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| IP Exchange Technical Description V6.6 |
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IP Exchange

Technical Description

Version 6.8

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# Document History

This section provides a brief description of the changes between issues of the BT IP Exchange Technical description.

| **Description of change** | **Issue** | **Date** |
| --- | --- | --- |
| Document reformatted. Updates to sections 3.1, 3.5, 4.4.3, 4.4.5, 4.4.6, 4.4.10, 4.7, App B, App C. New section 4.4.11, App | 6.6 | November 2020 |
| Updates to sections 3.4, 3.5, 4.8.3, 4,4.5.1, 4.8.5, 4.4.2, 4.4.10, 4.4.13, 4.4.14. New section 4.4.18. Removed section 4.8.6 | 6.7 | December 2022 |
| Minor changes to barring table information for mobile 07x and 159. Section 4 – admin updates to NICC and other technical standard references. Section 4.4.14.2 wording update, context same. Section 4.7 and Appendix B – voided/text removed. | 6.8 | April 2024 |

# Introduction

This document provides a technical description of the IP Exchange Service. The latest version of this document can be found at <https://www.btwholesale.com/pages/static/products-services/ip-exchange.htm> under the Handbook and Technical Documents section.

# Service Outline

## General

IP Exchange provides SIP signalling as a method for Communication Providers to interconnect with BT’s VoIP network, supporting VoIP to VoIP calling as well as calls to/from the PSTN.

Support for SIP-I (UK or otherwise) is no longer available on IP Exchange.

Signalling transport is over UDP by default. Signalling transport over TCP can be supported under exceptional circumstances. Signalling transport over SCTP is not available.

The following types of calls will be supported across this interface:

* Voice calls to/from PSTN destinations (UK and International)
* Voice over IP and Video over IP (UK and International) will also be supported where the origination and the destination are both on the BT IP Exchange network.
* Emergency calls (112, 999 & 18000) for UK originated calls with prior agreement only.
* Access to BT services such as Speaking Clock (123) and Directory Enquiry (BT and Non BT suppliers).

H.323 is out of scope of the service offering.

IP Exchange will, in future, fail calls that contain invalid CLIs. The definition of what constitutes a valid CLI can be found in Ofcom General Conditions C6 and revised CLI Code of Practice. The date at which IP Exchange will implement this can be found in the IP Exchange Roadmap – published quarterly.

BT IP Exchange is hosted on the BT SDIN platform. The terms SDIN, IPX and IP Exchange are effectively interchangeable within this document.

## Customer Interface

For a CP to access the BT IP Exchange service, the CP must be able to provide the following:

* A SIP proxy to interface to BT’s network over the access methods described below.
* Secure access to their network by means of a Firewall or a Session Border Controller.

Access to the BT IP Exchange network is provided across the internet or by means of Direct Access. This can be provided by BT with the following options:

* IP Exchange Direct Access provides a dedicated access for CPs that want to connect to BT. CPs can connect either at selected BT sites or a Neutral Access Point (NAP) located in London, see appendix A for details.  
  The Direct Access service logically extends the LAN at the BT IP Exchange POP to either the CP’s network termination point in the Neutral Access Point (NAP) or POP. This is separate to the LAN used for connectivity via the Internet.
* Internet access provided by BT or a third party ISP.

Direct access is normally provided with internet access as a fallback. Failover between the direct access and internet access is controlled at the SIP layer. Network layer resilience (e.g. BGP) is not supported.

## Access control

In order to provide secure access from the CP’s network into the BT network, security will be provided by means of access control using a Session Border Controller (i.e. providing firewall functionality). The CP will need to define the Signalling IP address and port addresses (default of 5060) which will be used for this service. BT will not respond to any signalling from addresses which have not been pre-agreed. In addition, BT will not respond to ICMP ping.

BT will only send signalling and media from defined IP addresses, with the media address negotiated within the signalling. Media will be sent from a range of ports between 32768 and 65535.

## Call barring

Number ranges that can be dialled through the UK IP Exchange service are managed through four profiles - profile 0 being the most open. The profiles are summarised below, with the full details of the barred number ranges on the following pages. Global CPs predominately use profiles 4 and 5.

|  |  |  |  |
| --- | --- | --- | --- |
|  |  | UK IPX | Global IPX |
| *Profile 0* | Unrestricted. Profile 0 will only be offered to CPs that have a credit risk assessment of very low risk. | Tier 1s only |  |
| *Profile 1* | Allow access to all CP number ranges  Allow access to UK PSTN ranges +441x, +442x, +443x & +445x  Allow access to International Number ranges  Allow access to mobile ranges +447x  Allow access to +449x  Allow access to some +44844 and +44871 ranges  Allow access to emergency services (999xxx, 112xxx, 18000xxx)  Allow access to service codes (123, 159, 118xxx, 18001-9)  Bar access to Service codes (100 etc), Customer Services, CLI prefixes, CLI related (147x), Indirect Access, Dialup internet & International Free Phone | ✓ |  |
| *Profile 2* | As Profile 1, but with +449x barred. | ✓ |  |
| *Profile 3* | As Profile 1, but with +449x and fixed fee call ranges barred. | ✓ |  |
| *Profile 4* | For GIPX CPs that do not send in ITFS calls |  | ✓ |
| *Profile 5* | For GIPX CPs that do send in ITFS calls |  | ✓ |
| *Profile 6* | Reserved |  |  |
| *Profile 7* | Reserved |  |  |
| *Profile 8* | For GIPX CPs only sending ITFS calls |  | ✓ |
| *Profile 9* | Reserved |  |  |

The full details of the barred number ranges per profile are as follows:

|  | **Numbers starting with** | **Profile 1** | **Profile 2** | **Profile 3** | **Profile 0** |
| --- | --- | --- | --- | --- | --- |
| Service Codes | 100 | X | X | X |  |
| Indirect Access | 124 | X | X | X |  |
|  | 125 | X | X | X |  |
|  | 126 | X | X | X |  |
|  | 127 | X | X | X |  |
|  | 128 | X | X | X |  |
|  | 129 | X | X | X |  |
|  | 13x (130 to 139) | X | X | X |  |
|  | 140 | X | X | X |  |
| CLI prefix | 141 | X | X | X |  |
|  | 143 | X | X | X |  |
|  | 145 | X | X | X |  |
|  | 146 | X | X | X |  |
| CLI related | 147 | X | X | X |  |
| IA | 148 | X | X | X |  |
|  | 149 | X | X | X |  |
| Access to voice mail | 1571 | X | X | X |  |
| Customer services | 15x (150 to 158) | ~~X~~ | ~~X~~ | ~~X~~ |  |
| IA | 16x (160 to 169) | X | X | X |  |
|  | 17x (170 to 179) | X | X | X |  |
| IA | 181 | X | X | X |  |
|  | 182 | X | X | X |  |
|  | 183 | X | X | X |  |
|  | 184 | X | X | X |  |
|  | 185 | X | X | X |  |
|  | 186 | X | X | X |  |
|  | 187 | X | X | X |  |
|  | 188 | X | X | X |  |
|  | 189 | X | X | X |  |
| Network address code | 944 | X | X | X |  |
|  |  |  |  |  |  |
| Personal Number | +448089 | X | X | X |  |
|  | +4482 | X | X | X |  |
|  | +448710 | X | X | X |  |
| Premium Rate | +4490 |  | X | X |  |
|  | +4491 |  | X | X |  |
|  | +4498 |  | X | X |  |
| Fixed fee calls | See embedded list for number ranges |  |  | X |  |
| International Free Phone | +800 | X | X | X |  |

The **X** in the profile column indicates numbers starting with those digits are barred.

Profiles 4 and 5 used for Global IP Exchange CPs are broadly similar to those shown above. The number ranges barred for Global IP Exchange CPs are shown here, with the difference between profile 4 and 5 being that profile 5 permits international toll free service ranges to be used. A plus prefix is automatically added in front of the dialled number if it not present in the original dialled number. The call is then routed in the expectation that the numbers immediately following the plus prefix identify the country code of the destination.

|  |  |  |  |
| --- | --- | --- | --- |
| **Global IP Exchange** | | | |
|  | **Numbers starting with** | **Profile 4** | **Profile 5** |
| *All non E.164 numbers* | *Any value not prefixed with a +* | n/a | n/a |
| Personal Number | +4470  +44701194 | X  X | X  X |
| UK Freephone | +44500 | X | X |
|  | +44808 | X | X |
|  | +44899 | X |  |
|  | +4482 | X | X |
|  | +448440 | X | X |
|  | +448710 | X | X |
| Premium Rate | +4490 | X | X |
|  | +4491 | X | X |
|  | +4498 | X | X |
| Fixed fee calls | +4484464 | X | X |
|  | +4484468 | X | X |
|  | +4487162 | X | X |
|  | +4487168 | X | X |
|  | +4487169 | X | X |
|  | +4487182 | X | X |
|  | +4487195 | X | X |
| PPC/Fixed fee ranges | See embedded list for number ranges | X | X |
| International free phone | +800 | X |  |
| Emergency Service | 999,112, 18000 | X | X |

Profile 8 only allows access to UK +44899 ranges used for ITFS calls. Some numbers that start with +44899 are barred as they are not used for ITFS calls. Number ranges allowed by Profile 8 are defined as:

|  |  |  |  |
| --- | --- | --- | --- |
| Only allow calls to numbers starting with | +44899 | except where the number range starts with: | +44899111  +448998  +4489990  +4489991  +4489992  +4489993  +4489994  +4489995  +4489996  +4489997  +44899985 |

If a call is rejected due to be being barred within IP Exchange then a 403 response is returned with a message of either “Call barred as it is not in agreed call profile” or “Call Barred By Network” in the status line.

