Telefonica

O2 business

SIP Trunking

SIP over O₂ Gateway

Company name(1)			
Contract type(2)	Please select	Quote reference(3)	

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.

Managed acceptance testing(4)	Please select	Managed acceptance testing includes service acceptance testing.	
No of SIP endpoints(5)	Please select		
Endpoint build option(9)	Please select		
Contact name(6)		Contact number(6)	
Email address(6)		Job title(6)	

Semafone enabled SIP Trunking If 'Semafone enabled SIP Trunking' please confirm the following details.

Deployment type(100)	Please select	Semafone build(103)	Please select
Contact name(101)		Email address(101)	
Contact number(101)		3rd party(102)	

Instructions:

This order form is for a new SIP Trunking over O_2 Gateway order. This order form is submitted with an O_2 Gateway Ethernet order form to your O_2 account team. An order form guide is also available from the following location. Addtional order forms flavours are available at www.o2.co.uk/business/sip-trunking.
To request additional new number(s) or the porting of existing numbers with this order form please also complete.

- New number customer request form
- Port a number customer request form
- This order form includes the request for one DDI range, additional DDI ranges or be-spoke number selection can be requested via the following customer request form
- If a DDI range isn't requested on this order form a single DDI will be provisioned for test purposes.
- SIP header format can be view at the following location.
- Semafone enabled SIP Trunking is not available in all markets.

SIP Trunking July 2019 version 2.1

Telefonica

O2 business

SIP Trunking

SIP over O_2 Gateway

SIP address

Please confirm the business address details via https://www.royalmail.com/find-a-postcode.
End user details, where possible please provide a 24 hour contact

Endpoint 01

Premise(7)		End User first name(8)	
Street/throughfare(7)		End User last name(8)	
Town/locality(7)		Job title(8)	
		Telephone(8)	
Country code(7)	GB	Mobile number(8)	
Postcode(7)		Email(8)	

Endpoint 02

Premise(7)		End User first name(8)	
Street/throughfare(7)		End User last name(8)	
Town/locality(7)		Job title(8)	
		Telephone(8)	
Country code(7)	GB	Mobile number(8)	
Postcode(7)		Email(8)	

Endpoint 03

Premise(7)		End User first name(8)	
Street/throughfare(7)		End User last name(8)	
Town/locality(7)		Job title(8)	
		Telephone(8)	
Country code(7)	GB	Mobile number(8)	
Postcode(7)		Email(8)	

Endpoint 04

Premise(7)		End User first name(8)	
Street/throughfare(7)		End User last name(8)	
Town/locality(7)		Job title(8)	
		Telephone(8)	
Country code(7)	GB	Mobile number(8)	
Postcode(7)		Email(8)	

SIP Trunking July 2019 version 2.1

SIP Trunking

SIP over O_2 Gateway

Endpoint details

O_2 will allocate the following IP address ranges in the form fields Endpoint IP address (13) and Network SBC IP address (100) are completed internally within O_2

Endpoint build option(9)	Please select	IP addressing Range(12)	O2 Gateway IP Address
Build Option(11)	Please select		
Endpoint 01 IP address(13)		Network SBC 01 IP address (100)	
Endpoint 02 IP address(13)		Network SBC 02 IP address (100)	
Endpoint 03 IP address(13)		Network SBC 03 IP address (100)	
Endpoint 04 IP address(13)		Network SBC 04 IP address (100)	

Endpoint hardware

If your endpoint hardware option isn't included within dropdown (14) please complete the bespoke CPE section.

Endpoint 01 hardware(14)	Please select	Bespoke CPE type(15)	
Endpoint 02 hardware(14)	Please select	Bespoke CPE vendor(15)	
Endpoint 03 hardware(14)	Please select	Bespoke CPE model(15)	
Endpoint 04 hardware(14)	Please select	Bespoke software version(15)	

Optional VLAN presentation

If a specific VLAN ID is required for presentation on the O_2 Gateway CPE LAN port to support an existing network environment please indicate below. If no VLAN ID is provided O_2 will automatically generate this.

