with the SIP Trunking Configuration Guide using the details supplied during the onboarding process.

The Supplier recommend the Buyer enables QoS for SIP traffic in their network. This will enable appropriate prioritisation within the Buyer's organisation where applicable.

When deploying the SIP Trunking Service, the Buyer may need to amend local firewall polices to allow the SIP Trunking Service to function correctly.

### 2.4.3 Onboarding

The Supplier understands the importance of making the onboarding process for the SIP Trunking service as seamless as possible for the Buyer which is summarised below:

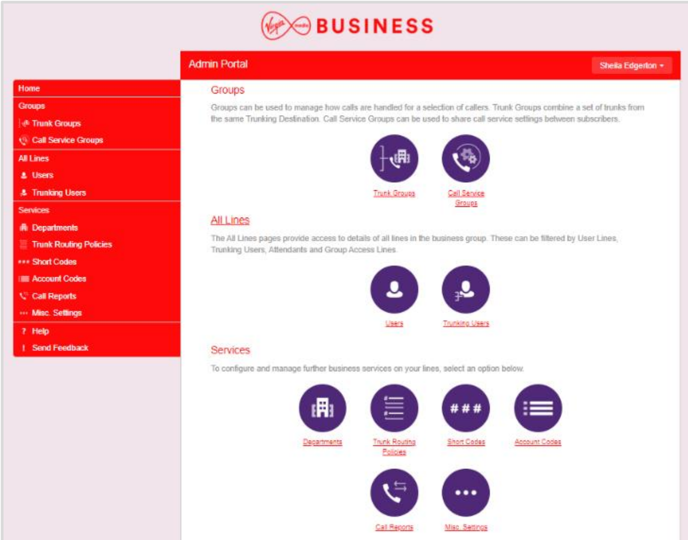
- The Buyer will be provided with an Order Manager who will be in contact throughout the entire onboarding process.
- The Order Manager will contact the Buyer after the order has been placed to confirm the configuration details needed for setting up the Buyer's SIP Trunking service.
- The Buyer's SIP Trunks will be provisioned on the SIP Trunking platform, any DDIs configured, and additional voice features are set up e.g., CLIP, CLIR, and Presentation Numbers.
- To help ensure that the testing goes as smoothly as possible, there will be a 30-minute Pre-Onboarding call to run through the onboarding process and check the Buyer is ready for the Onboarding process.
- Testing of the SIP Trunking service normally takes between 4-5 hours; this is dependent on the speed of the test calls and could be completed in a shorter period.
- The Supplier will analyse the results of the tests using SIP Trace Analysis.
- On completion of the analysis the Order Manager will contact the Buyer to confirm a Go-Live date for the SIP Trunking service.

## 2.5 Minutes

The Buyer will be charged for outbound calls made over their SIP Trunking service with two options to choose from:

- **PAYG** – pence per minute and/or call charging for all number types/destinations.
- **Flat Rate Tariff** – offers a bundled call package which includes a call allowance of voice minutes to UK 01/02/03 numbers and 07 numbers (the five principle mobile operators), with pence per minute and/or call charging for all number types.

# 2.6 SIP Trunking Portal




The SIP Trunking Portal provides Buyers with a web-based access to a configuration and reporting portal.

The Portal allows authorised users (Administrators) to make changes to calling features and have access to trunk side call reports for both inbound and outbound calls.

Administrators can view all DDIs or only those assigned to certain departments.

The SIP Trunking Portal is optional and can be ordered with the new SIP Trunking service and added or removed in-life without penalty.*