## Supported dial-in ranges for PSTN to IP Exchange calls.

Calls from the PSTN into IP Exchange are supported for the following UK ranges:

* Geographic starting with +441, +442
* UK wide ranges starting with +443
* Corporate/LIECS starting with +445
* NGCS (Non Geo Call Services) +448xx - selected ranges and charge bands
* NGCS short codes (for example 101/105/111/116/118/119)
* Premium Rate starting with +449
* Mobile starting with +447x
* Indirect access codes

## 101/111 calls

IP Exchange delivers these calls into BT’s PSTN with accompanying digits identifying IP Exchange as a transit VOIP network. The destination network receiving these calls should interpret these digits as a sign that any CLI accompanying the call is not a completely reliable method of identifying the call originator’s geographical location. Both Network and Presentation CLIs are visible to the terminating service – routeing to the appropriate authority is performed on the Network CLI.

## Indirect access / carrier pre-select calls

Indirect Access Codes are not in E164 format, are only diallable from within the UK, and are used to route calls to a particular operator. They come in two forms – those where the IA code is followed by a normal dialled number (e.g. 1800101473123456, aka 1-stage), and those where the IA code alone is dialled initially (e.g. 18001) and then the number the caller wants to reach is dialled “in band” once the call to the CP has been set up (2-stage). In both the above cases, calls are only permitted from the PSTN into IPX.

There is a potential post-dial delay (PDD) issue for 1-stage IDA calls as the PSTN does not know how many digits to expect, and therefore when to send an “end of pulsing” message to the gateway to forward the invite into IP Exchange. There is currently no option for custom IDA code builds to avoid the 4 second additional post-dial delay for 1-stage codes.

Carrier preselect codes are another variation on the design used for Indirect Access, in that a prefix is used to route calls into IP Exchange. These codes start with an 8 rather than a 1 in the above cases, but their behaviour is comparable to 1-stage IDA calls when ingressing IP Exchange from the PSTN.

## Minimum/maximum number lengths for international calls

For calls to international destinations, each country will have rules about how long numbers can be for the major call types (fixed, mobile etc.). SDIN/IPX/GIPX behaves as follows:

• If a call to that country has fewer digits than the minimum for the specific call type, the call will be failed with 403 Forbidden. This helps to terminate an invalid call at the earliest opportunity.

• If the number of digits exceeds the maximum for a country/destination/call type, the excess trailing digits will be removed, and the call will be progressed. This removes one exploitation scenario for making fraudulent calls by dialling long sequences of numbers. It should be noted that this truncation of the original dialled number value might result in behaviour that the caller does not expect. For example, a caller has dialled a number and entered DTMF tones during that call from a phone that stores key-presses. If the caller presses redial once the first call has finished the call then the phone will relay the full set of key-presses as the destination number, including any key-presses that relayed DTMF in the initial call. The redial should succeed as the extra digits should be removed leaving behind a routable destination number.

## SBC resilience

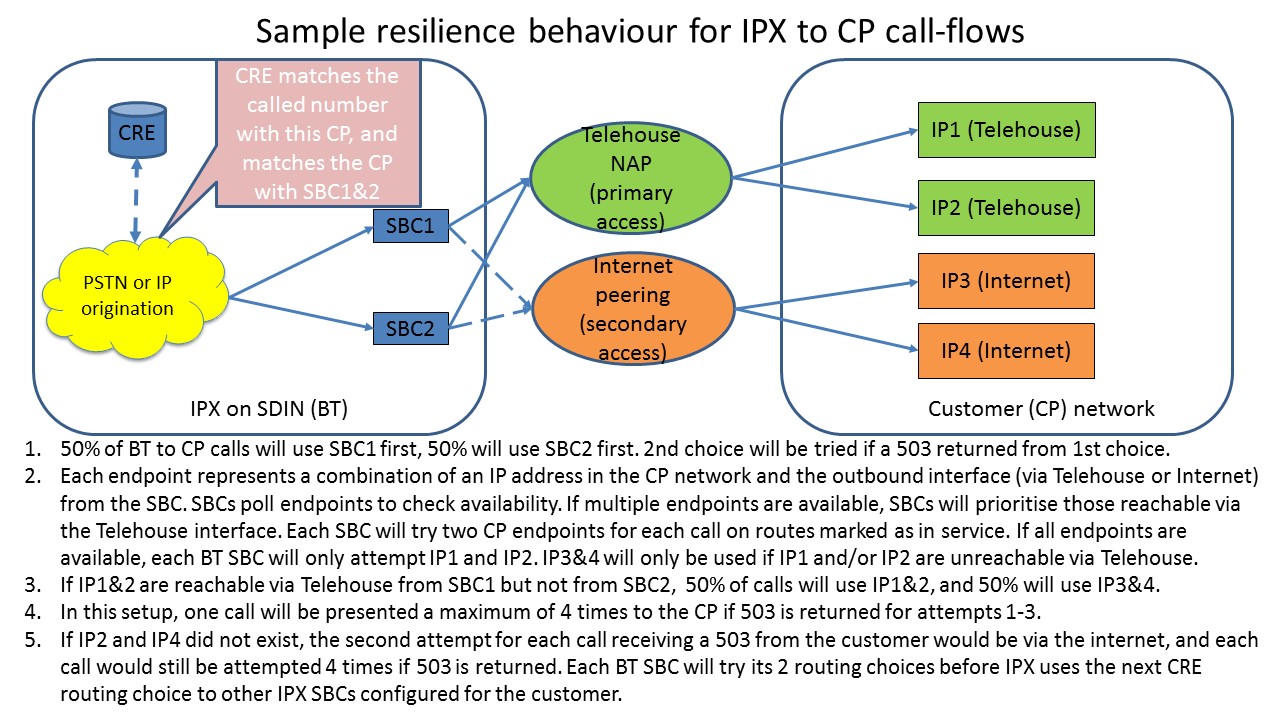
The SBCs deployed at the edge of the IP Exchange platform are deployed as high-availability devices (active/standby) across multiple sites. Subject to the rules governing the minimum port requirement, CPs can request their service is distributed over more than one SBC and site. Service provisions over 200 ports are normally split over two or more SBCs, e.g. 500 ports is normally provisioned 250 on each of two SBCs, one at each of two sites.

Calls sent to CPs from IP Exchange will normally be load-shared over all the available IP Exchange SBCs where a CP is hosted. If a CP has multiple access options (e.g internet and NAP/POP connections) to multiple SBCs then these are built in to the routing logic to the CP from each IPX SBC.

The distribution of calls from the IP Exchange SBCs to a CP’s call-servers can be load-shared or use a hunt strategy. The availability of servers within a CP’s network is determined by sending Options requests – see section 4.4.19 for more details.

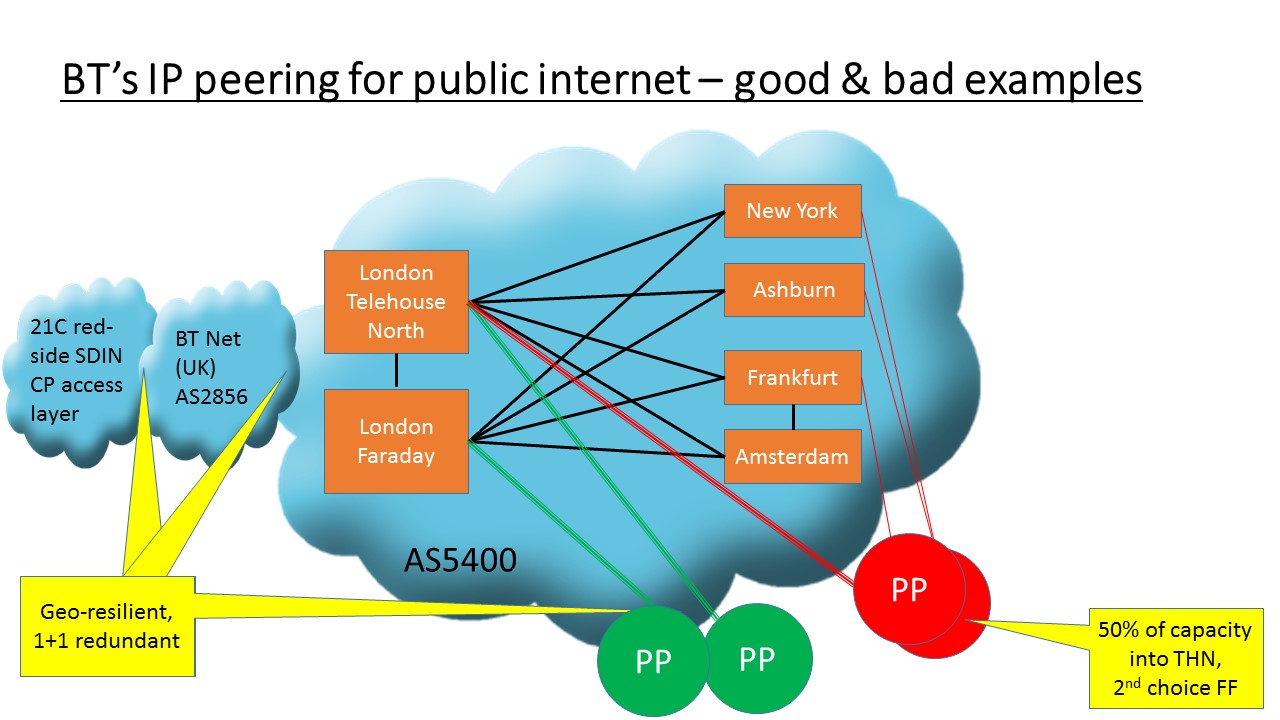
CPs sending calls into IP Exchange are strongly recommended to use a load-sharing strategy to distribute calls over the IP Exchange SBCs where they are hosted. If a CP has multiple access options through to a particular SBC, then the expectation is they are used in a primary/secondary sequence.

