SIP Trunking

Product Handbook

How to use this guide

This is an interactive guide which will help you to find the information you need on SIP Trunking. You can do this by using the right side banner, which displays each section of the document. All you need to do is click on the title you're interested in and the hyperlink will take you there. To view the full table of contents click on the top right side of the banner.

There are contact and help details at the end of the guide, if there is further information you need which isn't included in this document.

Why O_2 SIP Trunking?

ISDN to O_2 SIP Trunking

For businesses that rely on ten or more channels of ISDN, switching to SIP can deliver immediate cost savings. SIP channel rentals and calls are typically up to 40% cheaper than traditional ISDNs, and businesses have the extra benefit of consolidating their voice and data services into one single line for further savings.

- Move offices and keep the same geographic number with no call forwarding costs.
- Maintain better business continuity. Re-route calls to alternative destinations quickly and easily in times of disruption.
- Reduce call costs. IP connectivity costs less than ISDN and offers free internal calls with no call forwarding costs.
- Rationalise ISDN lines and reduce the number of PBXs – PBXs can even be hosted in the cloud.
- Rather than paying for unused capacity, the agile business can scale resources, meaning they will only ever have to pay for what they need.
- Truly unified communications support integration with existing line of business applications, such as Skype for Business.
- Predictive call spend with call bundle options of 01, 02 and 03 and mobile 07 calls.
- With O_2 Gateway use a single connection for high quality voice and data services.

Connectivity options

With O_2 SIP Trunking, ethernet connectivity will be provided via either our converged multi-service connectivity, O_2 Gateway, or O_2 Dedicated connectivity. These options are described further below and our dedicated solutions architect will assist you in deciding the best solution for your business needs.

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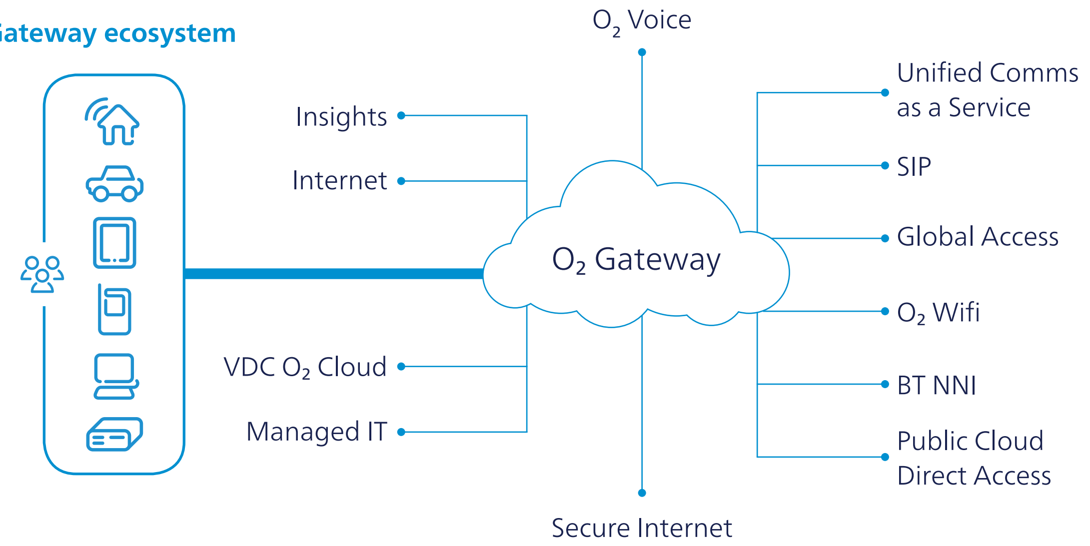
O_2 Gateway

O_2 Gateway offers dedicated connectivity for SIP Trunking but also a converged access capability for bringing multiple services across the same access connectivity with one point of contact and one SLA.

O_2 Gateway offers connectivity to the largest digital ecosystem in the market, allowing customers direct cloud connectivity to Amazon Web Servers, Microsoft Azure, Office 365, Google and Oracle cloud services. It seamlessly enables SIP Trunking, unified communications as a Service and O_2 Wifi access. And through our Telefonica global reach, O_2 Gateway WAN connectivity can be extended to over 125 countries around the world.

Once connected to O_2 Gateway, you'll be able to roll out new or additional services quickly and scale bandwidth up or down. Changes like these are unlikely to disrupt the physical infrastructure, which means rapid deployment and faster adoption.

The O_2 Gateway ecosystem



O_2 Dedicated connectivity

O_2 SIP Trunking connects your PBX to the O_2 network, enabling full PSTN breakout on the public telephone network. Connection from your site (or sites) to our network is via a dedicated IP connection (for example Ethernet FTTC, EFM or Fibre) and is delivered as an end-to-end service with an availability guarantee, voice channel guarantees and voice quality of service.

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O_2 SIP Trunking voice minutes

With the O_2 SIP Trunking service we have simplified how you pay for voice minutes with three simple packages.

- Pence per minute (ppm): When you just want to pay for what you use no minutes bundle need be purchased. In this case you will pay for all calls at the ppm rates defined in the rate card or in your O_2 contract. This tariff is shown as STRPPM on your O_2 bill.
- UK domestic bundle: This bolt-on option provides inclusive calls to 01, 02, 03 numbers (subject to a 5,000 minute per channel per month fair usage policy). This tariff is shown as STRAAA on your O_2 bill.
- UK domestic and mobile bundle: This bolt-on option provides inclusive calls to 01,02, 03 (subject to a 5,000 minute per channel per month fair usage policy) and 07 (mobile) numbers (subject to a 2,000 minute per channel per month fair usage policy and three year contract term). This tariff is shown as STRAAM on your O_2 bill.

Save money

IP connectivity costs less than ISDN with lower call costs, free internal calls between extensions and offices and lower line rental costs for multi-sites. Also, no expensive call-forwarding costs are required should you relocate or need to divert calls in the event of a disaster.

Line rationalization

For businesses with multiple sites, SIP Trunking provides the opportunity for line rationalisation and reduces the number of PBXs you need to maintain – while retaining full control of the numbers associated with your business.

Resilience

SIP Trunking provides a phone service that will cope with any situation and give you business-grade resilience for your telephony. Whether you need to keep your business running in a disaster or emergency or you need to load balance your calls between sites during peak hours, SIP Trunking delivers.

Flexibility with phone numbers

SIP Trunking enables you to move office and keep the same geographic number without any ongoing call-forwarding costs or those associated with producing new company stationery.

Business continuity

If your office has to be temporarily relocated in an emergency, this can be achieved quickly and cost-effectively with SIP Trunking to keep your business working.

PCI compliance

Semafone enabled SIP Trunking solves both compliance and security issues in one stroke by removing all the sensitive card data from your organisation. Our payment method uses DTMF (Dual Tone Multi Frequency) masking technology to conceal the sound of the keypad tones so you can enter their payment card numbers through the telephone handset.

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What flavour of SIP Trunking

O_2 offers five different flavours of SIP Trunking to meet the business needs. Each option type has a new order form that can be completed in [Adobe Actobat Reader](#).

All order forms are available at www.o2.co.uk/business/sip-trunking

SIP over public internet
Public internet connection between your on premise PBX and the O_2 SIP network.

SIP over JANET network
A JANET network connection between your on premise PBX and the O_2 SIP network.

SIP over single Ethernet
A single private Ethernet connection between your on premise PBX and the O_2 SIP network. Supports dedicated connectivity.

SIP over multiple Ethernet
Resilient private Ethernet connections between your on premise PBX(s) and the O_2 SIP network. Supports dedicated connectivity.

SIP over O_2 Gateway
Single or resilient private Ethernet connections between your on premise PBX(s) and the O_2 Gateway. Supports additional O_2 Gateway services.

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SIP channel options

The table below describes the SIP channel options available to O_2 SIP Trunking

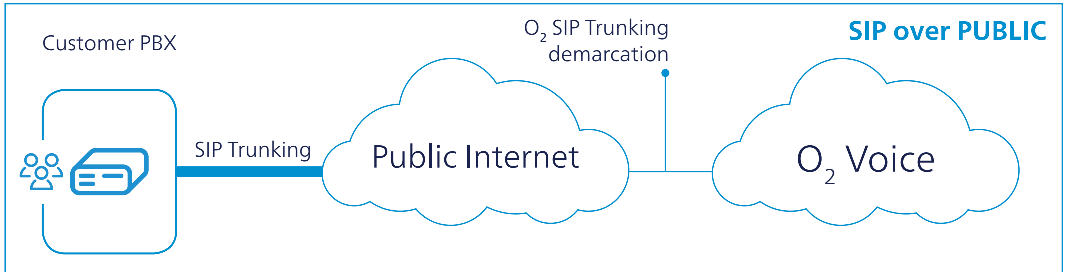
Channel type	Single	Active -standby	Resilience+	Loadshare
Description	Single route to the customer PBX to SBC cluster on the core network	Active - standby routes to the customer PBX	Active - standby routes to the customer PBX with DDI resilience.	Active - active loadshare routes to the customer PBX
SIP Trunking type	SIP over PUBLIC SIP over JANET SIP over ethernet single SIP over O_2 Gateway	SIP over PUBLIC SIP over JANET SIP over ethernet Multi SIP over O_2 Gateway	SIP over PUBLIC SIP over JANET SIP over ethernet Multi SIP over O_2 Gateway	SIP over PUBLIC SIP over JANET SIP over ethernet Multi SIP over O_2 Gateway
Semafone enabled SIP Trunking	•	•	•	•
SLA	99.95%	99.99%	99.99%	99.99%
Fraud management system (FMS)	•	•	•	•
Emergency Call Divert	•	•	•	•
High Availability Network SBC Pair	•	•	•	•
Geographical resilience Network SBC's	Not supported	•	•	•

Resilience+

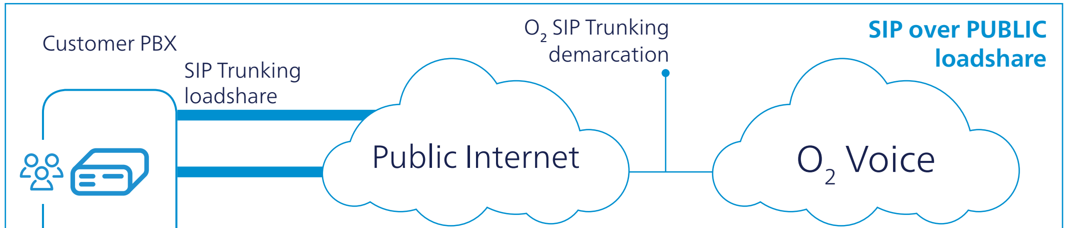
Resilience+ offers dual endpoints both in an active-standby setup, individual DDI ranges are allocated to each SIP endpoint and traffic routes accordingly. In the event of SIP endpoint being unavailable all calls will route to the alternate SIP endpoint. Benefits of the Resilience+ the ability to define both the DDI ranges and channel allocations at each site. In the event of SIP endpoint being unavailable, then the remaining site will receive all related traffic. Resilience+ accounts do not require symmetrical channels for example a 100 channel deployment can be allocated in a 60/40, 70/30 configuration if required

SIP over PUBLIC

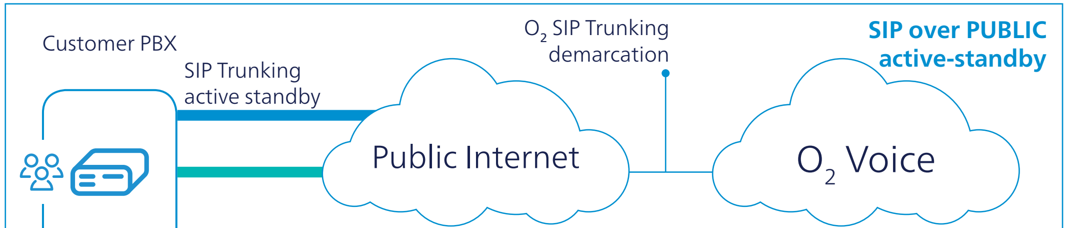
O_2 SIP Trunking can work with a third party public internet connection. Having an internet connection can be advantageous, provisioning time and cost is reduced using the existing connectivity from your PBX to the O_2 SIP Trunking service.