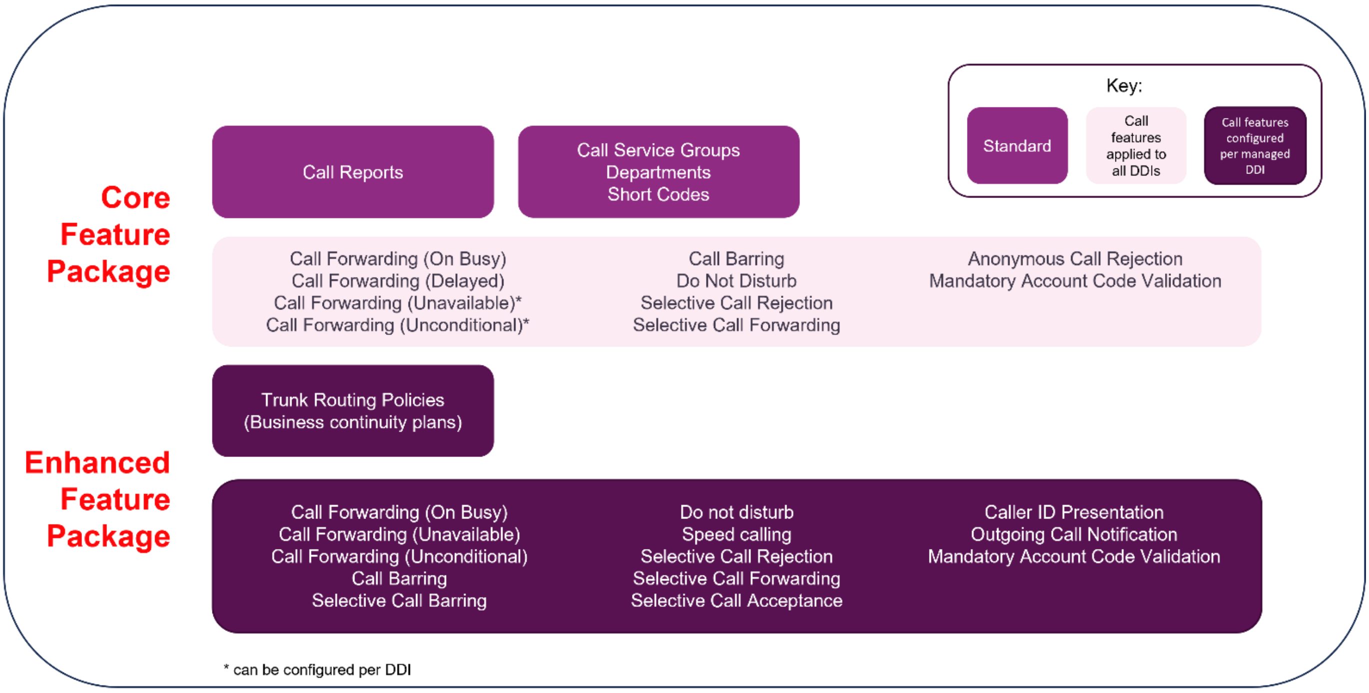
** Minimum 3 month term applies.*

## 2.6.1 Feature Packages

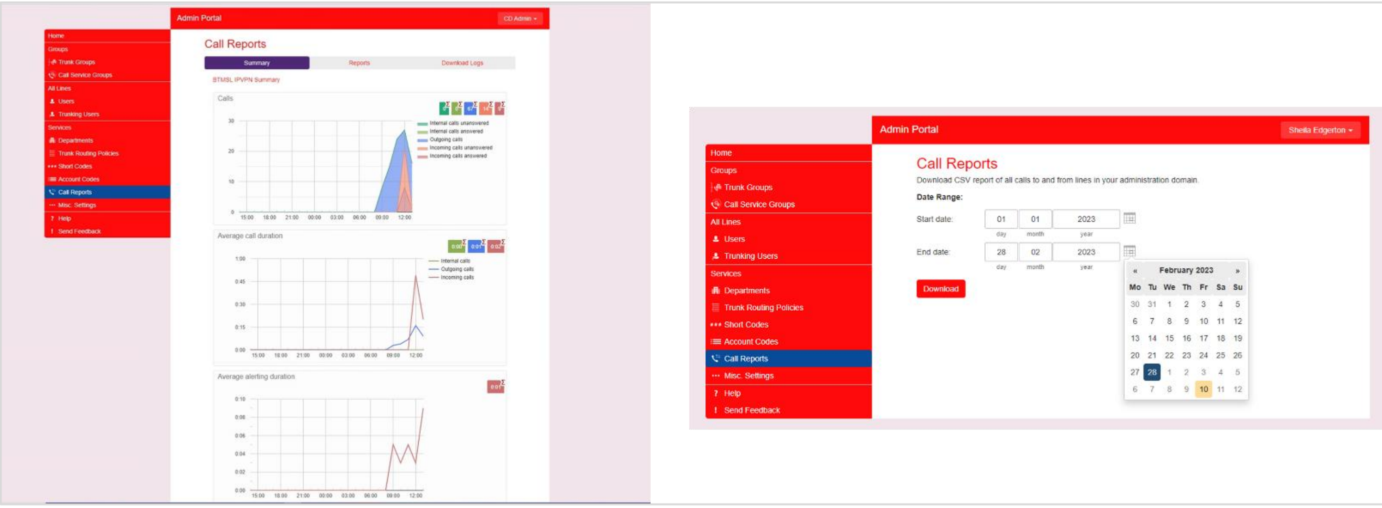
There are two Feature Packages available with the SIP Trunking Portal:

- **Core Feature Package** – is the base feature package and includes a rich reporting on incoming and outgoing calls for all DDIs, call forward options and ability to group DDIs within departments.
- **Enhanced Feature Package** – is an optional package which provides further features including the ability to create business continuity plans and can be purchased only for the DDIs the Buyer requires the additional functionality for.