An aggregated call admission control constraint is applied against the CP’s configuration on the SBC, regardless of whether the CP was the originator or terminator. Additional CAC constraints can also be applied on inbound or outbound call streams upon request.



## Internet Access resilience

The following picture is a high level view of BT’s internet peering arrangements, with the IP Exchange service running on the SDIN platform on the left, and peering partners that IP Exchange CPs may use for internet connectivity into IP Exchange shown as PP.



Good peering providers into BT have:

* Equal capacity links into BT via Telehouse and Faraday
* Redundancy – one of Telehouse/Faraday links or buildings can fail entirely and full service can be maintained e2e.

Bad (in comparison to “good” above) peering providers into BT have:

* One primary large bandwidth link into BT via either Telehouse or Faraday
* Resilience provided by a number of smaller links, none of which on their own can cope with traffic levels normally seen via the primary link into BT in addition to their normal traffic, and no reliable way of spreading capacity to alternative BT peering points in an emergency

IPX CPs relying on public IP connectivity into IPX/GIPX (either for primary or as a backup) are urged to check that their ISPs/peering partners used for BT/IPX traffic provide an end-to-end solution that is sufficiently resilient to a significant incident, e.g. a complete power outage at BT’s Faraday building, or at Telehouse North which is under third party management.

# Technical Specification

## IP addressing

IP version 4.0 is supported.

IP Version 6.0 is not supported.

## DNS

In terms of the CP interaction with BT’s network, DNS capability, including SRV and A record look up, is not supported.

## SIP transport

IP Exchange supports the transport of SIP signalling messages using UDP as the preferred option. SIP messages sent using TLS, SCTP and IPSEC are not supported at present.

## SIP signalling

The following is based on NICC ND1035 [12] and how BT support for IPEX.

### SIP Methods

The list of supported SIP methods are detailed below.

|  |  |
| --- | --- |
| **Method** | **Supported** |
| Invite | Yes |
| Ack | Yes |
| Bye | Yes |
| Cancel | Yes |
| Prack | Yes |
| Options | Yes |
| Info | Partial (See section 3.4.13) |
| Update | Yes |
| Refer | No |

### SIP Responses

Provisional and final response codes are normally forwarded through IP Exchange unchanged. There are exceptions to this.

* Codes in the 3xx series are mapped to a 403 Forbidden response.
* A 488 or 606 response code received before any 18x will cause the Invite to be forwarded to a transcoder, the ultimate final response returned to the call originator is unlikely to be a 488/606.

Upstream carriers may receive an SDP answer sent in an unreliable 18x. BT IP Exchange will regard this as the final answer as it cannot be different when sent in a subsequent reliable response (200 OK).

### Format of INVITE Fields

IP Exchange supports two types of addressing in terms of the format of the SIP signalling:

• Option a – FQDN based.

• Option b – IP based.

***Option a – FQDN Based CPs***

For Break Out calls (i.e. CP to IP Exchange), the host part of the INVITE Request URI and To field will contain the FQDN of the destination network (i.e. uk.sdin.bt.net for calls to IP Exchange). The host part of the From field will contain the FQDN of the origination network. The Contact header in the INVITE must be an IP address belonging to the origination network.

For Break In calls (i.e. IP Exchange to CP), the host part of the INVITE Request URI and To field will contain the FQDN of the CP’s network. The host part of the From field will contain the FQDN of BT’s network (i.e. uk.sdin.bt.net).

***Option b – IP Based***

For Break Out calls, the host part of the Request URI will contain the IP address of BT's SBC, as will the host part of the To header. The From header will contain the IP address of the CP’s signalling proxy. The Contact header in the INVITE must be an IP address belonging to the origination network.

For Break In calls, the host part of the Request URI will normally contain the IP address of the CP’s signalling proxy.

The host part of the To field will contain the IP address of the CP’s signalling proxy, and the From field will contain the IP address on BT’s SBC.

The tables below provide some examples of how the different fields in the INVITE need to be populated.

**INVITE from CP to IP Exchange**

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
|  |  | **Request URI** | **To URI** | **From URI** |
| a | FQDN Based | sip:+441214157178@uk.sdin.bt.net | sip:+441214157178@uk.sdin.bt.net | sip:+442073565000@CP’s FQDN |
| b | IP Based | sip:+441214157178@BT's SBC IP Addr | sip:+441214157178@BT's SBC IP Addr | sip:+442073565000@CP's Proxy IP Addr |

**INVITE from IP Exchange to CP**

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
|  |  | **Request URI** | **To URI** | **From URI** |
| a | FQDN Based | sip:+442073565000@CP’s FQDN | sip:+442073565000@CP’s FQDN | sip:+441214157178@uk.sdin.bt.net |
| b | IP Based | sip:+442073565000@ CP's Proxy IP Addr | sip:+442073565000@CP's Proxy IP Addr | sip:+441214157178@BT's SBC IP Addr |

Note that the “uk.sdin.bt.net” FQDN used does not resolve using DNS. Resolution of CPs’ FQDNs is by a static definition recorded as part of the Customer Requirements Form (CRF) process.

### Format of the numbering in the INVITE fields

For Break Out calls from the CP’s network, the INVITE Request Line must contain a SIP URI.

For calls from PSTN to CP the INVITE Request Line will contain a SIP URI.

***Destination Number***

The destination address can be a national number, international number or UK specific number.

UK Specific numbers include Emergency, TypeTalk and Ported Numbers where the number string is not conformant to E.164. The format for national/international numbers will be in the “+cc” format and not “00cc” format.

|  |  |
| --- | --- |
| **Type of destination** | **Example format for Request URI** |
| Calls to national or international numbers\* (Example of calls to SDIN) | SIP URI = <sip:+442073565000@uk.sdin.bt.net;user=phone> |
| Calls to UK access codes (i.e. short codes) (Example of calls to SDIN) | SIP URI = <sip:123;phone-context=+44@uk.sdin.bt.net;user=phone> |
| Calls to indirect access codes (Example of calls from SDIN to CP) | SIP URI = <sip:140802073565000;phone-context=+44@sip.company.com;user=phone>  (in this example 1408 is the indirect access code) |

\* This includes calls to UK international operator services, supported from May 2018. This covers the +44800799, +44899989 and +44910001000 options previously supported on BT’s TDM IDD interconnects, but not the code 11/code12 (b/c) options.

When forwarding a call to a CP that has been passed into IP Exchange through the end-user dialling an indirect-access code, carrier preselect code, CP-hosted short code, or CP-owned porting prefix the number string in the Request-URI and To fields will consist of the code, followed by the dialled digits. IP Exchange will not modify this number string to add or remove country codes.

As per the 123 example in the table above, IP Exchange expects calls to short codes not include +44 as a prefix.

***CLI Number***

The format of the number in the From / P-Asserted-Identity fields will be in E.164 international format as shown in the following example for calls from IP Exchange to the CP’s network. The format for CLI will be in the “+cc” format and not “00cc” format.

e.g.

From: <sip:+441214157178@uk.sdin.bt.net;user=phone>;tag=456248

***UK regulations regarding CLI***

In order to conform to the UK regulation conditions outlined within OFCOM General Condition C6 and compliance in general to UK CLI guidelines, BT IP Exchange have implemented CLI screening. Appendix F provides the details of briefings sent to UK IPX and GIPX customers.

### Calling Line Identity Presentation / Restriction support

IP Exchange supports both Network and Presentation CLIs - section 6 of <http://stakeholders.ofcom.org.uk/telecoms/policy/calling-line-id/caller-line-id/> provides a short definition of these terms.

#### CP to IPX

***CLI***

IP Exchange expects CPs to provide a valid network number in P-Asserted-Identity header conforming to Ofcom’s CLI Guidance [14] and NICC standards ND1035 [12] & ND1016 [11].