Endpoint 01 VLAN ID(101)	
Endpoint 02 VLAN ID(101)	
Endpoint 03 VLAN ID(101)	
Endpoint 04 VLAN ID(101)	

SIP Trunking July 2019 version 2.1

Telefonica

O2 business

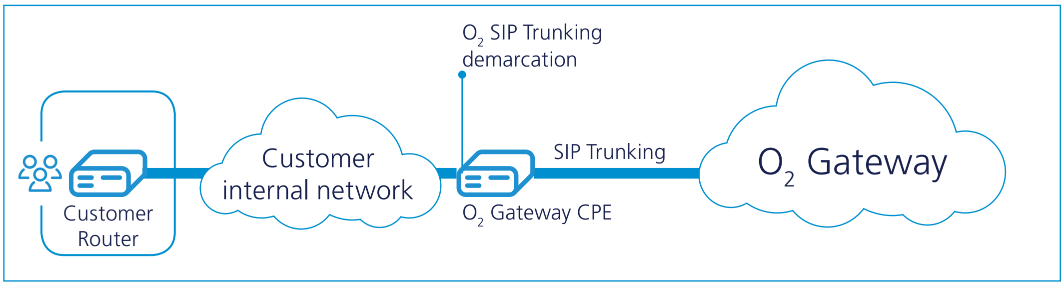
SIP Trunking

SIP over O_2 Gateway

Optional routing

If the customer SBC doesn't directly connect to the O_2 Gateway CPE. Please indicate the internal IP network used for transit routing and routing protocol.

	Network	Network mask	Host address	Routing protocol
Endpoint 01				Please select
Endpoint 02				Please select
Endpoint 03				Please select
Endpoint 04				Please select



O_2 Gateway Site type and channels

Access via O_2 Gateway(16)	Please select		
Endpoint 01 site type(17)	Please select	Endpoint 01 channels(18)	
Endpoint 02 site type(17)	Please select	Endpoint 02 channels(18)	
Endpoint 03 site type(17)	Please select	Endpoint 03 channels(18)	
Endpoint 04 site type(17)	Please select	Endpoint 04 channels(18)	

Codec

Primary codec(19)	G.711 a-law
Primary packetisation(20)	20ms
Secondary codec(21)	G.711 a-law
Secondary packetisation(22)	20ms

Fax(T.38)(23) Enabled

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 G.729 for in-band faxes is not supported, as its compressed nature may cause tones and messages to be lost.

The same CODEC configuration will be applied to all SIP endpoints. The primary and secondary codec must be different to apply a secondary codec. If both codec(s) are the same only a primary codec will be applied.

Telefonica

O2 business

SIP Trunking

SIP over O_2 Gateway

Outbound call barring

As standard outbound calls to international and premium rate numbers are blocked. If you require to support call terminations to International and / or premium rate numbers, please ensure this option is un-checked.

No call barring[24]		All check call options will be barred.	
International call barring[25]	●	Mobile call barring (071-079)[26]	
Premium number barring (09)[27]	●	Personal number calls barred (070)[28]	
Special services calls up to 7 ppm (084)[29]		Special services calls up to 13 ppm (087)[30]	
Directory enquiries calls barred (118)[31]		All call calling inc 01,02,03,08 and shortcodes[32]	

Network fraud management

Do you require network fraud management[33] Please select

Daily spend limit[34]		Spending must not exceed this value in any 24 hour period
Weekly spend limit[35]		Spending must not exceed this value across 7 consecutive days

DDI ranges

Do you require a new DDI range 1[36]	Qty		Area code[37]		Consecutive range[38] Yes
Do you require a new DDI range 2[36]	Qty		Area code[37]		Consecutive range[38] Yes

Resilience+ DDI range[39]		Resilience+ DDI range[39]	
Premise[7]		End user first name[8]	
Street/throughfare[7]		End user last name[8]	
Town/locality[7]		Job title[8]	
Country code[7]	GB	Telephone[8]	
Postcode[7]		Mobile number[8]	
		Email[8]	

Please confirm the business address details via Royal Mail.End user details, where possible please provide a 24 hour contact

If new additional DDI ranges are required please fill in DDI order form

Porting DDI ranges

Do you require number porting[40] Please select

Please complete 002_SIP Port Number order form

Please note porting orders can only be accepted after all SIP Trunking testing has been completed and test results submitted.

CLI set-up

CLI flexibility required[41] Please select

Incoming CLI rule[42] Please select

As an optional service, O_2 can enable the ability to present NON O_2 registered CLIs as the presentation A-Number CLI.

Telefonica

O2 business

SIP Trunking

SIP over O_2 Gateway

Managed acceptance testing

The following tests need to be undertaken in commissioning of customer premises equipment (CPE) that is connecting to the O_2 SIP Trunking service. The managed acceptance testing service will include the following 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 excluded from the contract, the customer's PBX maintainer will be responsible for the following service acceptance testing and the test results should be passed to CFOMT@o2.com. Should O_2 not receive a response or sign off to this document after 10 working days from handover to confirm testing has been completed, O_2 will assume that the service is working as per required and live.