The available features are summarised below:



## 2.6.2 Call Reports




Administrators can use the Call Reports screen to view and schedule a range of call activity reports and download call logs in csv format. If required Administrators can be limited to being only able to download call logs.

New reports can be created on an ad-hoc basic or scheduled on a regular basis and can include the following parameters:

- **Account Statistics**: outbound call statistics, shown by account codes
- **Call Duration Summary**: shows the length of incoming and outgoing calls
- **Call Log**: detailed overview of calls
- **Frequent Caller Summary**: the telephone number of callers who made the most incoming calls
- **Frequently Called Summary**: like Frequent Caller Summary, but grouped and ranked using the target telephone number
- **Location Analysis**: shows the location of incoming and outgoing calls, based on telephone number and location information configured in the portal.
- **Long Ringing Time**: contains any incoming calls that rang for longer than a configured threshold
- **Missed Calls Summary**: shows a high-level summary of incoming calls that were not connected
- **Missed Calls Detail**: shows all incoming calls that were not connected
- **Short Calls**: shows answered incoming calls that were shorter than a customisable threshold,
- **Top Talkers**: call data for users including total and answered calls and data on call duration, ordered by total call duration
- **Traffic by Day**: call data arranged by day
- **Traffic by Hour**: call data arranged by hour
- **Unreturned Calls Detail**: this is the same as Missed Calls Detail but minus any calls that were subsequently connected during the period of the report
- **User Statistics**: data on calls and call duration by each DDI

# 3.0 Glossary

### CLI

Caller Line Identification is a telephone service, available in analogue and digital phone systems and most voice over Internet Protocol (VoIP) applications, that transmits a caller's number to the called party's telephone equipment during the ringing signal, or when the call is being set up but before the call is answered.

### DTMF

Dual-tone multi-frequency signalling (DTMF) is an in-band telecommunication signalling system using the voice-frequency band over telephone lines between telephone equipment and other communications devices and switching centres.

### PBX

A Private Branch Exchange is a Buyer telephone system which delivers voice over a data access network and interoperates with the SIP Trunking service.

### PSTN

The public switched telephone network (PSTN) is the aggregate of the world's circuit-switched telephone networks that are operated by national, regional, or local telephony operators, providing infrastructure and services for public telecommunication.

### RTP

The Real-time Transport Protocol (RTP) is a network protocol for delivering audio and video over IP networks.

### Session Border Controller (SBC)

An SBC starts, stops and controls the signalling for a voice call. It provides a boundary between these sessions so they can securely traverse different networks to reach their destination. And it provides interoperability between different Voice over IP (VOIP) services.

### UDP

UDP (User Datagram Protocol) is an alternative communications protocol to Transmission Control Protocol (TCP) used primarily for establishing low-latency and loss tolerating connections between applications on the Internet.

### VoIP

Voice over IP (VoIP) is a methodology and group of technologies for the delivery of voice communications and multimedia sessions over Internet Protocol (IP) networks, such as the Internet.

# Company Statement

All information contained in this proposal is subject to contract and, unless expressly stated to the contrary, to Virgin Media Business Limited's standard terms and conditions.



Registered office: 500 Brook Drive, Reading, RG2 6UU, United Kingdom

**Brand Statement**

Buyers entering into a Call-Off Contract will be contracting with Virgin Media Business Limited ("The Supplier").

On the 1st August 2025 Virgin Media O2 Business and Daisy Group merged to create a new B2B telecoms powerhouse – O2 Daisy.

O2 Daisy is fully consolidated by Virgin Media O2 with a 70/30 ownership structure with Daisy Group. The corporate brands Virgin Media O2 Business, Virgin Media Business and O2 Business remain.

For contracting purposes, customers of these brands contract with Virgin Media Business Ltd. Depending on the services to be provided, affiliate companies may act as billing and collection agents, such as Telefonica UK.

Virgin media $O_2$

Business