Single SIP connection between customer PBX and O_2 network.

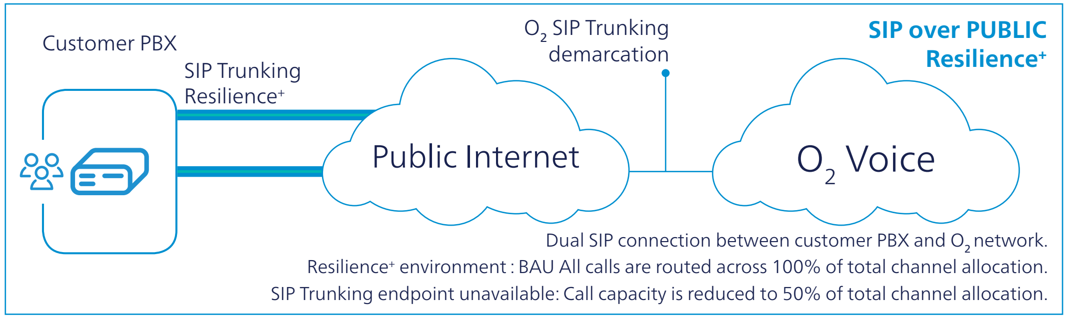


Dual SIP connection between customer PBX and O_2 network.
Loadshare environment : BAU All calls are routed across 100% of total channel allocation.
SIP Trunking endpoint unavailable: Call capacity is reduced to 50% of total channel allocation.



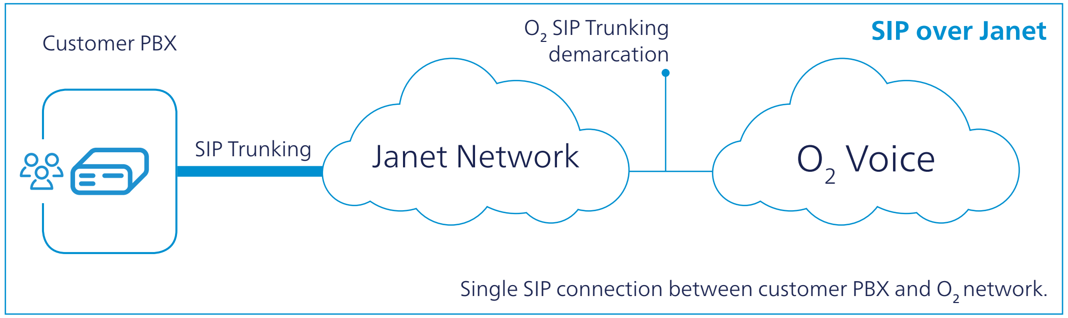
Dual SIP connection between customer PBX and O_2 network.
Active environment : BAU All calls are routed across the 'active' route.
SIP endpoint unavailable: All calls are routed across the 'standby' route.

O_2 business



SIP over Janet

JANET is a high-speed network for the UK research and education community. Serving over 18 million users, the JANET network provides UK research and education with a highly reliable and secure, world-class network, enabling national and international communication and collaboration. O_2 SIP Trunking is accredited with 'JANET Connected' with a minimum of two geographically diverse connections between us and JANET, allowing traffic to flow directly between the two networks.



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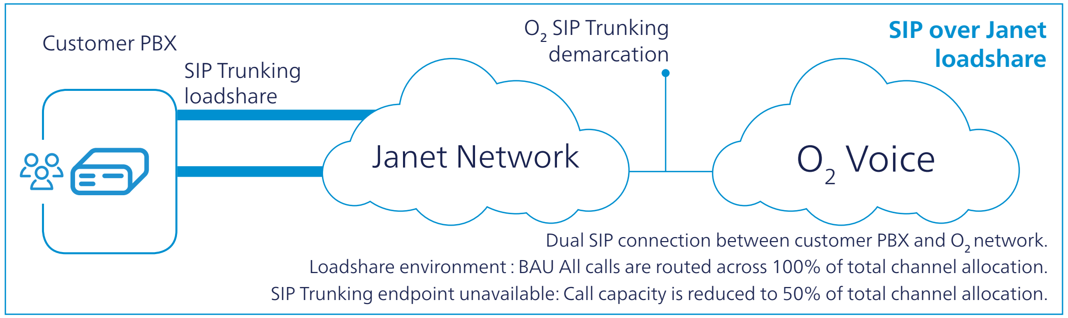
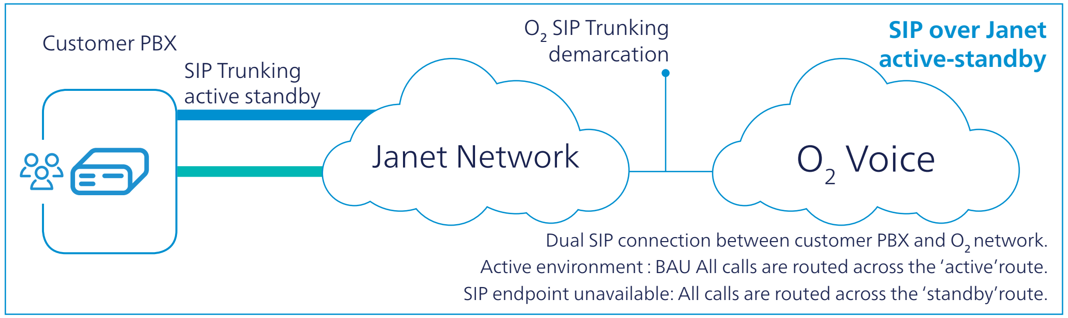
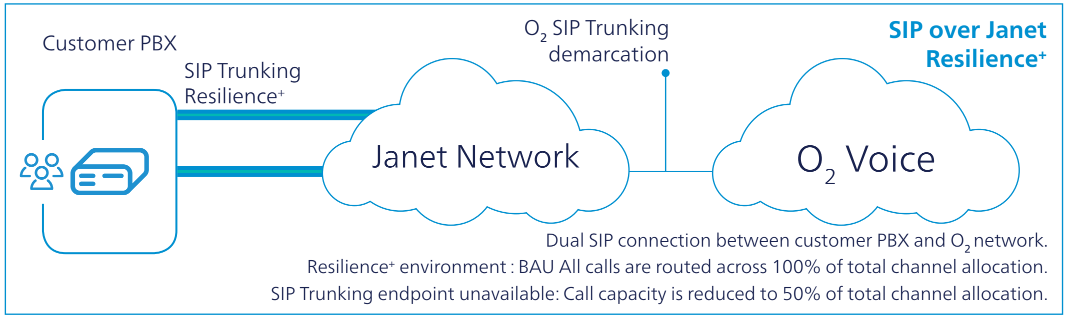
O_2 business







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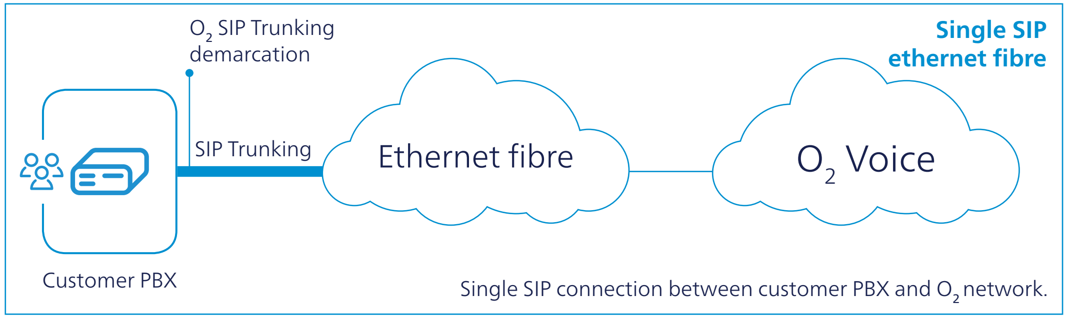
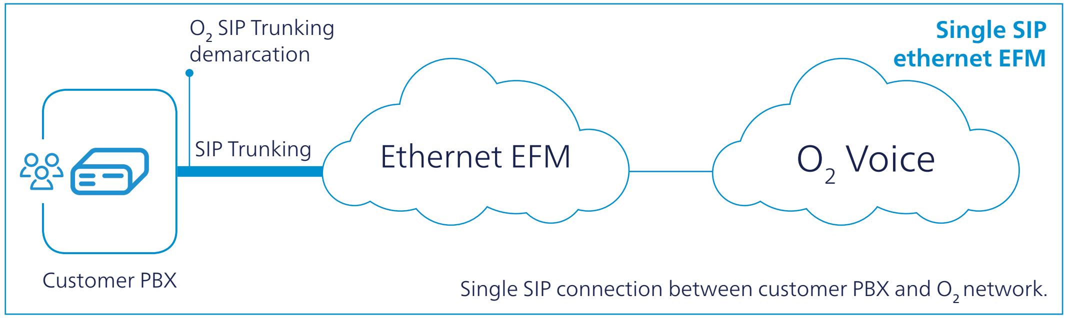
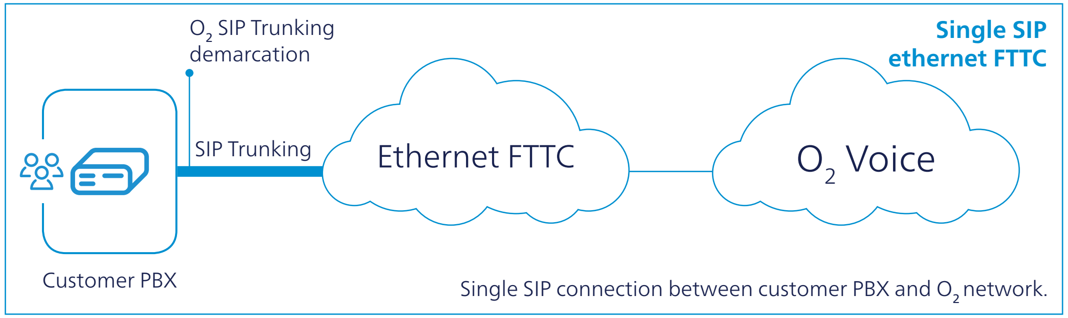
O_2 business

SIP over single ethernet

With O_2 SIP Trunking, ethernet connectivity will be provided via O_2 dedicated connectivity. These options are described further below and our dedicated solutions architect will assist you in deciding the best solution for your business needs.