Traffic on the following ports must be forwarded through the relevant routers and firewalls on the customer network.

Signalling: UDP port 5060 egress / ingress to:
Media: All UDP ports between 6000-40000 egress / ingress to:

Telefónica

O_2 business

SIP Trunking

SIP over O_2 Gateway

Managed acceptance testing

The following tests need to be undertaken in commissioning of customer premises equipment (CPE) that is connecting to the O_2 SIP Trunking service. The managed acceptance testing service will include the following 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 excluded from the contract, the customer's PBX maintainer will be responsible for the following service acceptance testing and the test results should be passed to CFOMT@o2.com. Should O_2 not receive a response or sign off to this document after 10 working days from handover to confirm testing has been completed, O_2 will assume that the service is working as per required and live.

Test 1 – 9: Inbound and outbound calls

Test 1 – O_2 SIP Trunkings call – SIP clear

Description	
Name:	Successful call – user A calling PSTN user. PSTN user clears
Preconditions:	Valid PSTN / mobile in place, that can be called and verified

Test 1 steps

Step	Description	Expected result
1	Place a call to the PSTN number using SIP service	Ring tone and answer
2	Answer the call	Speech
3	Wait approx 10 secs	
4	Clear the line on the caller side	PSTN side receives call clear down

SIP Trunking

SIP over O_2 Gateway

Test 2 – SIP call – PSTN clear

Description	
Name:	Successful call – user A calling PSTN user. PSTN user clears
Preconditions:	Valid PSTN / mobile in place, that can be called and verified

Test 2 steps

Step	Description	Expected result
1	Place a call to the PSTN number using SIP service	Ring tone and answer
2	Answer the call	Speech
3	Wait approx 10 secs	
4	PSTN clears call down	
5	Wait 120 Seconds	
6	Caller receives clear down	Called party

Test 3 – PSTN call – PSTN clear

Description	
Name:	Successful call – PSTN user calling user A. PSTN user clears
Preconditions:	Valid PSTN / mobile in place, that can place a call

SIP Trunking July 2019 version 2.1

SIP Trunking

SIP over O_2 Gateway

Test 3 steps

Step	Description	Expected result
1	Place a call from the PSTN line calling the SIP assigned geo number	Ring tone and answer
2	User A answers the call	Ringing before answer
3	Wait approx 10 secs	
4	Clear the line on the caller (PSTN) side	Called side receives call clear down

Test 4 – SIP call – SIP user release without answer

Description	
Name:	Calling party release before answer – calling party is user A1.
Preconditions:	Valid PSTN / mobile in place, that can be called

Test 4 steps

Step	Description	Expected result
1	Place a call on the SIP line calling the PSTN number	Ring tone
2	Release the call without answer	Ringing stops

Test 5 – SIP call – PSTN user release without answer

Description	
Name:	Calling party release before answer – calling party is PSTN user
Preconditions:	Valid PSTN / mobile in place, that can place a call

Telefonica

O2 business

SIP Trunking

SIP over O_2 Gateway

Test 5 steps

Step	Description	Expected result
1	Place a call on the PSTN line calling the SIP assigned O_2 geo number	Ring-back (Ring tone)
2	Release the call without answer	Ringing stops

Test 6 – Invalid number call test

Description	
Name:	Invalid number – user A calling from O_2 SIP Trunkings
Preconditions:	

Test 6 steps

Step	Description	Expected result
1	Call the invalid number	Number unobtainable tone or not recognized prompt.

Test 7 – Incoming call – PSTN busy

Description	
Name:	Incoming call – user busy. PSTN busy
Preconditions:	Valid PSTN / mobile in place, that has no answering machine or call waiting feature installed and a third phone number (user C) which can answer a call

SIP Trunking

SIP over O_2 Gateway

Test 7 steps

Step	Description	Expected result
1	Call the third phone number (user C) from the PSTN line, wait until it gets answered	
2	Place a call on the SIP line calling the PSTN number	Busy tone

Test 8 - Incoming call – SIP user busy

Description	
Name:	Incoming call – user busy. user A busy
Preconditions:	Valid PSTN / mobile in place, and a third phone number which can answer a call

Test 8 steps

Step	Description	Expected result
1	Call the third phone number from the SIP line, wait until it gets answered	
2	Place a call on the PSTN line calling the SIP number	Busy tone

Test 9 - address incomplete

Description	
Name:	address incomplete – user A calling
Preconditions:	

Test 9 steps

Step	Description	Expected result
1	Call the following incomplete number from the SIP line: 654321	Fast busy or service announcement

Test 10 – 13: CLI presentation

In order to successfully present an 'A number' we expect this to be sent to O_2 in either of the following formats in the FROM header of the SIP INVITE:

10 Digits without leading zero

From: <sip:1625827748@83.245.6.117>;tag=3541226335-339769

E164 Format (+44)

From: <sip:+441625827748@83.245.6.117>;tag=3541226335-339769

The A-number is checked against a database on the O_2 network of geographic numbers that are allocated to the SIP endpoint. If the number presented does not meet the above criteria, the A-Number CLI presented will be a default CLI, which is the first number in the O_2 allocated geographic DDI range.