Additional information on how to populate SIP URI header fields with a telephone number can be found in ND1439 [13].

BT IP Exchange will remove the P-Preferred-Identity (PPID) header if it is present for calls from CPs to BT IP Exchange. P-Preferred-Identity (PPID) is a header normally associated with the user-to-network layer. BT IP Exchange does not expect to see a PPID used over a network-to-network interface (NNI).

***CLI Restriction***

The CP must indicate that CLI is not for CLI display purposes by use of the method detailed in Section 7 of ND1035 [12]. BT IPEX/SDIN does not currently support the use of priv-value ‘user’, therefore CPs should send From headers with a ‘Anonymous’ URI.

#### 4.4.5.2 IPX to CP

In line with Ofcom UK regulations, CLI Guidelines, ND1016 [11] and ND1035 [12], calls to CPs will contain a From header field populated with the information required to be used for display to the called party. Regardless of Privacy header settings the content of the P-Asserted-Identity must not be displayed to the called party.

#### 4.4.5.3 PSTN to IP Exchange

The mapping of CLI information from the PSTN to IP Exchange is based on ND1017 [10].

#### 4.4.5.4 IP Exchange to PSTN

The mapping of CLI information from IP Exchange to the PSTN is based on ND1017 [10].

### Diversion / History-Info

Within the core IPX network the Diversion header is used. If there is a requirement to interwork with History-Info (RFC7544 [16]), then CPs can request this.

### Call routing

CPs can send calls from multiple addresses into one or more BT POPs. Calls can be sent from the BT POPs to multiple addresses within the CP’s network. If a CP has requested to be hosted at multiple IP Exchange POPs the call constraints will be split equally between the two BT POPs. The minimum concurrent calls allowance that will be configured at each POP is 30.

In line with ND1657 [15] re-routeing is applied within the IP Exchange platform on a range of SIP server failure response codes received back from downstream networks. CPs are expected to conform to ND1657 [15].

For calls that are terminated to an IP Exchange CP then alternative routeing attempts will be made to terminate the call whenever possible, e.g. via an alternative access /connection/SBC. However, once the terminating network has returned a SDP answer (even though the call may not have been answered) then re-routeing is not possible.

Pre-pay CPs, whose service has been temporarily suspended due to their balance falling below the critical level, will receive a 503 response.

*Break Out calls – CP to IP Exchange*

For Break Out calls, it is the CP’s responsibility to ensure that the calls are delivered appropriately to the IP Exchange POPs. The Call Admission Control constraints will be shared over each of the BT POPs. For a CP to utilise the full call limit, the CP will need to evenly distribute calls over all BT POPs. Call Admission Control constraints also include break in calls and so these also need to be taken into account.

In the event of failure of connectivity to one of the BT POPs, the total call constraint at the remaining BT POP(s) will remain unchanged.

*Break In Calls – routing within IP Exchange*

Break In calls will be delivered to the CP network from one or more BT POPs. If the multi-hosted option has been chosen then by default BT will attempt to distribute break-in calls equally between the available POPs. However, one POP can be nominated as the primary one and all break-in calls will originate from there, only failing over to the secondary POP on receipt of a 503 response or if the primary route has been marked out of service.

Calls arriving into IP Exchange from the PSTN are subject to Routeing Engine lookup to determine the BT POP(s) the CP is hosted at. If the Routeing Engine has been configured to use a weighted distribution algorithm (the default) for that number then it will alternate the order of the BT SBCs that will route the call each time a query is made.

*Break In Calls – IP Exchange to CP*

For Break In calls, the CP can request the preferred strategy for receiving calls. The two available options are:

* Primary/Secondary – With this option calls are always routed to the primary proxy, unless there are connectivity problems or CP SIP proxy failure, then the secondary proxy will be selected.
* Round robin – With this option calls are shared between the primary proxy and the secondary proxy. In case of are connectivity problems or CP SIP proxy failure associated with one of the proxies, the remaining proxy will receive all the calls.

The failure of a CP SIP Proxy will be detected by the timeout of SIP Options messages, or by two consecutive timeouts of any other requests (excluding Options).

**INVITE Req URI +441234567890@north.morecalls4u.com**

**IP Exchange**

**CP SIP**

**Interface**

Call Count limit applied to the SIP Interface for the CP

**INVITE Req URI +441234567890@south.morecalls4u.com**

**Routeing Engine**

**morecalls4u’s network**

### Call forking

BT’s IP Exchange platform does not currently support receipt of Forked Responses. SIP forking should therefore be handled in the forking customer network. Calls forked before being presented to IPX will be handled as separate calls. Responses received by IPX for calls forked in downstream networks are not supported. Networks performing forking are advised that they should handle the forked responses. When/if SDIN is enhanced to handle forked responses, CPs will be informed via the BT IP Exchange Roadmap and updates to this document.

### Prevention of Session loss

In order to minimise the impact of failure of components in the CP's network or BT's network, it is recommended that within the CP network (Proxies and CPE) session timers, as specified by RFC 4028 [8] are implemented. The preferred method for CP 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-New Invite 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 [8]), 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 CPs that 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.

### Codecs

Recommended codecs for use on IP Exchange

Codecs commonly in use across IPEX are shown in the table below. The capabilities of the PSTN gateways still have a strong influence on codecs used across IP Exchange even for “on-net” calls between two IPX CPs.

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
|  | **Codecs and p-time supported on PSTN gateways** | | | **Commonly seen across IPEX** |
|  | 10ms | 20ms | 30ms | Typically 20ms |
| G711 A-law - European standard | **✓** | **✓** | **✓** | **✓** |
| G.711 µ-law - USA standard | **✓** | **✓** | **✓** | **✓** |
| G.729 | **✓** | **✓** | **✓** | **✓** |
| DTMF transport as for RFC4733 [5] | **✓** | **✓** | **✓** | **✓** |
| G.711 A-law in-band fax | **✓** | **✓** | **✓** | **✓** |
| G.722 | 🗶 | 🗶 | 🗶 | **✓** |
| AMR-WB (all rates) | 🗶 | 🗶 | 🗶 | **✓** |
| AMR-NB (all rates) | 🗶 | **✓** | 🗶 | **✓** |
| GSM-EFR | 🗶 | **✓** | 🗶 | **✓** |
| Clearmode (as defined in RFC 4040 [17]. Other variants such as ccd, x-ccd, are not supported for calls terminating on PSTN) | **✓** | **✓** | **✓** | **✓** |
| T.38 (/UDPTL) | **✓** | | | **✓** |

It is *strongly* *recommended* that all CPs offer G.711 a-law (20ms) as a common denominator when negotiating calls although there is no requirement for this to be the first choice of codec.

Transcoding on the SDIN platform is covered in Appendix C.

The G.711 codec specification defines two companding algorithms: A-law and µ-law. µ-law tends to be the preferred choice in North America and Japan, whereas A-law is common to Europe and the rest of the world. These codecs are not interoperable, and so it is recommended that the call originator supports the variant of G.711 that is in common use in the countries where they are trying to terminate calls.

The default packetisation delay for G.729 on SDIN is 20ms unless negotiated to a new value by use of the ptime parameter.

HD codecs are supported by the SDIN platform but should not be presented as the only codec in the SDP offer. A narrow-band codec (i.e. G.711 or G.729) should also be included in the offer.

The Opus codec is not currently supported on IP Exchange. The support of Opus codec has been added to the IP Exchange roadmap with estimated delivery 2023/2024 (transparent support and narrow band transcoding support).

SDIN accepts offers of both octet-aligned and bandwidth-efficient modes for AMR-NB and AMR-WB codecs. However, it will only offer bandwidth-efficient mode in Break In calls or the terminating leg of on-net transcoded calls.

Where interoperability testing has identified that a CP’s platform does not support a SDP offer/answer exchange over the 200OK/ACK messages during the call establishment phase, SDIN will insert a default codec offer (G711a) into an initial Invite request that does not contain SDP.

**Codec profiles**

SIP messages offering a large number of codecs have been observed to cause interworking problems with some vendors’ equipment. To mitigate against this, CPs hosted on IP Exchange will be placed in one of four profiles, which filter the codecs presented in the offer by varying amounts. A CP’s assigned codec profile can be altered by the usual order modification process. Note this applies to the SDP in Invites only, it does not change the SDP in the 200OK for late offer calls (Invite without SDP).

The four profiles are:

|  |  |
| --- | --- |
| 1. Full | Pass original codec offer unchanged |
| 1. Narrowband and fixed line HD | Removes the following from the offer: GSM-AMR, GSM-EFR, GSM-FR, GSM-HR |
| 1. Narrowband | Removes the following from the offer: GSM-AMR, GSM-EFR, GSM-FR, GSM-HR, G.722, AMR-WB, VMR-WB |
| 1. Minimum | Removes GSM-AMR, GSM-EFR, GSM-FR, GSM-HR, G.722, AMR-WB, VMR-WB, G.723.1, G.728, EVRC, EVRC-B, G.723.1ar, G.726, SPEEX, QCELP, iLBC, H.261, H.262, H.263 |

**Additionally Supported Codecs**

In addition to the above codecs if a call is believed to route to another IP Exchange CP or a destination network that allows them then one or more of the following additional codecs may be offered in addition to recommended codecs. BT’s IP Exchange platform would not perform any interworking between different codecs but the bandwidth requirements for the codecs listed below are defined to allow the media to pass through without restriction.