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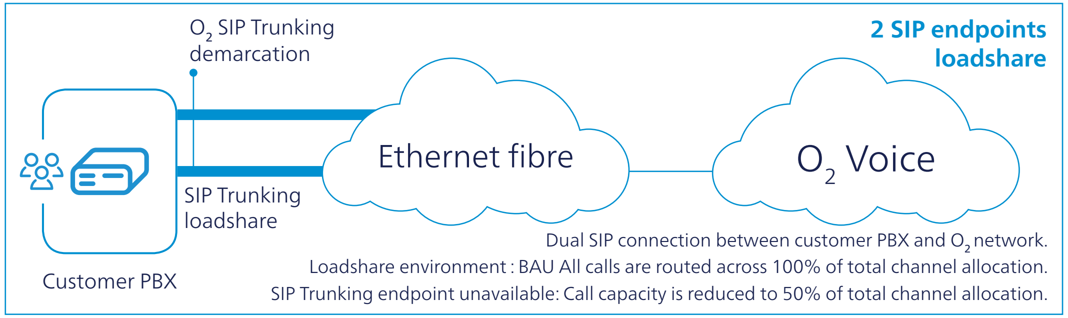
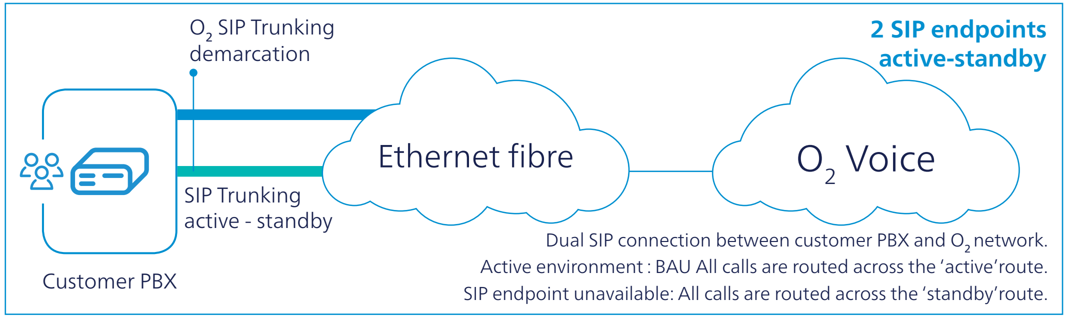
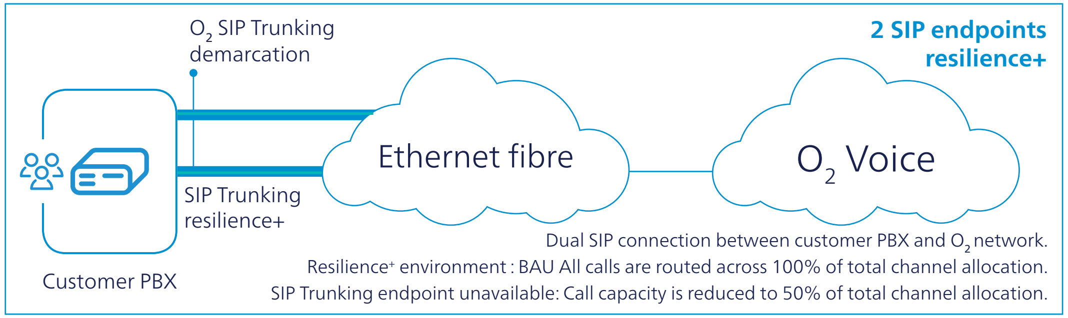
O_2 business

SIP over multiple ethernet

With O_2 SIP Trunking, ethernet connectivity will be provided via O_2 dedicated connectivity. These options are described further below and our dedicated solutions architect will assist you in deciding the best solution for your business needs.







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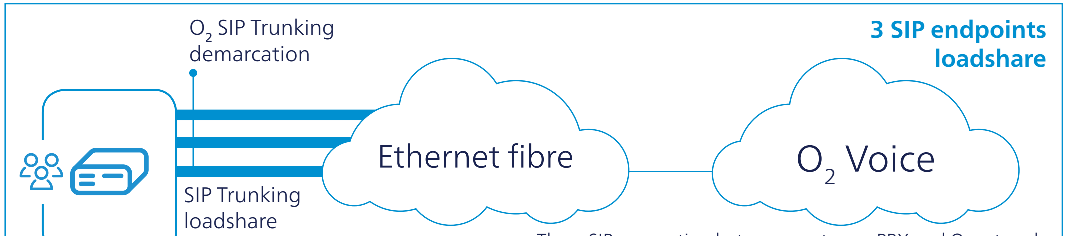
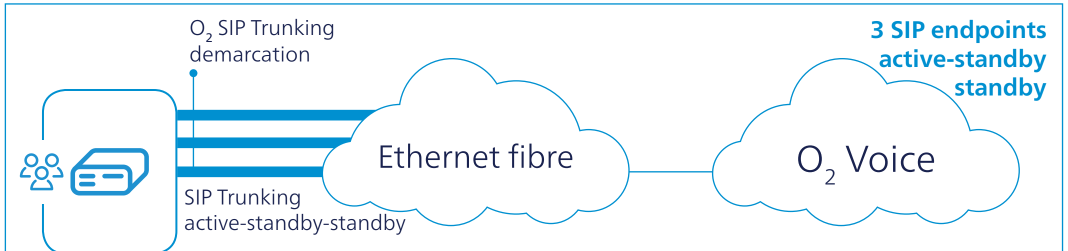
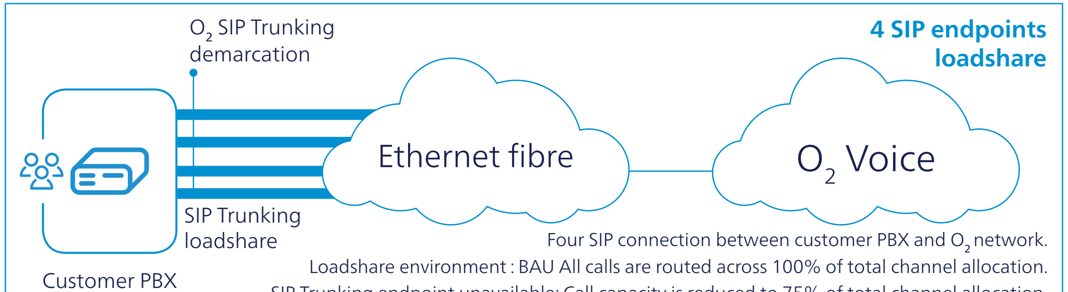
O_2 business



Three SIP connection between customer PBX and O_2 network.
Loadshare environment : BAU All calls are routed across 100% of total channel allocation.
SIP Trunking endpoint unavailable: Call capacity is reduced to 66% of total channel allocation.



Three SIP connection between customer PBX(s) and O_2 network.
Active environment : BAU All calls can are routed across the 'active' route.
SIP endpoint unavailable: All calls are routed across the first 'standby' route.
Second SIP endpoint unavailable: All calls are routed across the second 'standby' route.



Four SIP connection between customer PBX and O_2 network.
Loadshare environment : BAU All calls are routed across 100% of total channel allocation.
SIP Trunking endpoint unavailable: Call capacity is reduced to 75% of total channel allocation.

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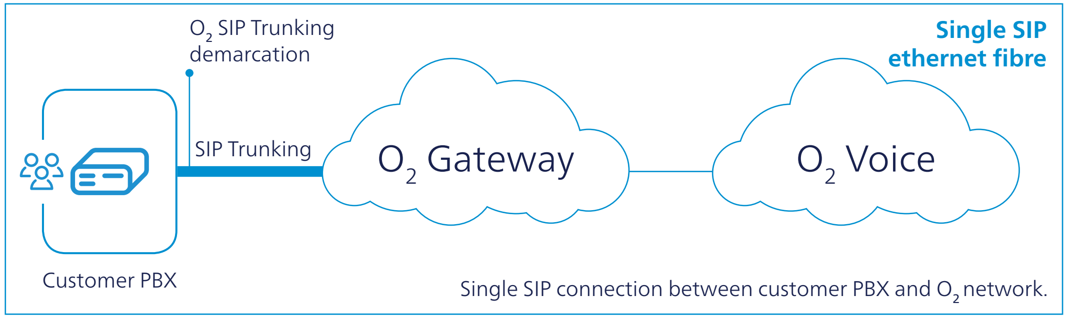
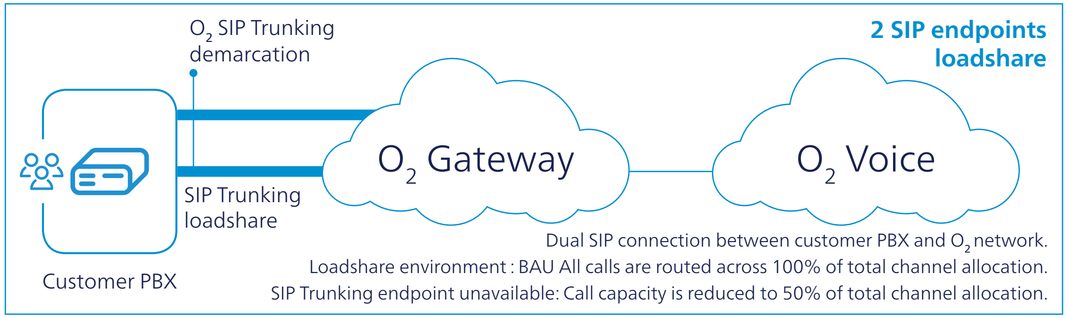
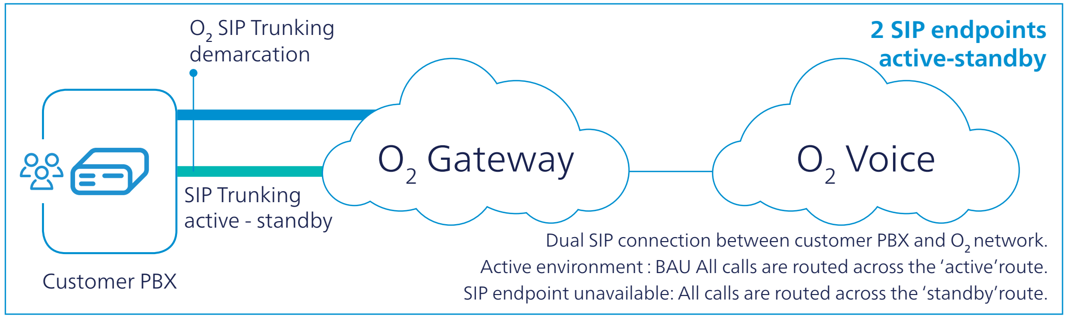
SIP over O_2 Gateway

With O_2 SIP Trunking, ethernet connectivity will be provided via our converged multi-service connectivity, O_2 Gateway. Once connected to O_2 Gateway, you'll be able to roll out new or additional services quickly and scale bandwidth up or down These options are described further below and our dedicated solutions architect will assist you in deciding the best solution for your business needs.







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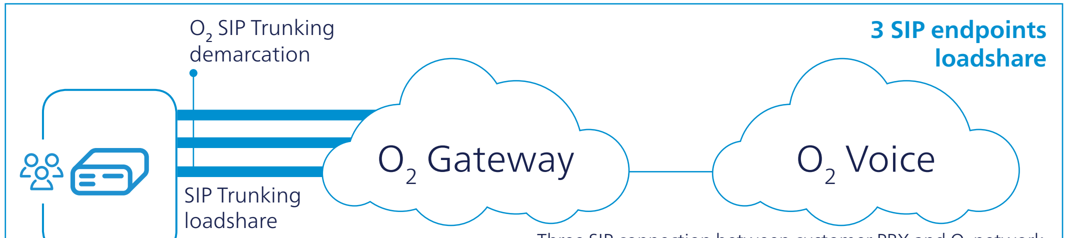
O_2 business

Dual SIP connection between customer PBX and O_2 network.
Resilience+ environment : BAU All calls are routed across 100% of total channel allocation.
SIP Trunking endpoint unavailable: Call capacity is reduced to 50% of total channel allocation.