Test 10 - CLI Presentation test - PSTN > SIP

Description	
Name:	CLIP – PSTN calling end. Confirm A-end phone rings and number is displayed
Preconditions:	Valid PSTN / mobile in place which has present CLI setting enabled, and CLI capable phone set connected to SIP line

Test 10 steps

Step	Description	Expected result
1	Place a call on the PSTN line calling the SIP assigned O_2 geo number	Ring-tone and correct CLI

Telefonica

O$_2$ business

SIP Trunking

SIP over O_2 Gateway

Test 11 - CLIR test PSTN > SIP

Description	
Name:	CLIR – PSTN calling end. Confirm A-end phone rings and number is withheld.
Preconditions:	Valid PSTN / mobile in place which has CLI setting set to withheld, and CLI capable connected to SIP line.

Test 11 steps

Step	Description	Expected result
1	Place a call on the PSTN line calling the SIP assigned O_2 geo number	Ring-tone and no CLI or withheld shown as CLI

Test 12 - CLI presentation test - SIP > PSTN

Description	
Name:	CLIP – A-user calling end. Confirm that PSTN phone rings and caller number is displayed
Preconditions:	Valid PSTN / mobile in place which is capable showing CLI , and SIP CPE set to enable CLI sending

Test 12 steps

Step	Description	Expected result
1	Place a call on the SIP line calling the PSTN number	Ring-tone and correct CLI own with leading zero

SIP Trunking

SIP over O_2 Gateway

Test 13 - CLIR test SIP > PSTN

Description	
Name:	CLIR – A-user calling end. Confirm PSTN phone rings and caller number is withheld.
Preconditions:	Valid PSTN / mobile in place which is capable showing CLI , and SIP CPE set to privacy full (hide CLI)

Test 13 steps

Step	Description	Expected result
1	Place a call on the SIP line calling the PSTN number	Ring-tone and no CLI or

Test 14 - 15: Fax

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 G.729 for in-band faxes is not supported, as its compressed nature may cause tones and messages to be lost.

Test 14 - CLI presentation test - PSTN > SIP

Description	
Name:	Fax call – from PSTN (G.711)
Preconditions:	Fax equipment connected to PSTN line, second fax machine connected to SIP line, SIP Service CPE set to g.711 fax pass through.

Test 14 steps

Step	Description	Expected result
1	Send a 5 page fax call on the PSTN line calling the SIP assigned O_2 geo number	Ring-tone, training and answer
2	Check fax transmission	Fax machines should train and send all pages with no errors

Test 15 - Fax call to PSTN (If configured for Fax support)

Description	
Name:	Fax call – from fax user in 'A' domain (G.711)
Preconditions:	Fax equipment connected to PSTN line, second fax machine connected to SIP line, SIP Service and CPE can negotiate the codec to g.711

Test 15 steps

Step	Description	Expected result
1	Send a 5 page fax call on the PSTN line calling the SIP assigned O_2 geo number	Ring-tone, training and answer
2	Check fax transmission	Fax machines should train and send all pages with no errors.

Test 16 - 17: Call barring

Test 16 - Call barring – If requested to be set up on the SIP account

Description	
Name:	Call barring – confirm that prohibited numbers are blocked.
Preconditions:	Provisioning of call barring SIP service should be requested and completed.

Test 16 steps

Step	Description	Expected result
1	Call a number which is barred from the SIP line (premium, mobile, international)	Fast busy signal or barred prompt

Test 17 – Call international number (if no international bar is in place)

Description	
Name:	Call barring – confirm international numbers can be dialled
Preconditions:	International call barring = No

Test 17 steps

Step	Description	Expected result
1	Call the following number from O_2 SIP Trunkings (Paris, France): 0033170758109	Ring-tone, will be unanswered

SIP Trunking

SIP over O_2 Gateway

Test 18 - 24: Shortcode dialling

As part of the provisioning process the endpoint is automatically configured to a range of short codes for emergency services and directory enquiries.