* G.723.1 (6.3 Kbps) (30ms packetisation delay)
* G.723.1 (5.3 Kbps) (30ms packetisation delay)
* G.728 (16 Kbps) (30ms packetisation delay)
* G.726 (16 Kbps)
* G.726 (24 Kbps)
* DVI4
* iLBC
* Speex
* H.263 – for video
* H.264 - MPEG4 – for video

(These are not supported for calls to the BT PSTN, and may not be for Mobile networks or International calls).

BT IP Exchange will also identify the following pseudo-codecs as 64kbit/s calls

* X-CCD
* CCD
* G.nX64
* G.Clear

These are not supported for calls to the BT PSTN and may not be by the terminating IPX CP.

**Additional Information and Caveats on Codec usage and selection**

IP Exchange polices a media-stream’s bandwidth, which can vary based on the packet size specified. If a CP exceeds the bandwidth for a specific codec, RTP packets will be discarded and this will result in poor voice quality.

***Ptime*** can be used to negotiate a different packet size to that indicated above. However, if the reduction in packet size caused by end-user ptime negotiation results in bandwidth requirements in excess of what the platform is designed for then this will result in very poor voice quality for which BT will not be held responsible.

Video is not supported for calls passing through the PSTN gateway. Video passthrough is also currently disabled between two parties that are in different IP Exchange regions. There are currently two IP Exchange regions where a CP could reside. The IP Exchange SBC where a CP is configured determines the region they are in. IP Exchange has SBCs in the UK and Bahrain.

### Suppression of RTP streams

Timers are enabled on IP Exchange that trigger automatic call termination if a media stream inactivity is detected that persists for greater than two minutes.

The timers can be disabled on a call by call basis by indicating a mode change in the media through the use of a Re-Invite message. Alternatively, if a client could produce periodic media in the absence of other speech etc., that would reset the media-inactivity timers. Default platform behaviour is that RTP-keepalive packets would need to be sent at least every 119 seconds to avoid any risk of calls dropping out. CPs can make a specific request to switch to RTCP timers instead, any need for this should ideally be verified during IOT.

AMR, AMR-WB and EFR codes have silence suppression supported by default, without requiring SDP negotiation to enable it. RTCP or SID frames would provide sufficient media activity to prevent the media-loss feature from triggering.

### Transport of Fax and Modem

FAX and Modem transport in band using G.711 A-law codec is supported. Renegotiation to T.38 is supported for calls involving the PSTN gateway but may not be for “on-net” calls. In-band with codec up-speed to G.711 is only supported for calls to/from the PSTN where renegotiation is initiated per-call by the customer SIP server, or on-net where agreed by both SIP end points.

If calls are made to another CP that does not support the method of transport for the tones, the IP Exchange platform will not perform any form of inter-working between the two different methods.

Certain types of data call are not well supported by networks that involve TDM->SIP->TDM solutions. These include calls made by PDQ credit card validation machines, and faxes configured to attempt end-to-end SuperG3 transmission.

### Transport of DTMF tones

Two methods of DTMF transport are supported on the platform: telephone-events (RFC 4733 [5]) and in-band (G.711). Support of telephone-events is dependent on successful codec negotiation. Where a mixture of narrow/wide band codecs are offered then a telephone-event payload types would need to be defined for each clock rate (8000/16000). Many CPs using IP Exchange use the dynamic payload type of 101 to indicate telephone-event/8000, although this should not be assumed to be always the case.

It is recommended that CPs support both forms of DTMF transport to ensure that “IP to IP on-net” calls can carry tones successfully. If telephone-events cannot be negotiated, a fall-back to in-band tones over G.711 is recommended.

The IP Exchange PSTN gateways support both telephone-event and in-band G.711 transport for DTMF.

Where CPs can only support telephony events, and this is declared during provisioning, SDIN will interwork between the two forms of DTMF where there is no common method between two SIP end points for on-net calls.

Transport of DTMF within a G.729 media stream is not guaranteed, and use of telephone-event codec is recommended for G.729 calls.

Telephone-events shall be used, when successfully negotiated, regardless of the voice codec used.

Use of the SIP INFO message to pass DTMF information is not supported.

### Emergency Calls

Emergency calls from a CP will require the SIP Request URI header to contain the emergency number (e.g. 999, 112 & 18000) suffixed with the agreed CP’s Interconnect Identifier (ii) digits prior to sending to IP Exchange.

Under rare and exceptional circumstances Emergency calls from a CP may receive a SIP 503 response from BT IPEX. **ALL CPs should re-route on a 503 response for Emergency traffic.**

#### Policy for handling overdialled emergency calls:

In theory, callers calling 999 could, through the stress of the situation, press 9 on their keypad more than three times. Whilst every effort should be made to connect genuine emergency calls, we cannot allow the ii digits that follow to be corrupted in the process, as these are important to the handling of the call. Further, recent investigation into examples of overdialled calls have revealed that none were genuine emergency calls.

The requirement is therefore that :-

- CPs should only forward emergency calls to BT’s SDIN/IPX platform where 999 or 112 has been recognised within the CP’s network, with additional codes 998, 991 and 992 for mobile CPs.

- if CPs detect additional digits after 999 or 112 they should either:

(a) return their normal “number not recognised” message to the caller and terminate the call, or

(b) as an additional safeguard, for 9999 (i.e one extra 9 digit as most likely misdial) they may forward the call but showing only 999 as dialled number plus their normal ii digits (and zone code/cell ID for mobile networks).

#### Policy for handling priority calls with RPH (Resource Priority Header):

There are 2 call flows where RPH needs to be considered:

* First leg emergency call from CP into IP Exchange for onward routing to BT 999 call handling centre;
  + RPH should be sent by the originating CP, IP Exchange will also prioritise stage 1 PSAP emergency calls based upon dialled digits received from the CP.
* Second leg emergency call from BT 999 call handling centre to emergency authorities e.g. police, ambulance, fire or coastguard,
  + Appendix E provides further guidance for emergency authorities and their communications provider. Only with prior agreement, IP Exchange will accept and deliver second leg UK originated emergency calls with RPH. The use of ported numbers, disaster recovery service numbers or translated numbers instead of directly routed numbers towards EAs can introduce more potential points of failure and transit legs to the call flow, as well as more possible TDM-IP conversions affecting voice quality, and may inadvertently result in 999 calls routing through networks which do not have call prioritisation in place. Each CP involved in the routing of these second leg calls has an equal responsibility to ensure that priority call marking is honoured and that calls are not inadvertently blocked by CLI screening when EISEC digits are present in the CLI.

Any agreed use of RPH for the call flows outlined above must be compliant with Annex A of NICC ND1035 [12]. Specifically that “esnet.2” is used for the value of the RPH field.

RPH MUST not be used for standard non-emergency calls. Inappropriate use of RPH by CPs will be acted upon by BT and may lead to the withdrawal of priority handling for RPH marked calls.

### Bandwidth and Call Control

It is the CP’s responsibility to keep within subscribed bandwidth and the maximum number of concurrent calls allowed. These constraints apply to all calls associated with the CP, regardless of whether they were the originator or terminator, and are applied at each BT POP that the CP connects to.

In the case that the call control constraints are exceeded at a BT POP, Invites will be rejected with a SIP 503 Response[[1]](#footnote-1). There is one exception:

If all the ports provisioned on an IPX SBC for customer use for calls in either direction are full of any type of traffic, a further 5 emergency calls will be permitted. These 5 additional ports will only be consumed by emergency calls if all the normal ports are full. If a customer has a bespoke design where, for example, two service provisions such as a type A and type B TNOR have been set up to share the same pool of ports, a single instance of 5 additional ports will be available for emergency calls from either TNOR.

A maximum calls per second limit is set based upon the number of ports purchased as shown in the following table. This limit is shared between all customer end points configured against each IPX SBC, and applies to calls from the CP to BT. A SIP 503 Response1 is returned if this limit is reached.