Three SIP connection between customer PBX and O_2 network.
Loadshare environment : BAU All calls are routed across 100% of total channel allocation.
SIP Trunking endpoint unavailable: Call capacity is reduced to 66% of total channel allocation.

Three SIP connection between customer PBX(s) and O_2 network.
Active environment : BAU All calls can are routed across the 'active' route.
SIP endpoint unavailable: All calls are routed across the first 'standby' route.
Second SIP endpoint unavailable: All calls are routed across the second 'standby' route.

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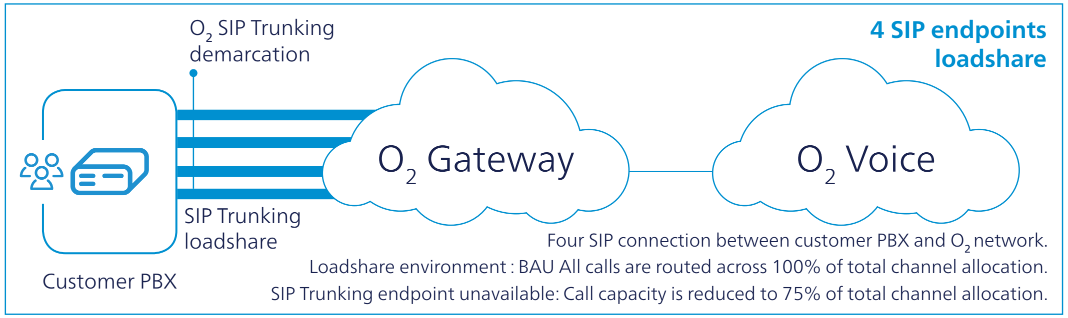

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Semafone enabled SIP Trunking

What is PCI DSS compliance?

The Payment Card Industry Data Security Standard (PCI DSS) is the proprietary information security standard defined by the major card companies to help combat fraud and protect consumer card data. Its members include Visa, MasterCard, American Express, Discover and JCB.

What is the Semafone PCI service?

The Semafone PCI service consists of two elements; the Semafone PCI solution and the Semafone enabled SIP Trunks which are integrated with the Semafone PCI solution.

The Semafone PCI solution holds all four of the leading security and payment accreditations: ISO 27001:2013, PADSS certification, PCI DSS Level 1 Service Provider and Visa Level 1 merchant agent status.

The Semafone PCI solution uses DTMF (Dual Tone Multi Frequency) masking technology to conceal the sound of the keypad tones so your customers can enter their payment card numbers through the telephone handset in a secure manner. The numbers are sent straight to the Payment Service Provider e.g. Sage Pay, which means the sensitive card details never enter your network and neither are they seen by the agent as the numbers are replaced by asterisks on the agent's screen. Whilst attaining PCI DSS compliance certification remains the responsibility of the customer, the Semafone PCI solution takes you out of scope for many of the controls required to be PCI DSS compliant.

What SIP Trunking features are support?

Semafone enabled SIP Trunking is available for all 'flavours' of SIP Trunking and all features offered on the standard SIP Trunking service. However, Semafone enabled SIP Trunks are classed as a 'manual build' and changes to those SIP endpoints will take longer see our In life changes section for details.

What are my Semafone PCI solution purchase options?

Customers have the option to purchase the Semafone PCI element directly from Semafone, a third party or you can request O_2 to support this through our managed service offering.

Can I upgrade from Standard SIP Trunking?

The upgrade path requires a new standalone Semafone enabled SIP Trunking service to be provisioned. Once testing is completed the associated DDIs on the standard SIP Trunking service will be migrated to the new Semafone enabled SIP Trunking service.

What Service Levels apply to Semafone enabled SIP Trunking?

The Service Charter detailed in this handbook is applicable for both standard SIP Trunking and Semafone enabled SIP Trunking.

SIP service components

Numbering

Size of ranges available

In line with Ofcom requirements and best practice O_2 holds blocks of 1,000 or 10,000 numbers depending on location type (noting that it is likely that Ofcom will be allocating just 100 numbers blocks in some areas in the future). If O_2 holds a range of 10,000 numbers only 1,000 will be available until used up and a further 1,000 can be released. Consecutive number ranges above 300 may require a manual check.

In order for O_2 to be assigned a new number range certificate by Ofcom and then build and test this range over all major UK Communications Providers the average lead time is 2 months for a geographic range from all information required being made available.

Golden numbers

Golden numbers and other related commercial offerings are not recognised by Ofcom, who discourage the practice of 'cherry picking' number ranges as the consequence is the utilisation of ranges drops as it 'writes off' whole blocks. With this in mind O_2 does not supply golden numbers within its Geographic ranges.

Number retrieval

O_2 reserves the right to revoke the right to use a number(s) in certain cases including:

- If a customer does not make a new number(s) live within 3 months of its initial allocation;
- If a customer has ceased service on a number(s) and does not re-use the number(s) within 3 months;
- If a customer has a sizeable allocation of existing numbers on their account for a specific opportunity that is not implemented.

Conservation areas

Some locations offered by Ofcom are classed as a conservation area which means the number demand is so high in these locations that restrictions have been put in place. In these areas O_2 are only able to gain a range of a 1,000 numbers which can quickly be used up, before securing a new range. At present there are some 600 conservation area codes and these can be identified in the numbering plan. Please note

- 0207 and 0208 will be provided with alternative area code 0203

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999 and emergency services database

For all numbers where outbound telephone calls can be made, the customer is obligated to provide accurate caller location information to the emergency services database(s) and ensure this is accurately and timely maintained. The initial record is created via O_2 through the provisioning processes, and therefore it is imperative that the correct address information is provided by the customer.

Where a 999 call is made from an O_2 number and the address information found to be incorrect O_2 will be notified of this by BT and/or Ofcom and will update the customer. Ofcom require an amended record to be submitted within two days from notification, and where this is consistently not met, further action may well be taken by Ofcom against the O_2.

Number porting

Number portability

In order to port a number to or from O_2 the number being requested must be live, fall under porting agreements in place between the original range holder and the current owning Communications Provider, and the port must be authorised by the customer.

Where imported numbers are ceased on the O_2 network these will be returned to the original range holder in accordance with industry convention.

The industry has 'multi line' numbers and 'single line' numbers. This goes back to traditional telephony systems and how numbers were built within an exchange.

- O_2 can port numbers within UK including the Isle of Wight and Northern Island;
- O_2 can't port numbers within in bailiwicks of Jersey and Guernsey or the Isle of Man

The customer should be prepared for a short period of downtime whilst the numbers are ported from their previous operator across to the O_2 service. If all the pre-port checks and works have been completed the downtime should be minimal (minutes in most cases) and is an unavoidable part of the porting process.

Large number blocks

In some cases where a number port is required that is of sufficient size to be considered a whole OFCOM allocated geographic number block or where the customer owns the majority of numbers on a given block the geographic number block transfer process must be utilised.

Under ordinary circumstances the number blocks affected by this process will normally be those of 10,000 numbers or those of 1,000 numbers within a Type B conservation area.

Single lines

Single lines are lines that terminate on a single socket, and do not have any other form of associated numbers. The most common single line is your home telephone line (providing you just have the one number!). This is a single telephone line that terminates at your socket (i.e. the call is delivered to your line, and that is where the routing of the call finishes). These types of ports have shorter lead times.

Multi lines

Multi lines are lines that terminate on equipment such as a PBX. A Multi Line can be a single CLI that goes to numerous lines, or if it is an ISDN product.

Letter of authority LOA

It is both accepted industry process and a contractual requirement between networks that a (LOA) is available on request. For expediency and efficiency, it is taken on trust throughout the value chain that the required Letter of Authority exists. The (LOA) can be digitally signed and is considered with the same weight as one signed by hand. There is nothing in the porting industry guides to say that this has to be hand signed, so O_2 will accept a digitally signed (LOA). In the unlikely event where a LCP (losing communication provider) wouldn't accept a digitally signed LOA, the customer will be requested to hand sign the LOA.

Number porting CRF requirements

The following information will be required to complete this CRF.

1. Terminating SIP endpoint reference, if available
2. Digitally Signed letter of authority
3. Main billing number of the DDI range (indicated on your current bill)
4. Confirmation of your current supplier LCP (losing communication provider)
5. Confirmation of no associated products e.g Redcare
6. Port requested date and timeslot
7. 999 details for the DDI range,
8. Address location for the DDI range (indicated on your current bill)
9. Contact details (please include a 24*7 contact where possible, (for example building security).
10. Porting numbers from an existing ISDN30/2 service, please include all ranges that will remain on the ISDN30. Please note ranges can only remain active on ISDN30/2 if the main billing number is not ported.

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Range holder

The range holder is the communications provider (CP) who has been allocated a range of numbers by Ofcom that includes the number to be ported.

Out of hours porting

Out of Hours ports introduce a higher risk to the port order for a numbers of reasons. These include limited support offered by the range holder to complete the port, technical support staff at the range holder may not be available until the next working day and emergency restores can take up to 3 days to complete.

Where possible, O_2 would recommend to complete a port in hours as soon as possible in the day. The recommended window is early in the morning on a weekday as support at the range holders will be available sooner.

If OOH porting is required O_2 can investigate weekend cover but this will be a special request and subject to the availability of suitable resources at O_2 prior to O_2 being able to accept the OOH port request. O_2 will investigate the cover upon receipt of an OOH port request but availability is not guaranteed. Should no resources be available an in-hour porting slot will be suggested. If resources are available the OOH port will attract additional fees due to the need for a dedicated project manager needing to be available to ensure the port proceeds in the OOH time slot.

O_2 Managed porting

The O_2 porting project exists to provide an industry skilled resource acting as a point of contact to assist the customers with the coordination of a programme of porting orders. The aim is to reduce the risk of projects over-running and to support the smooth transition of services to the O_2 network. A porting project manager will be assigned to each validated porting project and this individual will be responsible for the delivery for all components accepted as within scope of the project through to overall project completion. The Project manager is contactable by phone and email within working hours for the duration of the project. Upon project completion, the services are formally handed over to the O_2 operational teams.

Service features offered

- Provide project management expertise and a point of contact for porting components
- Act on behalf of the customer to assist with third party management.
- Support the customer with the pre-validation of port requests and therefore reduce order rejections.
- Agree a communications plan and provide regular project reporting.
- Agree timescales and mitigate risks to ensure these are met.
- Help the customer reduce costs associated with rejections.
- Support with problem & change management

Service features not offered

- Reduce the contractual lead times or to expedite defined industry order processes.
- Place orders on your behalf for the customer.
- Manage the delivery of an O_2 SIP Trunking service used to terminate the number.
- Replace standard O_2 escalation processes.
- Porting numbers out of hours

Order rejections

Where an order is rejected by the losing communications Provider due to incorrect or missing information provided by the customer, O_2 reserves the right to charge, please see our online pricing sheet.