Test 18 – Dial 999 shortcode

Description	
Name:	Dial 999 shortcode
Preconditions:	Confirm that a valid End user 999 address is provisioned against the calling party number (All 999 calls without EU address details are reported to OFCOM)

Test 18 steps

Step	Description	Expected result
1	Place a call to 999 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 19 – Dial 100 shortcode

Description	
Name:	Dial 100 shortcode
Preconditions:	Call 100 from the SIP line

SIP Trunking

SIP over O_2 Gateway

Test 19 steps

Step	Description	Expected result
1	Place a call to 100 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 20 – Dial 101 shortcode

Description	
Name:	Dial 101 shortcode
Preconditions:	Call 101 from the SIP line

Test 20 steps

Step	Description	Expected result
1	Place a call to 101 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 21 – Dial 111 shortcode

Description	
Name:	Dial 111 shortcode
Preconditions:	Call 111 from the SIP line

SIP Trunking July 2019 version 2.1

SIP Trunking

SIP over O_2 Gateway

Test 21 steps

Step	Description	Expected result
1	Place a call to 111 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 22 – Dial 112 shortcode

Description	
Name:	Dial 112 shortcode
Preconditions:	Call 112 from the SIP line

Test 22 steps

Step	Description	Expected result
1	Place a call to 112 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 23 – Dial 195 shortcode

Description	
Name:	Dial195 shortcode
Preconditions:	Call 195 from the SIP line

SIP Trunking

SIP over O_2 Gateway

Test 23 steps

Step	Description	Expected result
1	Place a call to 195 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 24 – Dial 123 shortcode

Description	
Name:	Dial 123 shortcode
Preconditions:	Call 123 from the SIP line

Test 24 steps

Step	Description	Expected result
1	Place a call to 123 via SIP line	Ring-tone
2	B party answers call	Speech
3	Confirm to B party that test call is being made and terminate call	Clear down (BYE message on SIP)

Test 25: DTMF

The following methods will be supported to transport DTMF tones:

The 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_2will not guarantee the delivery of in-band DTMF over a G729 codec.

Test 25 – DTMF

Description	
Name:	Test DTMF
Preconditions:	Call 08081788000

Test 25 steps

Step	Description	Expected result
1	Place a call to 08081788000 via SIP line	Ring-tone
2	Connect to IVR	Speech
3	Press 1 to navigate menu	Tone recognised, forwarded to next stage of IVR

SIP Trunking

SIP over O_2 Gateway

SIP header configuration

Request-URI Header

The number within the request-URI is used for routing calls and should contain the called number. The format of this number can either be UK national format (with leading zero), UK international format (with two leading zeros and a country code), or full e.164 format (with leading +).

Examples

Option	Format	SIP header
1	E.164	INVITE sip:+441618777148@100.100.100.29 SIP/2.0
2	International format	INVITE sip:00441618777148@100.100.100.29 SIP/2.0
3	UK format	INVITE sip:01618777148@100.100.100.29 SIP/2.0

TO header

Header must contain SIP Trunking SIP gateway address and the called number. The format of this number can either be UK national format (with leading zero), UK international format (with two leading zeros and a country code), or full e.164 format (with leading +).

Examples

Option	Format	SIP header
1	E.164	To: <sip:+441618777148@100.100.100.29>
2	International format	To: <sip:00441618777148@100.100.100.29>
3	UK format	To: <sip:01618777148@100.100.100.29>

Telefonica

O$_2$ business

SIP Trunking

SIP over O$_2$ Gateway

FROM Header

The user part of the SIP URI within the 'From' header must contain the calling line identity of the originating device. This identity must be a number registered to the endpoint. The format of this number can either be national significant (with or without a leading zero), or in full e.164 format. The domain part of the SIP URI within the 'From' header can contain either the IP address of the public facing interface or a customer-defined FQDN. O$_2$ will not use this FQDN for any purpose

Examples

Option	Format	SIP header
1	E.164	From: <sip:+441618777148@100.100.100.29> or using FQDN From: <sip:+441618777148@your.domain.local>
2	International format	From: <sip:01618777148@100.100.100.29> or using FQDN From: <sip:01618777148@your.domain.local>
3	UK format	From: <sip:1618777148@100.100.100.29> or using FQDN From: <sip:1618777148@your.domain.local>

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, O$_2$ will insert a PAID header with the default network CLI.

Examples

Option	Format	SIP header
1	E.164	P-Asserted-Identity: <sip:+441618777148@100.100.100.29>
2	International format	P-Asserted-Identity: <sip:01618777148@100.100.100.29>
3	UK format	P-Asserted-Identity: <sip:1618777148@100.100.100.29>

SIP Trunking July 2019 version 2.1