Additionally, unless advised otherwise, each end point has the same limit applied for calls from BT to the CP, to ensure that BT cannot flood the customer SIP server with call attempts in the event of a major customer equipment failure. For example, if a customer has 350 ports and 4 trunks configured on a BT IPX SBC, up to 7 CPS will be permitted from the CP to BT in total from all 4 trunks, and BT will send no more than 7 CPS to each trunk, giving an absolute maximum of 28CPS for BT to CP traffic.

|  |  |  |  |  |
| --- | --- | --- | --- | --- |
| **Ports** (inc. free port allowance) | **Calls/second limit** |  | **Ports** (inc. free port allowance) | **Calls/second limit** |
| 30 | 3 |  | 3500 | 35 |
| 50 | 3 |  | 4000 | 40 |
| 100 | 4 |  | 4500 | 45 |
| 200 | 5 |  | 5000 | 50 |
| 300 | 6 |  | 5500 | 55 |
| 400 | 7 |  | 6000 | 60 |
| 600 | 9 |  | 6500 | 65 |
| 800 | 11 |  | 7000 | 70 |
| 1000 | 13 |  | 7500 | 75 |
| 1300 | 16 |  | 8000 | 80 |
| 1650 | 18 |  |  | |
| 2000 | 21 |  |
| 2500 | 26 |
| 3000 | 30 |

*Note: The SDIN IP Exchange platform will send back a 503 in the SIP response in a number of circumstances. CPs will be able to tell the difference between these by examining SIP traces, as follows:*

* *When a call set-up request exceeds the Calls Per Second limit, the SBC sends “SIP/2.0 503 Overloaded” Note – this does not indicate that the BT SBC is overloaded, just that the CPs calls per second rate limit has been exceeded.*
* *When a call set-up request would exceed the ports/bandwidth limit, the SBC sends “SIP/2.0 503 Service Unavailable-No Bandwidth Available”*
* *When a 503 is sent into SDIN from the recipient SIP operator, or there is Trunk Network congestion in the TDM layer, and that in either case BT is unable to reroute the call to the range-holding operator, the SBC sends “SIP/2.0 503 Service Unavailable”*

*503s can therefore be an indication that additional capacity is required, but not in all cases.*

As part of a software upgrade being rolled out across BT’s IP Exchange SBCs during Autumn 2018, policing of bandwidth used by video streams is being enabled. Policing of video bandwidth occurs at two levels:

* an aggregated constraint policing all video streams per SBC of 250Mbps.
* a constraint per video session of 0.5Mbps.

In early 2019 bandwidth policing will be enabled on audio streams. This will police the bandwidth based on the expected requirements of the negotiated audio. Ptime values are taken into account when determining bandwidth requirements. If no ptime is specified in the SDP then a default ptime value of 20ms is used.

### SIP Error Code Mapping

***SIP to ISUP error code mapping***

For Break In calls, to ensure that the correct ISUP error codes are returned to the calling party when the calling party originates on the PSTN, the CP will return the following error codes for the failure conditions specified:

|  |  |
| --- | --- |
| **SIP Response received by IP Ex** | **IP Ex SIP-ISUP Mapping** |
| 400 Bad request | 95 Invalid message |
| 403 Forbidden | 63 service/option not available |
| 404 Not found | 1 Unallocated No. |
| 406 Not acceptable | 79 Service or option not implemented |
| 408 Request timeout | 18 no user responding |
| 409 Conflict | 41 Temporary Failure |
| 410 Gone | 22 Number changed |
| 413 Request Entity too long | 111 protocol error |
| 414 Request-URI too long | 111 protocol error |
| 415 Unsupported media type | 79 Service or option not implemented |
| 416 Unsupported URI Scheme | 127 Interworking |
| 480 Temporarily unavailable | 31 Normal, unspecified |
| 481 Call leg/transaction | 95 Invalid message |
| 482 Loop detected | 25 Exchange rtig error |
| 483 Too many hops | 25 Exchange rtig error |
| 484 Address incomplete | 28 Invalid No. format |
| 485 Ambiguous | 1 Unallocated number |
| **486 Busy here** | **17 User busy** |
| 487 Request Cancelled | 31 Normal, unspecified |
| **488 Not acceptable here** | **88 Incompatible destination** |
| 500 Internal server error | 47 resource unavailable |
| 502 Bad gateway | 111 protocol error) |
| 503 Service unavailable | 42 switching equipment congestion (EET) |
| 504 Server time-out | 102 Recovery on timer expiry |
| 505 Version not supported | 127 Interworking |
| 513 Message Too Large | 111 protocol error (NU) |
| 600 Busy everywhere | 17 User busy |
| 603 Decline | 21 Call rejected |
| 604 Does not exist anywhere | 4 send special information tone (NU) |
| 606 Not acceptable | 79 Not implemented, unspecified (174) |

*Values conform to the mappings in NICC ND1017 [10] except where marked in bold.*

***ISUP to SIP Error code mapping***

For Break Out calls to the PSTN via the IP Exchange SDIN gateways, BT will return the following error codes for the failure conditions. The ISUP to SIP mappings are designed to allow calls to be retried via alternative routes where there is a reasonable chance that the reattempt would succeed.

For definitions of the location values see ITU-T Rec. Q.850. (NP = Not Present)

|  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
|  | Location Value | **NP** | **LPN** | **LN** | **TN** | **INTL** | **RLN** | **RPN** | **U** | **BI** |
| 1 | UNALLOCATED NUMBER | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 |
| 2 | NO ROUTE TRANSIT NET | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 |
| 3 | NO ROUTE DEST | 503 | 404 | 404 | 503 | 503 | 404 | 404 | 404 | 503 |
| 4 | SEND SPCL TONE | 604 | 604 | 604 | 604 | 604 | 604 | 604 | 604 | 604 |
| 5 | MISDIALED TRUNK | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 |
| 8 | PREEMPTION | 503 | 480 | 503 | 503 | 503 | 503 | 480 | 480 | 503 |
| 9 | PREEMPT RESRV USE | 503 | 480 | 503 | 503 | 503 | 503 | 480 | 480 | 503 |
| 14 | QUERY ON RELEASE | 410 | 410 | 410 | 410 | 410 | 410 | 410 | 410 | 410 |
| 16 | NORMAL CALL CLEARING | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 |
| 17 | USER BUSY | 486 | 486 | 486 | 486 | 486 | 486 | 486 | 486 | 486 |
| 18 | NO USER RESP | 408 | 408 | 408 | 408 | 408 | 408 | 408 | 408 | 408 |
| 19 | NO ANS FROM USER | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 |
| 20 | SUBSCRIBER SABSNT | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 |
| 21 | CALL REJECTED | 603 | 603 | 603 | 603 | 603 | 603 | 603 | 603 | 603 |
| 22 | NUMBER CHANGED | 410 | 410 | 410 | 410 | 410 | 410 | 410 | 410 | 410 |
| 23 | UNALLOC DEST NUMBER | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 24 | UNKNOWN BUSINESS GRP | 433 | 433 | 433 | 433 | 433 | 433 | 433 | 433 | 433 |
| 25 | EXCHANGE ROUTING ERROR | 483 | 483 | 483 | 483 | 483 | 483 | 483 | 483 | 483 |
| 27 | DEST OUT OF ORDER | 503 | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 |
| 28 | INVALID NUMBER FORMAT | 484 | 484 | 484 | 484 | 484 | 484 | 484 | 484 | 484 |
| 29 | FACILITY REJ | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 31 | NORMAL UNSPECIFIED | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 | 480 |
| 34 | CIRCUIT CONGEST | 503 | 486 | 486 | 503 | 503 | 486 | 486 | 486 | 486 |
| 38 | NETWORK OUT OF ORDER | 503 | 500 | 500 | 503 | 503 | 500 | 500 | 500 | 500 |
| 41 | TEMPORARY FAILURE | 503 | 480 | 500 | 503 | 503 | 500 | 480 | 480 | 500 |
| 42 | SWITCH CONGEST | 503 | 486 | 503 | 503 | 503 | 503 | 486 | 486 | 503 |
| 43 | ACCESS INFO DISCARD | 503 | 500 | 500 | 500 | 500 | 500 | 500 | 500 | 500 |
| 44 | REQ CHANNEL UNAVAIL | 503 | 500 | 500 | 503 | 503 | 500 | 500 | 500 | 500 |
| 46 | PRECEDENCE CALL BLKD | 503 | 500 | 500 | 503 | 503 | 500 | 500 | 500 | 500 |
| 47 | RESOURCE UNAVAIL | 503 | 500 | 500 | 503 | 503 | 500 | 500 | 500 | 500 |
| 50 | FAC NOT SUBSCRIBED | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 53 | OUT CALL BARRED CUG | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 55 | INC CALL BARRED CUG | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 57 | BC NOT AUTHORIZED | 488 | 488 | 488 | 488 | 488 | 488 | 488 | 488 | 488 |
| 58 | BC UNAVAILABLE | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 62 | INCONSISTT OUT ACC INFO | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 63 | SERVICE UNAVAILABLE | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 65 | BC NOT IMPLEMENTED | 501 | 501 | 501 | 501 | 501 | 501 | 501 | 501 | 501 |
| 69 | FAC NOT IMPLEMENTED | 501 | 501 | 501 | 501 | 501 | 501 | 501 | 501 | 501 |
| 70 | ONLY RSTR DIGITAL AVAIL | 488 | 488 | 488 | 488 | 488 | 488 | 488 | 488 | 488 |
| 79 | SERVICE NOT IMPLEMENTED | 501 | 501 | 501 | 501 | 501 | 501 | 501 | 501 | 501 |
| 87 | NOT MEMBER CUG | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 | 403 |
| 88 | DEST NOT COMPATIBLE | 488 | 488 | 488 | 488 | 488 | 488 | 488 | 488 | 488 |
| 90 | NONEXISTENT CUG | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 |
| 91 | INVALID TRANSIT NET SEL | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 | 404 |
| 95 | INVALID MESSAGE UNSPEC | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 |
| 97 | MSG TYPE NOT IMPL | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 |
| 99 | IE NOT IMPL | 503 | 502 | 502 | 503 | 503 | 502 | 502 | 502 | 502 |
| 102 | TIMER RECOVERY | 504 | 504 | 504 | 504 | 504 | 504 | 504 | 504 | 504 |
| 103 | PARAMETER NOT IMPL | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 |
| 110 | PARM DNE UNIMPL DISCARD | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 |
| 111 | PROTOCOL ERR UNSPEC | 503 | 502 | 502 | 503 | 503 | 502 | 502 | 502 | 502 |
| 127 | INTERWORKING UNSPEC | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 | 502 |