Order change/emergency restore

Last minute order changes and/or emergency restores divert resources away from servicing other customers and discharging our regulatory obligations. O_2 encourage our customers to only ever submit an order that can flow through without alteration and to ensure that the customer is ready for the change at the allotted time.

Date changes and cancellations can be made to a porting order up to 48 hours prior to the date of the port, additional charges will apply.

Export of an O_2 number to another network

The regulatory obligation to provide Number Portability to end users technically falls on the party with the contract with the end user, regardless of whether or not they have a network. When such a request comes in, O_2 validates it in accordance with established industry processes and provides losing notifications and manages the process with the gaining network from start to finish.

The provision of this service export of a telephone number on the O_2 network to another public electronic communications network is a chargeable fee, please see our online pricing sheet.

Internal number transfers

Porting an existing customer number or number range between active SIP Trunking endpoint

- Internal number transfers are a best endeavours process and the stated downtime cannot be guaranteed.
- When completing an internal number transfer the customer may transfer all or part of a number range
- Internal transfers are unrestricted in terms of number ranges so the request can be for a single number from the range if required or some of the range or all of the range.

Managed acceptance testing

The following tests are to be undertaken on site in the commissioning of the customer premises equipment (CPE) that is connecting to the O_2 SIP Trunking service.

Managed acceptance testing will include the following service acceptance tests and test results will be passed to CFOMT@O2.com. On successful completion of the tests, O_2 will complete the service provisioning and the service will be passed into support.

If managed acceptance testing is not purchased, the customer's PBX maintainer will be responsible for completing these (or similar) acceptance tests to validate the operation of the SIP Trunking. Should O_2 not receive a response or confirmation of completing acceptance testing within 10 working days of the service being commissioned O_2 will assume that the service is working as per required and the remaining activities will be completed to transition the SIP Trunking to in-life support.

- Test 1 – O_2 SIP Trunkings call – SIP clear
- Test 2 – SIP call – PSTN clear
- Test 3 – PSTN call – PSTN clear
- Test 4 – SIP call – SIP user release without answer
- Test 5 – SIP call – PSTN user release without answer
- Test 6 – Invalid number call test
- Test 7 – Incoming call – PSTN busy
- Test 8 - Incoming call – SIP user busy
- Test 9 - Address incomplete
- Test 10 - CLI Presentation test - PSTN > SIP
- Test 11 - CLIR test PSTN > SIP
- Test 12 - CLI presentation test - SIP > PSTN
- Test 13 - CLIR test SIP > PSTN
- Test 14 - CLI presentation test - PSTN > SIP
- Test 15 - Fax call to PSTN (If configured for Fax support)
- Test 16 - Call barring – If requested to be set up on the SIP account
- Test 17 – Call international number (if no international bar is in place)
- Test 18 – Dial 999 shortcode
- Test 19 – Dial 100 shortcode
- Test 20 – Dial 101 shortcode
- Test 21 – Dial 111 shortcode
- Test 22 – Dial 112 shortcode
- Test 23 – Dial 195 shortcode
- Test 24 – Dial 123 shortcode
- Test 25 – DTMF

Technical service overview

Voice policy guidelines

O_2 reserves the right to limit or prevent traffic that breaches our guidelines in the event any particular traffic presents a risk to the integrity of O_2's network.

Dialler policy

Diallers or any "automatic call generation" service connected to the O_2 network, must comply with the following standards:

- Ensuring an abandoned call rate (including a reasoned estimate of false positives) of no more than 3%[1] of live calls per campaign or per call centre over any 24 hour period;
- Ensuring that people are not contacted within 72 hours of their receiving an abandoned call without the guaranteed presence of a live operator;
- Playing an automated message in the event of an abandoned call telling the person called on whose behalf the call was made and providing them with a number to dial to stop any future marketing calls from that organisation;
- Making valid and accurate calling-line identification ("CLI") information available to call recipients so they can identify who rang them via caller display or by dialling 1471 in the event of a silent call; and
- Ensuring that where a call has been identified by dialler equipment as being picked up by an answer machine, any repeat calls to that specific number within the same 24 hour period are only made with the guaranteed presence of a live operator.
- This standard is a verbatim replication of the Ofcom guidelines[2] and as such would be considered by O_2 to be a regulatory requirement to be adhered to by any signatory to our contracts.

(1) Ofcom's latest policy decision is that this is not a "safe harbour" figure, i.e. a volume below this will not exempt someone from investigation and enforcement, it is merely a prioritisation threshold for their work

(2) http://stakeholders.ofcom.org.uk/consultations/silent-calls/statement/

Dialler operational standard

The O_2 Operational definition of dialler Traffic is as follows:

This is traffic which typically has

- An Average Length of Call ("ALOC") of under 30 seconds with
- An Answer Seize Ratio ("ASR") less than 60%.

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Dialler removal policy

O_2 voice quality of service department runs daily reports which will quickly identify traffic of this nature. The customer will be asked to stop sending this traffic but O_2 reserve the right to reduce their capacity or turned off to protect the integrity of the network.

Operational standard

O_2 reserve the right to limit (through call gapping or other operational intervention as we see fit in our sole discretion) which we feel may endanger the rest of the network. Any (but not limited to) of the following traffic patterns are not allowed:-

- Large amounts of call attempts hitting the same area or number type No one geographic dialling code should exceed 5 CPS unless previously agreed. Large, unexpected and unmanaged spikes of traffic cause network monitoring fault alarms and should be avoided.
- Time of day: dialler traffic will be the first type to be shed during any network faults or high traffic periods.
- Call attempts to a large percentage of unallocated numbers. ASRs below 40% will be deemed as suspect (e.g. data cleansing activities) and would probably be a breach of the Regulatory Standard above. Immediate action will be taken to limit such traffic.
- Each endpoint will have a defined calls per second limit. The endpoint must control their traffic within the agreed limits. Sending too many calls will get a 486 response

Maximum calls per second (CPS)

For security reasons, O_2 will also set limits for the maximum calls per second (CPS). The limits dependent on the endpoint design type.

- Single endpoint design 2 calls per second
- Multiple endpoint design 5 calls per second

If this constraint is reached, O_2 will log and reject calls with a SIP response 486.

CLI presentation

O_2 offers customers flexibility in what CLI they present

- That the allocated entity for the number being presented has authorized its use for this purpose.
- The number being presented is not one to a revenue sharing number that generates an excessive call charge. That means you cannot present 09 or 118 and that 070/076 are likely to be in breach.

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SIP Trunking calls

Call park, transfer and conferencing

The O_2 SIP Trunking Service provides the features listed below in the majority of cases but are not guaranteed on every platform connected to SIP Trunking because of vendor interoperability issues:

- Call Parking
- Call Transfer
- Conferencing

These features are supported via the SIP re-invite mechanism

Long duration calls

O_2 has a policy of terminating any call that exceeds eight hours.

Emergency call divert

The O_2 SIP Trunking service provides the facility to pre-configure call diverts for both individual numbers and DDI number ranges. Under failure conditions, the customer can log a ticket to activate either all pre-configured numbers with a single action, or activate individual diverts as necessary. Once activated, these diverts become effective. Deactivation is performed in the same manner as activation.

- The diverted destinations are subject to the same call barring option as the main SIP Trunking, e.g. if the SIP endpoint does not allow calls to mobiles, then the divert destination options will also exclude mobile numbers.
- The user will be billed for the diverted leg for all diverted calls. (*)
- Emergency call diverts are excluded from the fraud management service (FMS) and do not form part of the aggregated call spend for this purpose.
- Charges for emergency call diverts and associated calls will be chargeable
- O_2 does not support emergency diverts to international or other numbers which exceed 11 digits in length.
- SIP endpoints are constrained to a total maximum of 150 emergency call diverts configured at any point in time.
- The range of standard SIP Trunking endpoint features (e.g. fraud alerts, CLI flexibility, call admission control) do not apply to CLIs with emergency diverts enabled.

(*) Diverted calls for customers with inclusive minute bundles will not be billed for the diverted leg (subject to the associated fair usage policy).

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Emergency service calls

O_2 provide a VoIP service as defined by Ofcom, this 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 one of a number of national emergency call handling agents. The emergency personnel would need to confirm the identity and the actual location of the caller when they dial 999/112.

This emergency call handling agent may not be geographically the closest to the area code indicated by the calling CLI. The CLI presented will always be the site CLI, indicated as a VoIP service type from O_2, so that the emergency services operator will check the address details on the national database. It is the operators responsibility to ensure that the address associated with the default site CLI is always up to date. O_2 provide customers the tools to maintain these addresses.

O_2 and Ofcom expect that any calls originating on the O_2 network to emergency services will be presented with a CLI relating to the SIP service.

As a VoIP service, SIP Trunking may not be possible in the following circumstances:

- During a service outage where the end-customer loses connectivity for example, owing to a power outage or the failure of CPE routing equipment
- If a SIP endpoints account has been suspended

In such circumstances the customer should use their PSTN line to make the emergency call.

Short codes calls

The SIP Trunking service supports routing the following dialled short codes:

- 999 (Access to the Emergency services)
- 100 (Access to Operator Assistance)
- 101 (The national single non-emergency number for the Police Force)
- 111 (The national single non-emergency number for the NHS)
- 112 (Access to the Emergency services)
- 116 xxx (Harmonised Services of Social value)
- 118 (UK Directory enquiries)
- 123 (Access to Speaking Clock)
- 18000* to *18009 (Access to Voice Text Services for the Deaf)
- 195 (Access to Blind & Disabled Directory Enquiry Facilities)

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Fraud alert

The fraud management system (FMS) feature allows O_2 SIP Trunking customer to protect themselves from fraudulent activity from endpoints that have fallen victim to hacking or excessive unauthorized call spends. The feature allows customers to pre-set individual call limits against specific SIP endpoints and have automatic call barring invoked if these thresholds are breached.

The daily spend limit is a rolling 24 hour aggregation of call charges across all channels on that endpoint, the time starts when the endpoint is successfully commissioned as part of a new order or when the feature is configured and then 'saved' on an existing endpoint.

- The 24 hour clock will re-set if the threshold is breached and subsequently has the blocking removed.

The weekly spend limit is a rolling 7-day aggregation of call charges across all channels on that endpoint, the time starts when the endpoint is successfully commissioned as part of a new order or when the feature is configured and then 'saved' on an existing endpoint.

- The 7-day clock will re-set if the threshold is breached and subsequently has the blocking removed.

Call admission control

Through a process known as 'call admission control' (CAC), the maximum call limit of an endpoint defines its capacity for routing calls in the network. SIP Trunking customers pay a fixed monthly charge for the number of concurrent calls allowed on their endpoint.

Each endpoint will have 2 ports, one for outgoing and one for incoming the CAC limit will be allocated to both ports to allow maximum flexibility. Thus O_2 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 ingress and egress.
- Maximum ingress calls – specifies the maximum calls that may be placed from that endpoint to the O_2 network.
- Maximum egress calls – specifies the maximum number of calls that may be placed to that endpoint by O_2.

For example, if the channel limits is 100 concurrent calls, and there are 70 ingress calls, the maximum number of egress calls allowed will be 30.

In the case that the call control constraints are exceeded at a SBC, the invites will be rejected with either a SIP Response 486 or 503.

Number portability

Number portability for the SIP Trunking service is fully supported and tested. Number Porting changes are currently carried out manually, and it should be noted that BT will only port to a 'live' number; hence the O_2 SIP Trunking endpoints must be configured before the porting can take place.