|  |
| --- |
| Cause code that maps to 503 and allows hunting |

### Circuit suspend/resume and Re-Invites

Calls to POTS lines via BT’s PSTN conform to the same *calling party clears* behaviour exhibited by PSTN to PSTN calls. The call is cleared down only when the calling party goes on hook, so that the party paying for the call controls when it ends. This feature would allow the called party to transfer the call from one extension to another within the same house, in a manner known as suspend/resume. (The practical use of suspend/resume in the PSTN has been vastly reduced in order to prevent fraud, so the following has become less relevant.)

CPs on SDIN can choose from two variants of SUS/RES treatment: (i) as per the description below; and (ii) where the ISUP SUS/RES is absorbed by SDIN and not mapped into any SIP signal(s).

When the called party goes on hook, a SUS signal is generated by the network and sent through to the IP Exchange PSTN gateway. This message is translated to a Re-Invite message signifying the call has been put on hold, as described in section 9 of RFC 3398 [18]. If the called party resumes the call the RES signal results in another Re-Invite that re-establishes the call at the IP level. The distinguishing feature about these Re-Invite messages is the change of the media direction attribute, as per SIP “hold” described in RFC 3264 [4] section 8.4.

If the called party remains on hook, and the calling party does not clear down the call, then the CSH timer in the PSTN will expire, resulting in the REL signal. The PSTN gateway maps this to the SIP BYE request which is forwarded to the call originator, thus terminating the call.

CPs wishing to release the line and proceed to the next call, on the assumption that a call which has been placed on hold will not be resumed, can treat the Re-Invite as a signal to terminate the call.

The duration of the CSH timer, and even which party can clear the call, can vary depending on which carrier is terminating the call. BT (and most other CPs) have now changed the timer at the called end to be 2 seconds in order to combat certain fraudulent calls. This means the “move rooms” scenario described above is no longer viable. It also means that in reality the call releases much more often from the called end than it used to, and so race conditions where the release messages from either end pass each other are more common.

A Re-Invite indication of SIP “hold” received by IP Exchange is not mapped into a ISUP SUS signal.

### SIP hold IP<>IP

Due to the variation in CP behaviour and interpretations of RFC3264 [4] and RFC6337 [19], BT IP Exchange recommends that CPs follow the guidance and recommendations in RFC6337 [19]. A common approach across the majority of CPs will reduce the risk of calls potentially becoming stuck on hold with no way of reinstating 2 way speech.

### Bearer rearrangement

As a result of call transfer by means of Re-INVITES, CP sometimes request rearrangement of bearers associated with the calls. Where a CP wishes to rearrange bearers to connect two call legs together, it is preferred that this is be carried out in the CPs own network.

### Pinging end-points using SIP OPTIONS

BT’s SBCs use the SIP OPTIONS request to “ping” a CP’s proxy addresses to establish whether the end-point is available or not. Any value of SIP response that is returned is taken as confirmation that the end-point is available. If there are sufficient volumes of Invites through to the CP, the transmission of Options requests are suspended.

If BT’s SBC marks an endpoint as out of service, INVITE messages will no longer be forwarded to it. OPTIONS requests will continue to be sent, and the response to the OPTIONS request will cause the SBC to mark that endpoint as back in service.

CPs can send Options requests to confirm the availability of the IP Exchange SBCs. Out of dialogue Options requests sent to BT’s IP Exchange platform will receive a 200 OK.

ICMP ping requests are deliberately not supported by IPX as part of DOS attack prevention

### Max-Forwards

The Max-Forwards header field serves to limit the number of hops a request can transit on the way to its destination. The recommended minimum value for Max-Forwards in a request being forwarded into IP Exchange is 50. RFC3261 [1] recommends the an originating UAC include a value of 70 in the original Invite, and the value of 50 allows for some of the hops to have been consumed before the Invite arrives at IP Exchange whilst allowing a reasonable number for routeing through the IP Exchange and any subsequent SIP networks.

## Loudness Levels

The CP shall comply with the loudness standards ETSI TBR 8, and the CP's CPE must be able to support the following loudness rating for calls to/from IP Exchange: SLR=7dB, RLR=3dB where SLR is the sending loudness rating and RLR is the receive loudness rating

## Maximum call duration

Assuming no other factors are involved in terminating a call, the maximum duration of a call on IP Exchange is capped at 1 week (604800 seconds). This is a safety net, as individual service platforms run their own shorter timers.

## VOID

## Direct-access interface specification for IP Exchange on SDIN

### IP QoS labels

Differentiated Services Code Point (DSCP) values in the IP header help to classify the nature of the contents of the IP packets so that they can be prioritised according to whether the content can tolerate delay, jitter, packet loss etc. Details are included in RFC 5865 [20] (previously RFC4594 applied). The IP Exchange service expects that packets being sent by IP Exchange or GIPX CPs into any of the points of interconnect listed in appendix A (the IP Exchange NAP or one of BT’s IPX 21C nodes) are classified as follows:

* Signalling: AF31 or AF41 (AF11 or AF21 would also be acceptable although less preferred)
* Media: EF

This ensures that the above two traffic types are not impacted by occasional surges in “best efforts” traffic such as web browsing or file downloads that coincide with a loss of resilience on BT’s 21C core network. If CPs do not mark packets in this manner, BT cannot provide assurances over the consistency of the associated IP Exchange service.

### Connections via a Neutral Access Point (NAP) – Telehouse North, London

#### Physical Layer

The point of connection within the IP Exchange room in the NAP is a HA-pair of Access Ethernet Switches (AES) in which SFPs are installed to support the following options:

* 1000BaseSX (Multi-mode fibre)
* 1000BaseLX (Single-mode fibre)
* 1000BaseT

Fibre Gigabit Ethernet connections should be presented with a LC connector.

A CP may connect into one or both AES, depending on the resilience required and what the CP can support.

Auto-negotiation for both speed and duplex setting will be configured by default on all 1000BASE-T connections, which should usually negotiate to speeds less than 1000Gbps if necessary.

### Connections within the NAP

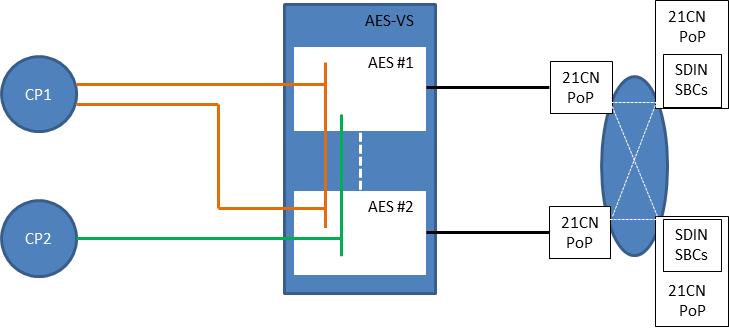
The CP is responsible for ordering (and maintaining) the circuits between BT and the Direct Access CP. The BT Technical Account Manager will advise the CP of the exact point (location, port number) that the circuits must be terminated on. This will be the demarcation point between BT’s network and the CP’s network.

When ordering the in-building cable through to the BTIS room in Telehouse North, please request an LC/PC connector type and the cable to be left coiled as it enters the room with sufficient slack to reach cabinets C00, C01 and C02 from the cable ingress point. Telehouse will request access to the room normally from the person who placed the order. This should be forwarded to svi.provide.g@bt.com who will obtain approval and forward back to Telehouse.

When the CP receives notification from Telehouse that the cable has been provided and is sending light, please inform the BT Technical Account Manager who will then progress the request into the SVI provide G team for an engineer to complete the cable connection.

Telehouse might recommend a particular OM rating for multimode fibre cable within the building based on the length required. BT recommends a minimum of OM2.

#### Logical layer



To connect IP Exchange via the NAP the CP must provide a subnet of RIPE addresses (minimum /30[[2]](#footnote-2)) from which an address will be assigned on the AES switches. This acts as the gateway address into the CP’s network.

Each CP may connect to one or more switch ports on the pair of Access Ethernet Switches. The available connection types are:

* Single switch port
* Pair of switch ports that can be configured to use either:
  + 802.1w - Rapid Spanning Tree Protocol backwards compatible with 802.1D
  + 802.3ad - LAG using Fast LACPDUs (timeout = 3s)

Load sharing over multiple connections only really works with the LAG set-up. The rapid spanning tree configuration would tend to operate as active/standby.