Call barring

By default, calls to international and premium numbers will be barred, calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied. Customers are able to modify their profiles to allow or restrict access to;

- **No call barring:** allow calls to international, mobile (071-079), premium rate (09), personal numbers, special services up to 7 ppm (084), special services up to 13 ppm (087), directory enquiries calls barred (118) and including 01,02,03,08 and shortcodes.
- **International call barring:** call barring to international numbers. Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.
- **Mobile call barring**: call barring to mobile number (071-079). Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.
- **Premium number barring:** call barring to premium numbers (09). Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.
- **Personal number calls barred:** call barring to personal numbers (070). Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.
- **Special services calls up to 7 ppm:** call barring to special services calls up to 7 ppm(087). Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.
- **Special services calls up to 13 ppm:** call barring to special services calls up to 13 ppm(087). Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.
- **Directory enquiries calls barred (118):** call barring to directory enquiries(118). Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.
- **All call calling inc 01,02,03,08 and shortcodes:** call barring to 01,02,03,08 and shortcodes. Calls to the emergency services 999, 112 remain unaffected irrespective of the barring applied.

FTTC considerations

The following considerations should be made when using FFTC as the connectivity option for SIP Trunking services.

- FTTC ethernet requires a copper WLR3 line to be provided, the WLR3 service must be ordered and installed before an FTTC ethernet order can be placed using the CLI number.
- Charges for this WLR3 are not included in the FTTC ethernet quote.
- WLR3 lines should be ordered WLR3 care level 4 to ensure any faults are resolved in the quickest possible manner.
- The FTTC ethernet service offers guaranteed symmetrical bandwidth of up to 20Mbps. Any downstream bandwidth available and purchased above this will be provided as best efforts and subject to network congestion.

SIP endpoint

SIP endpoints ID

The following naming convention is used for identifying endpoints on the O_2 network: Unique endpoint name + suffix e.g. DC2NYYABC1234_L1

The Suffix differentiates the endpoint design type i.e.

- _L1: A Loadshare endpoint
- _A1: An Active endpoint in an active / standby design
- _S1: A Standby endpoint in an active / standby design
- _R1: A Resilience+ endpoint

The suffix will increment in the case of multiple loadshare or standby endpoints associated with the same design.

Customer premises equipment

To ensure compatibility of equipment and ease of installation, O_2 are continually undertaking conformance testing with equipment vendors.

If connection is required for a device that has not previously been connected to O_2, the customer can request O_2's cooperation with conformance testing to ensure that the device is fully compatible with the O_2 network prior to provisioning.

Network security

Access to the O_2 SIP Trunking Service from the CPE is 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 (SBC).

Codecs

Voice encoding can be G.711 A-law or G.729-A with sample periods of 20 ms.

- O_2 do not support the use of video codecs and customers should make every effort to ensure that no video codecs are included in any SIP requests.
- O_2 polices the media-stream bandwidth based on the negotiated codec. If a customer exceeds the bandwidth for a specific codec, RTP packets will be discarded and this will result in poor voice quality.
- Silence Suppression and Comfort Noise: O_2 support for silence suppression and comfort noise is available to Microsoft Skype for Business connections only.
- Ptime:- O_2 do not support the negotiation of codec Ptime.

Channel bandwidth

The table below gives an estimate of the bandwidth requirements for VoIP calls using G.711, and G.729a, note that sample periods of 20 ms are supported. The table below is minium bandwidth consumption over ethernet, per channel

Codec	Sample Period	Encoded bandwidth	IP /UDP / RTP overhead	Ethernet overhead	Total bandwidth
G.711	20 ms	64 kbps	16 kbps	15.2 kbps	95.2 kbps
G.729	20 ms	8 kbps	16 kbps	15.2 kbps	39.2 kbps

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Fax and DTMF support

The O_2 SIP Trunking Service will support Fax and Modem transmission subject to the following constraints

- FAX and Modem transport in band using G.711 a-law codec is supported. Renegotiation to T.38 is supported (subject to interoperability testing).
- The use of G729 for in-band faxes is not supported as its compressed nature may cause tones and messages to be lost.
- If the fax option for an endpoint is set to T.38 enabled then:
    - The O_2 network will attempt to re-negotiate [re-INVITE] to T.38 for fax calls for ingress (Customer to O_2) calls on detection of a fax tone.
    - The O_2 network will accept a re-negotiation [re-INVITE] to T.38 for fax calls for egress (O_2 to Customer) calls.
    - The re-negotiation must be done using the re-INVITE mechanism after answer.
- Due to different vendor implementations O_2 cannot guarantee T.38 interoperability with all vendors and will not accept responsibility if T.38 interoperability cannot be achieved with a specific vendors' implementation.

If the fax option for an endpoint is set to T.38 disabled, then:

- All fax calls will be handled as G.711 pass through providing the customer has a G.711 codec available in his media profile.
- If calls are made to another CP that does not support the method of transport for the tones, O_2 will not perform any form of inter-working between the two different methods.
- The following methods will be supported to transport DTMF tones:
- The O_2 core network will support the generation of 'In-band' or 'RFC2833' DTMF transport based on end to end negotiation.
- 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 over G.711 codec only
- If a G729 codec is being used then DTMF tones should not be sent in-band, O_2 will not guarantee the delivery of in-band DTMF over a G729 codec.

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IP addressing and DNS

IP version 4.0 is supported. IP Version 6.0 is not supported.

In terms of the customer interaction with O_2's network, DNS capability, including SRV and A record look up is not supported

CLI presentation

CLI flexibility

O_2 support network number CLI presentation and privacy definitions in accordance to RFCs 3323 and 3325, using P-asserted ID and Privacy Headers only. O_2 require customers to supply the PAID and, where required, the privacy headers, O_2 do not support the use of the invalid statute Remote Party ID (RPID) definitions and such Headers will be removed.

If the CPE connected to the O_2 network presents a geographic number in the UK national format, the O_2 network will pass these details as the A-number CLI into the PSTN or mobile network. This outbound presentation will be supported by default if the number presented is as follows:

- A number in the UK national format without a leading zero presented by the customer premises equipment (CPE) as the A-number.
- An O_2 provided Geographic Number that is allocated to the Endpoint at order creation
- An O_2 provided geographic number that is allocated to the SIP endpoint at a later date via a customer change request
- A geographic number that is ported from another carrier to the O_2 network
- CLI flexibility is enabled on the SIP endpoint.

The A-number is checked against a database on the O_2 network of geographic numbers that are allocated to the O_2 SIP Trunking endpoint.

If CLI flexibility is disabled and the number presented does not meet the above criteria, the A-number CLI presented will be a default CLI, which by default is the first number in the O_2 allocated geographic DDI range.

O_2 cannot guarantee consistent presentation of intended CLIs for calls made to mobile carriers as successful presentation of the intended CLI is entirely dependent on the mobile carriers use of these numbers and specific call flow.

Mobile missed calls and voicemail notifications can often use the default CLI – the underlying network CLI (PAID CLI) - which is the customer selected default number or the first number in the O_2 allocated account range, rather than the intended CLI for presentation.

PAID header

The format of the PAID number can either be national significant (with or without a leading zero), or in full e.164 format. If no PAID header is provided, a PAID header with the default network CLI will be inserted.

Examples:

- P-Asserted-Identity: <sip:+441618777148@100.100.100.29>
- P-Asserted-Identity: <sip:01618777148@100.100.100.29>
- P-Asserted-Identity: <sip:1618777148@100.100.100.29>

Presentation CLI (A-number)

A-numbers (SIP FROM, P-sssserted-ID) should be presented in E.164 format, i.e. +441611234567. The A-number is validated by the SBC and if it is not in the O_2 range it is overwritten with an agreed network CLI from the customer's O_2 allocated range. It is a requirement of the SIP Trunking service that the calling party (A-number) be validated to confirm the format and ensure that the number is owned by O_2, so that the emergency services have an accurate record of the calling customer.

If the CPE connected to the O_2 network presents a geographic number, the O_2 Network 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 are presented by the customer premises equipment (CPE) in the SIP FROM and P-Asserted-ID fields as E.164 format
- It is an O_2 provided geographic number that is allocated to the endpoint at order creation
- An O_2 provided Geographic Number that is allocated to the Endpoint at a later date via a Customer Change Request
- A Geographic number that is ported from another carrier to the O_2 network

If the number presented does not meet the above criteria, the A-Number CLI that will be presented will be a default CLI, which is the first number in the O_2 allocated Geographic DDI range.

For ingress calls, A-numbers are sent to customers as received by O_2 from other network operators.

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Network CLI (A-number)

Every endpoint must have at least one CLI from the O_2 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 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.

O_2 supports both network and presentation CLIs. For calls from the customer to O_2 where the call terminates on the PSTN, the SIP FROM field is mapped to the presentation CLI and the SIP P-Asserted-ID is mapped to the network CLI.

To ensure the correct Network CLI is passed into the O_2 network and then forwarded to the PSTN O_2 will insert the customer Network CLI Number in E.164 format into the PA-ID, replacing any value received from the customer CPE.

For calls from the PSTN to the O_2 SIP Trunking endpoint, the SIP FROM field will contain the Presentation Number when available and the P-Asserted-ID field will contain the Network Number when available. If only the Network Number is available, then this will be mapped to both the SIP FROM field and the P-Asserted-ID field.

B-numbers

B-numbers should be sent to O_2 in the following format:

- UK National 0 NSN (National Significant Number) 44 NSN, +44 NSN and 0044 NSN.
- International 00 CC NSN and +CC NSN (+CC NSN format is offered on limited connections currently, check with support).
- Service and emergency calls no leading 0(s), + or CC (Country Code) to be used.

As a default configuration B-numbers will be presented to the customer including a leading 0, (0 NSN); however, this is flexible, customers can request that a prefix be inserted, or that their number range include a country code.

Calling line restriction (CLIR)

When the calling (A-Party) has requested privacy (CLIR), O_2 will enforce privacy in accordance with RFC3325. Calling party information, including From Address, Contact and associated Privacy Header (PAID) are withheld from an endpoint. For calls from the customer to O_2, the customer must indicate that CLI is to be withheld by the following mechanism:

- RFC3261 Section 8.1.1.3 and 20.20 which describes the use of an “Anonymous” display field to the From: header to indicate that the client is requesting privacy.
- SIP Privacy which is described by RFC 3323 and RFC 3325
- For calls from O_2 to the customer CPE: the from address is set to - “Anonymous”<sip: anonymous@anonymous.invalid> as per RFC 3323, the contact header is set to anonymous and the privacy header (P-Asserted-identity RFC 3325) is removed from the outbound INVITE to the SIP Trunking endpoint to provide true CLI restriction.

If the Privacy header contains only “id,” or only “id” and “critical” values, O_2 removes the privacy header completely.

The Contact user part is changed to "anonymous."

Example:

- Public Side Invite; (Going out to SIP Trunking Endpoint)
- From: Anonymous <sip:anonymous@anonymous.invalid>;tag=3541226365-699915
- Contact: <sip:anonymous@83.245.6.117:5060>

Note: NO PAID Header is provided for anonymous inbound calls. It is removed to provide the calling party full anonymity.

SIP signalling and failover

Session failover and endpoint resilience

The following scenarios will result in an endpoint as being tagged as 'out of service'. In the case of resilient designs these scenarios initiate failover to alternative sites.

- Destination Unreachable (ICMP unreachable response)
- SIP ping failure
- SIP failure response code

Destination unreachable

This happens in two ways:

- O_2 receives an ICMP unreachable message in response to the INVITE message that it sends out to the endpoint. This could indicate that there is no network route to that destination (i.e. the access method has failed) or the destination is temporarily out of service.
- Outgoing INVITEs are retransmitted from O_2 3 times. If that limit is reached, O_2 will stop trying that endpoint and initiate failover to another endpoint.

In the case of resilient designs, failover is initiated when O_2 concludes that a SIP Trunking endpoint cannot be reached.

Prevention of session loss

In order to minimise the impact of failure of network components, it is recommended that within the CPE session timers, are as specified by RFC 4028. The preferred method to request a change of the refresh time is by means of a SIP error response 422 or a Re-INVITE.

To avoid a high volume of 'Invite-422-ReInvite' iterations at the start of the call, where the 'session- expires' value in the originating Invite is less than 1800 seconds (as per RFC 4028), it is recommended this value should not be less than 600 seconds. This will not prevent 'session interval too small' responses entirely and it is highly recommended that the advertise support of the 'timer' feature enable their call-servers to resend the 'invite' request with the new Session-Expires value upon receipt of a 422 response.

The session refresh time cannot be negotiated by means of UPDATE. The session can be refreshed by means of a Re-INVITE or an UPDATE.

RFC support

The O_2 SIP Trunking service supports the following RFC's. Full compliance is subject to differences in interpretation, interoperability constraints and the exceptions noted below.

O_2 SIP Trunking reference RFCs and other Standards

RFC	Supported
RFC 2327 Session description protocol	Yes
RFC2543 SIP: Session initiation protocol	Yes, putting media streams on hold, Indicate the IP in SDP message when call-hold mechanism is used according RFC 2543: O_2 will support receiving either the actual IP or 0.0.0.0 if interworking to TDM. If the incoming/outgoing route is via a SIP interconnect partner this may not be the case.
RFC 2833 - RTP payload for DTMF Digits, telephony tones and telephony signals	Yes
SIP RFC 3261- SIP messages, headers and protocol	Yes
RFC 3262 - Reliability of provisional responses in SIP:	Yes
RFC 3264 - An offer/answer model with SDP:	Yes
SIP RFC 3311 - SIP update method	Yes (only during unconfirmed dialogue). Re-INVITE should be used when dialogue is in confirmed state.
SIP RFC 3323 - Privacy mechanism for SIP	Yes
SIP RFC 3325 - Privacy extensions to SIP for asserted identity within trusted networks	Yes
SIP RFC 3326 - Reason-header Field for SIP	Yes
RFC 3398 - ISDN to SIP mapping:	Yes
RFC 3551 - RTP profile for audio:	Partial, O_2 will only support G711, G729 and G729A codecs
RFC 4028 session timers in the session initiation protocol	Yes
NICC ND1017:2006/07. TSG/SPEC/017. interworking between session initiation protocol (SIP) and UK ISDN user part (UK ISUP)	Yes

Supported SIP methods

Methods	Extent of support	Notes
ACK	Full, receive and transmit	
BYE	Full, receive and transmit	The Reason-header field in BYE or CANCEL message is preferred but not essential
CANCEL	Full, receive and transmit	The Reason-Header field in BYE or CANCEL message is preferred but not essential
INVITE	Full, receive and transmit	
OPTIONS	Full, receive and transmit	
PRACK	Full, receive and transmit	

Supported SIP responses

Responses	Extent of support	Notes
100 TRYING	Full, receive and transmit	
180 RINGING	Full, receive and transmit	
181 Forwarded	Minimal, receive only	
183 Session progress	Full, receive and transmit	In cases of TDM interworking then a 183 session progress will be sent when the outgoing MGW is seized regardless of the subscriber state.
200 OK	Full, receive and transmit	O_2 will include SDP in 200-Ok answer message which will be the same as any SDP previously sent in a 18x message
400 Bad request	Full, receive and transmit	
403 Forbidden	Full, receive and transmit	
404 Not found	Full, receive and transmit	
405 Method not allowed	Minimal, receive only	
406 Not acceptable	Minimal, receive only	

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Responses	Extent of support	Notes
408 Request timeout	Minimal, receive only	
415 Unsupported media type	Full, receive and transmit	
422 Session timer interval too small	Full	
480 Temporarily unavailable	Full, receive and transmit	
481 Call leg / transaction does not exist	Full, receive and transmit	
482 Loop detected	Full, receive and transmit	
483 Too many hops	Full, receive and transmit	
484 Address incomplete	Full, receive and transmit	
486 Busy here	Full, receive and transmit	
487 Request terminated	Full, receive and transmit	
488 Not acceptable here	Full, receive and transmit	
491 Request pending	Full, receive and transmit	
500 Internal error	Full, receive and transmit	
501 Not implemented	Full, receive and transmit	
502 Bad gateway	Minimal, receive only	
503 Service unavailable	Full, receive and transmit	
504 Server time-out	Full, receive and transmit	
600 Busy everywhere	Minimal, receive only	
603 Decline	Minimal, receive only	
604 Does not exist	Minimal, receive only	
606 Not acceptable	Minimal, receive only	

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Hunt-able SIP responses

The following SIP responses when sent back from site A (Active/Standby) or either site (Load Balanced), will cause the network to hunt to the next available SIP endpoint.

SIP Status Code received	Cause/ CNA/ Q850	Hunt-able SIP response
400 (Bad request)	NONE	Yes
401 (Unauthorised)	NONE	
402 (Payment required)	NONE	
403 (Forbidden)	NONE	
404 (Not found)	NONE	
404 (Not found)	1 (Unallocated number)	
405 (Method not allowed)	NONE	
406 (Not accepted)	NONE	
407 (Proxy auth. required)	NONE	
408 (Request timeout)	NONE	Yes
410 (Gone)	NONE	
413 (Req.entity too long)	NONE	
414 (Req. URI too long)	NONE	
415 (Unsupported media type)	NONE	
416 (Unsupported URI scheme)	NONE	
420 (Bad extension)	NONE	
421 (Extension required)	NONE	
423 (Interval too brief)	NONE	
480 (Temporarily unavailable)	NONE	
480 (Temporarily unavailable)	34 (No channel available)	
480 (Temporarily unavailable)	41 (Temporary failure)	Yes (with q.850 reason header containing reason 41)
480 (Temporarily unavailable)	42 (Switching equip. cong)	
481 (Call / transaction not exist)	NONE	Yes
482 (Loop detected)	NONE	
483 (Too many hops)	NONE	

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SIP Status Code received	Cause/ CNA/ Q850	Hunt-able SIP response
484 (Address incomplete)	NONE	
484 (Address incomplete)	28 (Address incomplete)	
485 (Ambiguous)	NONE	
486 (User busy)	17 (User busy)	
486 (User busy)	NONE	
487 (Request terminated)	NONE	
488 (Not acceptable here)	NONE	
491 (Request pending)	NONE	
493 (Undecipherable)	NONE	
500 (Server Int. error)	NONE	Yes (with q.850 reason header containing reason 41)
500 (Server Int. error)	34 (No channel available)	
500 (Server Int. error)	41 (Temporary failure)	Yes (with q.850 reason header containing reason 41)
500 (Server Int. error)	42 (Switching Equip. cong)	
502 (Bad Gateway)	NONE	Yes
503 (Service unavailable)	NONE	Yes (with q.850 reason header containing reason 41)
503 (Service unavailable)	41 (Temp fail)	Yes (with q.850 reason header containing reason 41)
504 (Server timeout)	NONE	Yes
600 (Busy everywhere)	NONE	
603 (Decline)	NONE	
604 (Does not exist anywhere)	NONE	
606 (Not acceptable)	NONE	

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Connectivity incident priority definitions

O_2 will use its reasonable endeavours to resolve incidents relating to the connectivity in accordance with the applicable target service levels.

Priority	Definition	Update frequency	Resolution
1	Customer's data access network is down causing critical impact to business operations if service is not restored quickly. Example of scale can be; Single critical Site or multiple none critical Sites down. No workaround is available	Every hour	6 hours Ethernet 8 hours EFM/FTTC
2	Customer's data access network is severely degraded impacting significant aspects of business operations unavailability Example of scale can be: Single none critical Site down or degradation to single critical Site. No workaround is available	Every two hour	12 working hours
3	Customer's data access network performance is degraded. Data Access Network functionality is noticeably impaired but most business operations continue. Example of scale can be: Service unavailability affecting a single user or general minor impact for multiple users. No workaround is available	Provided through ticket updates as information becomes available	3 working days
4	Individual User Data Access Network performance is degraded. Network/ service functionality is noticeably impaired but most business operations continue. Scale can be: Minor problem which affects the usability of the service, and where a mutually agreed workaround is generally available which causes minor disruption to the Customer.	Provided through ticket updates as information becomes available	5 working days

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Customer service charter

Service levels overview

SIP service availability

The service level targets for service availability of the SIP Trunking Service depend on whether or not the SIP Trunking has a resilient build. The service availability applies to the following; the infrastructure, transmission equipment and core network, the service that supports call routing and termination, excluding the data connectivity.

SIP Option	Availability	Supported hours
SIP over PUBLIC	99.95%	24 x 7 x 365
SIP over PUBLIC (loadshare)	99.99% resilient build	24 x 7 x 365
SIP over PUBLIC (active - standby)	99.99% resilient build	24 x 7 x 365
SIP over PUBLIC (resilience+)	99.99% resilient build	24 x 7 x 365
SIP over JANET (Single)	99.95%	24 x 7 x 365
SIP over JANET (loadshare)	99.99% resilient build	24 x 7 x 365
SIP over JANET (active - standby)	99.99% resilient build	24 x 7 x 365
SIP over JANET (resilience+)	99.99% resilient build	24 x 7 x 365
Single SIP over ethernet FIBRE	99.95%	24 x 7 x 365
Single SIP over ethernet EFM	99.95%	24 x 7 x 365
Single SIP over ethernet FTTC	99.95%	24 x 7 x 365
2 SIP endpoints loadshare	99.99% resilient build	24 x 7 x 365
2 SIP endpoints active-standby	99.99% resilient build	24 x 7 x 365
2 SIP endpoints resilience+	99.99% resilient build	24 x 7 x 365
3 SIP endpoints loadshare	99.99% resilient build	24 x 7 x 365
3 SIP endpoints active-standby-standby	99.99% resilient build	24 x 7 x 365
4 SIP endpoints loadshare	99.99% resilient build	24 x 7 x 365

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Number porting timescales

Timescales below are based on a 'subsequent port' meaning the Losing communication provider (CP) and the range holder are different for the number porting range.

Subsequent Port	O_2 Total Lead-time working days	Time to complete	Standard Trigger time	Slot Options	Out of hours
Single line	13	up to 2 hours	10 am	8am 9am 10am 11am 12 noon	Weekday: Monday-Friday 1600-2200 Weekend: Saturday 0800-1600.
Multi line including associated & other numbers	15				
Multi line (150 lines or less)	18			09:00 - 12:00 10:00 - 13:00 11:00 - 14:00 12:00 - 15:00	Weekday: Monday-Friday 1600-2200 Weekend: Saturday 0800-1600
Multi line (151 lines or more)	25				
Complex DDI & BT FeatureNet	30				
Geographic IPEX port	Best endeavours				
Geographic Mixed Operator Port	Best endeavours				

SIP Trunking incident priority definitions

O_2 will use its reasonable endeavours to resolve incidents relating to the SIP Trunking Service in accordance with the applicable target service levels.

Priority	Definition	Update frequency	Resolution
1	Customer's SIP Trunking service is unavailable with complete loss of service causing a critical impact to the customers' business operations. Immediate restoration of service is expected. Example of scale would be the complete loss of SIP Trunking resulting in an entire customer site being unable to make or receive telephone calls.	Every hour	6 hours
2	Customer's SIP Trunking service is available but operating with reduced functionality or degraded service that is causing significant business impacts to the end user. Example of scale would be consistently intermittent ability to use voice services due to poor quality or dropped calls	Every two hour	12 hours
3	Customer's SIP Trunking service is available but an issue is causing reduced functionality or degraded service but which does not cause a significant business impact to the end user Example of scale would be CLI presentation being inhibited, unable to send FAX	Provided through ticket updates as information becomes available	3 days
4	Customer's SIP Trunking service is available but an issue has been raised which impacts an individual user in a minor capacity but does not impact the customers' business operations. Example of scale can be an issue which effects the usability for an end-user but where an agreed workaround is available that causes minimal disruption to the customer.	Provided through ticket updates as information becomes available	No resolution target

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Logging an incident

- If you've purchased an O_2 Manage service you will log an incident with the O_2 Business Service Operations (BSO) team
- If you have purchased a standalone O_2 SIP Trunking service you will log an incident with the Connected for Business (C4B) team.

What is a 'managed services customer'?

If one of the following services are taken in conjunction with SIP Trunking you are defined as 'managed services customer' and incidents are raised with the BSO team.

Mandatory information

- SIP endpoint ID for the affected service.
- Reference to historical existing Problem, Change and Request number (if applicable)?
- Company and business unit name (if applicable).
- Location of incident.
  - Any site access restrictions (security requirements, access hours etc.)
  - Contact name, telephone number (if different to above) e.g. site contact details
  - Contact email address
  - Brief description of fault/incident.
  - Any recent changes to the network / affected area (where known).
  - Impact to the business (to ensure call is correctly prioritized).
- Contact method
- Where applicable
  - Details of how to recreate the incident.
  - Details of any recent changes or amendments.
  - Screen shots.
  - Site access requirements.
  - Alternate contact details of person logging incident.
- Points to consider:
  - What were you trying to achieve when the incident occurred?
  - Was an error message displayed?
  - Have you completed this procedure successfully in the past?
  - Has anything changed since then?
  - Is there any diagnostic information? If so, please keep this available for investigation.
- Documentation:

All or any other relevant documentation applicable to the specific Incident

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BSO contact details

Contact details for managed services customers.

BSO service desk function	Contact details
Service desk – telephone	0800 652 3123
Service desk – email	O2unifydesk@O2.com
Service desk – SNOW portal	https://unify.service-now.com/unify/

Once the required information has been supplied and entered in to the O_2 Service Management systems, the appropriate priority will be applied and a unique reference number allocated to the incident record and provided to the customer via email/telephone.

All incidents logged via the web portal or email will be assigned the status of 'work in progress within 4 hours

C4B contact details

Contact details for SIP Trunking customers.

C4B service desk function	Contact details
Service desk – telephone	0844 463 2617
Service desk – email	connectedworldtst@o2.com

Once the required information has been supplied and entered in to the O_2 Service Management systems, the appropriate priority will be applied and a unique reference number allocated to the incident record and provided to the customer via email/telephone.

All incidents logged via the web portal or email will be assigned the status of 'work in progress within 4 hours.

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Escalation management

BSO escalation

Business solution operations team - available 24 hours a day, 7 days a week, 365 days a year

Escalation	Name	Contact details
1st	Service Desk	0800 6523123
2nd	Service Desk Major Incident Manager	0800 6523123
3rd Out of hours	Head of BSO Operations – Jon Miles	07584 702752

C4B escalation

Connected 4 business team - available 24 hours a day, 7 days a week, 365 days a year

Escalation	Name	Contact details
1st	Faults team	TEL : 08444632617 connectedworldtst@o2.com
2nd	Duty manager (8am-9pm, 7 days)	TEL : 07801105354 c4bmanagers@o2.com and c4bescalations@o2.com
2nd	Duty manager (8pm-8am, 7 days)	TEL:07801105354 or 07713787576 claire.douglas@o2.com and senga.chapman@capita.co.uk
3rd	Khail Rashid (first line C4B advisors) Clare Rawson	TEL :07710382385 khalil.rashid@o2.com TEL : 07730523784 claire.rawson@o2.com
Out of hours	Dawn Saxton	dawn.saxton@capita.com

In life changes

To request an in life change the customer need to complete a customer request form (CRF) and submit it to unifyorderfulfilment@o2.com. All order forms are available at www.o2.co.uk/business/sip-trunking

I need help with	What CRF do I submit to Unify order fulfilment	What is the target lead time?
Can I add new number(s) to my SIP endpoint? Yes, you can add UK based geographic numbers ranges to your existing SIP endpoint.	SIP New number	2.5 working days
Can I port number(s) to my SIP endpoint? Yes, you can port your UK based geographic number ranges to your existing SIP endpoint. Subject to porting agreements.	SIP Port a number	Please see porting timescales
Can I cease my SIP endpoint? Yes, your can request a SIP endpoint cease. Please note cancellation charges may apply.	SIP ceased	2.5 working days
Can I add or decrease my channels on my SIP endpoint? Yes, your can request additional channels or decreasing of channels subject to contract agreement	SIP channels	[4] Decreased channel 2.5 working days [1][2][3][4] Increase channels 2.5 working days
Can I request CLI flexibility? Yes, CLI flexibility allows you to present non O_2 registered CLIs as the presentation A-number CLI. **Can I change the CLI presentation to the SIP endpoint?** Yes, CLI presentation changes to the incoming CLI format can be made (leading zero 0203*, no leading zero 203* and E.164 format +44203*).	SIP CLI changes	2.5 working days

[1] If SIP Trunking option is SIP over Public or SIP over Janet, then the target lead time will be 2.5 working days

[2] If SIP Trunking has available capacity on Ethernet connectivity, then the target lead time will be 2.5 working days

[3] Channel increases above 30% or where additional bandwidth is required will incur a lead-time (18.5 working days) for core network capacity checks.

[4] For Semafone enabled SIP Trunking the target lead time will be an extra 5 working days. For decreasing/increasing SIP channels additional time will be required to allow for verification with Semafone

I need help with	What CRF do I submit to Unify order fulfilment	What is the target lead time?
Can I set a SIP Trunking call diverts? Yes, predefined diverts for a single number or DDI range can be pre configured to your SIP endpoint. Divert activation and deactivation is implemented by ticket request.	SIP Trunking divert	2.5 working days
What is a letter of authority? When porting a geographical number range a letter of authority (LOA) is required. The letter of authority isn't required to place an order, however the range holder or losing communications provider can, and do, request a letter of authority from time to time.	SIP LOA	not applicable
Why update the 999 records for my SIP Trunking numbers? As a fully compliant and regulated telecoms provider O_2 support the emergency services database. O_2 can update the emergency services database with corrected or updated address information for allocated numbers to your SIP endpoint.	SIP 999 change	2.5 working days

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I need help with	What CRF do I submit to Unify order fulfilment	What is the target lead time?
Can I change the IP address of IP-PBX or session border controller? Yes, this is possible for SIP Trunking services please completed the CRF. **Can I change my fraud management setting on the SIP endpoint?** Yes, fraud management setting allows protections from fraudulent activity from the SIP endpoint that have fallen victim to hacking or excessive unauthorized call spends. The feature allows to pre-set individual call limits. **Can I change the call barring options of the SIP endpoint?** O_2 offers five call barring options, please complete the CRF to change the call barring settings. No call barring, international call barring, mobile call barring, premium call barring and all call barring (expect freephone) **What over features can I change to my SIP endpoint?** The following technical parameters can be changed on the SIP endpoint the supported codec , enabling or disabling Fax T.38 support. **Can I remove a number range for a SIP endpoint?** Yes, a number range can be removed from a SIP endpoint, please note this option will completely remove the number range and this range will be returned to the original range holder. **Can I upgrade from standard to enhanced SIP Trunking channels ?** Yes, an enhanced build has additional network resilience this can be upgrade from a single SIP Trunking channel.	SIP endpoint change	2.5 working days

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O_2 business

Current pricing

- The current voice tariff can be viewed from the following location
- The current new DDI and porting charges can viewed from the following location

Glossary

This appendix provides a basic glossary for some of the terms, acronyms, and abbreviations used in this document.

Term	Description	Term	Description
ANI	Automatic number identification	IPT	Internet telephony
ASR	Answer seizure ratio	L2TP	Layer 2 tunnelling protocol
ATM	Asynchronous transfer mode	LAN	Local area network
BAU	Business as usual	MOS	Mean opinion score
CAC	Call admission control	NSN	National significant number
CDR	Call detail record	OSS	Operational support systems
CLI	Calling line identity	PABX	Private automatic branch exchange
CLIP	CLI presentation	PDD	Post-dial delay
CLIR	CLI restriction	POP	Point of presence
CPE	Customer premise equipment	PSTN	Public switched telephone network
CPS	Calls per second. The maximum number of new call attempts per second.	RTP	Real-time transport protocol
DTMF	Dual tone multi-frequency	SBC	Session border controller
EDD	Ethernet demarcation device	SDH	Synchronous digital hierarchy
G.711	ITU recommendation for compounding digital audio	SIP	Session initiation protocol. A signalling protocol for Internet conferencing, telephony, presence, events notification and instant messaging.
G.729	Audio data compression algorithm	SLA	Service level agreement
H.323	ITU-T VoIP protocol	UDP	User datagram protocol
HA	High availability	VoIP	Voice over IP
IP	Internet protocol (shorthand for TCP/IP)	VPN	Virtual private network
IP PBX	IP Private branch exchange		

Document version history

Document	Description	Version
SIP Product handbook	Draft release	5th May 2018
SIP Product handbook	Updated draft changes and topology diagrams	10th May 2018
SIP Product handbook	Diagram updates and minor changes	11th May 2018
SIP Product handbook	Updated channel upgrades	18th May 2018
SIP Product handbook	Added draft service charter for internal review	21st May 2018
SIP Product handbook	Added dialler policy	22nd May 2018
SIP Product handbook	Updated service charter and minor changes	4th June 2018
SIP Product handbook	Removed connectivity timescales	5th June 2018
SIP Product handbook	Product name change to SIP Trunking	8th June 2018
SIP Product handbook	Removed single SIP connectivity resilience options	19th June 2018
SIP Product handbook	Updated based on brand review	22nd June 2018
SIP Product handbook	Target lead times added to in life changes	28th June 2018
SIP Product handbook	Added voice policy details	5th July 2018
SIP Product handbook	Added IPEX port lead-time	6th July 2018
SIP Product handbook	Updated C4B escalation in customer charter	20th July 2018
SIP Product handbook	Added additional internet options	8th August 2018
SIP Product handbook	Current voice tariff, new DDI and porting charges added	10th October 2018
SIP Product handbook	Updated Emergency restore section in number porting	12th October 2018
SIP Product handbook	Updated new voice tariffs including ppm	15th November 2018
SIP Product handbook	Added website link	26th November 2018
SIP Product handbook	Minor updates	16th January 2019
SIP Product handbook	Updated URL links to website	26th January 2019
SIP Product handbook	Added Semafone capability	21st April 2019
SIP Product handbook	Updated out of hours porting details	15th July 2019

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