No layer 3 option (e.g. eBGP) is currently available.

The port on the BT’s switch must be presented as an untagged switch port. BT will not support trunked / dot1q tagged interconnects.

### Connections via a BT POP in the UK

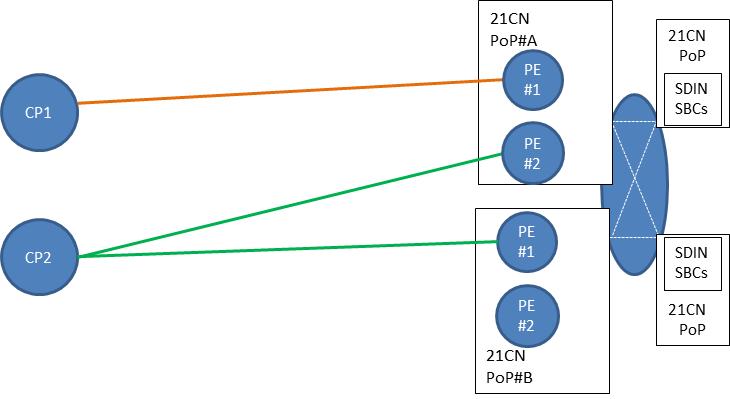
#### Physical Layer

The point of connection within the BT POP is a cable/connector provided by BT to the CP’s NTE within in the MUA room. Only 1000BaseLX (single-mode fibre) is supported. Connector types can be LC or SC.

A CP will typically connect into a single PE router via the MUA ANC. If resilience is required a separate connection to a second PE router can be established, either within the same or different BT POP.

CPs are responsible for providing the point of interconnect within the MUA to their own network.

#### Logical layer



To connect to IP Exchange via a BT POP, the CP must provide a subnet of RIPE addresses (minimum /30[[3]](#footnote-3)) from which an address will be assigned on the PE routers. This acts as the gateway address into the CP’s network. It is expected the lower address of the subnet to be allocated to the BT PE.

The port on the PE must be presented as a dot1q tagged sub-interface; the VLAN ID is provided by BT. For POP connections in the UK this will be in the range of 700-799.

No dynamic routeing is run over this interface – all routeing is provided via static routes redistributed into the IP Exchange VPN by BT. It is possible to provide an active/standby arrangement via the use of a second connection and distribution of static routes with a lower preference by BT.

### Multiple NAP and POP connections in the UK (SDIN only)

A CP connecting via a NAP or a POP communicates with a SBC in BT’s IP Exchange network through BT’s 21C network. The physical location of the SBC is not restricted by the CP’s inter-connect point. Once configured, there is a relation defined between that specific access method and the SBC.

For CP’s connecting via both NAP and POP, for example a single NAP link and a single POP link, BT IP Exchange can configure POP as 1st choice and NAP as 2nd choice for BT>CP traffic. An active/active load share arrangement is not possible.

# References

[1] RFC 3261 SIP: Session Initiation Protocol

[2] RFC 4566 Session Description Protocol

[3] RFC 3262 Reliability of Provisional Responses in the Session Initiation Protocol (SIP)

[4] RFC 3264 An Offer / Answer Model with SDP

[5] RFC 4733 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals

[6] RFC 3325, Private Extensions to SIP for Asserted Identity within trusted networks

[7] RFC 6086 The SIP INFO Method

[8] RFC 4028 Session Timers in the Session Initiation Protocol

[9] RFC 3323 A Privacy Mechanism for the Session Initiation Protocol (SIP)

[10] ND1017 Interworking between Session Initiation Protocol (SIP) and UK ISDN User Part (UK ISUP)

[11] ND1016 Requirements on Communications Providers in relation to Customer Line Identification display services and other related services

[12] ND1035 SIP Network to Network Interface Signalling

[13] ND1439 Guidance for Implementing ND1016 in SIP networks

[14] Annex 2: Updated guidance on the provision of Calling Line Identification facilities and other related services (ofcom.org.uk)

[15] ND1657 SIP – Overload Control

[16] RFC 7544 Mapping and Interworking of Diversion Information between Diversion and

History-Info Header Fields in the Session Initiation Protocol (SIP)

[17] RFC 4040 RTP Payload Format for a 64 kbit/s Transparent Call

[18] RFC 3398 Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping

[19] RFC 6337 Session Initiation Protocol (SIP) Usage of the Offer/Answer Model

[20] RFC 5865 (previously RFC4594) A Differentiated Services Code Point (DSCP) for Capacity-Admitted Traffic

[21] RFC 4960 Stream Control Transmission Protocol

[22] RFC 3021 Using 31-Bit Prefixes on IPv4 Point-to-Point Links

1. Direct access sites

|  |  |  |  |
| --- | --- | --- | --- |
| **BT UK POPs for IP Exchange on SDIN** | | | |
| ***Birmingham\****  Telephone House  104 Newhall Street  Birmingham  B3 1BA | ***Glasgow\****  Dial House  57 Bishop Street  Glasgow  Lanarkshire  G3 8UE | ***Manchester\****  Dial House  21 Chapel Street  Salford  Manchester  M3 7BA | ***Leeds\****  Basinghall TE  14 Butts Court  Leeds  LS1 5JS |
| ***London\****  Faraday House  1 Knightrider Street  London  EC4V 5BT | ***London\****  Colindale SSC  The Hyde  Edgware Road  London  NW9 6LB | ***London\****  Colombo House  Joan St  London  SE1 8NZ | **London\***  Upton Park ATE  473 Green Street  London E13 9AX  *(subject to capacity survey)* |
| ***Milton Keynes\****  ATE  Victoria Rd  Bletchley  Milton Keynes  MK2 2PA | ***Bristol\****  CTE  Marsh St  Bristol  BS1 4AY | ***Hornchurch\****  Hornchurch ATE,  64 North Street, Hornchurch,  Essex, RM11 1SS | ***Sheffield\****  Eldon House  Charter Row  Sheffield  S1 3EF |
| **Slough\***  Slough ATE  100 Wellington Street  Slough SL1 1YW | **Wolverhampton\***  Wolverhampton ATE Red Lion Street  Wolverhampton  WV1 1SR |  |  |

\*Please check direct-access availability at the POP with your Technical Account Manager.

|  |
| --- |
| **Neutral access point** |
| Rack C0 (AES 1), rack C2 (AES2), BTIS room, 3rd floor  London Docklands North Building,  Coriander Avenue,  London E14 2AA |
| *The East building is on the same street, but BT’s main presence is in the North building* |

**Access to London Telehouse North**

Access to the BTIS room in Telehouse north must be arranged and authorised in advance. Cabling into the BTIS room can be via approved Telehouse personnel, or a third-party supplier. The latter must be overseen by BT personnel. Physically connecting the cable to the IP Exchange equipment in Telehouse will be done by BT personnel.

The BT Technical Account Manager will be able to advise on the logistics of connecting to IP Exchange.

|  |
| --- |
| **BT Global POPs for IP Exchange on SDIN** |
| **Bahrain Global IPx POP** Latitude / Longitude: 26.192239 / 50.551526  1st Floor  Manama-Viva  Road No 1330  Manama  Bahrain |

1. VOID
2. Transcoding/transrating within the SDIN platform.

Wherever possible end-to-end negotiation will be used to choose a common set of codec/ptime options for all parties involved in the call so that transcoding is avoided for IP-IP calls. The media capabilities supported by each CP on IP Exchange are captured in the CRF and this forms part of the routeing decisions performed by the SDIN routeing engine. If the routeing engine detects that there are no common codecs shared between the originating and destination parties then the transcoder is brought into the call-flow before making the initial attempt to the destination party. Where the capabilities supported by the originating and destination parties share common attributes, no transcoding is necessary.

Within the same Service Options section of the CRF the recorded value for ptime is also used to determine whether transcoding resource is required up front (proactive transcoding/transrating decision). As AMR-WB intrinsically has a packetisation delay of 20ms, the recommendation for CPs wanting to use HD codecs (and when appropriate, enable transcoding to other HD codecs) is to select 20ms.

If the offer/answer results in a codec mismatch, and a “488 Not Supported Here” response is returned, then the platform will retry the call having introduced a transcoder into the call-flow (reactive transcoding/transrating decision). Transcoding resource is only engaged when the call is initially set up. A mid-call codec change that attempted to alter the session to a mutually incompatible codec would not result in transcoding.

SDIN will pass through any voice codec (including HD) for direct on-net calls where both A & B parties:

* Are configured with same preferred packetisation time

and either of the following

* Mutually support said voice codec
* Are configured to mutually support one of the following codecs:
  + G.711A
  + G.711u
  + G.729

The transcoding capability on the SDIN platform is available between any of the codecs/ptimes supported by the SDIN platform for Break Out/Break In calls. These codecs are listed in the table in section 3.4.10.

1. End-to-end resilience of the IP Exchange platforms

End-to-end resilience of the IP Exchange platforms supporting the UK IPX and GIPX products



1. Briefing to CPs concerning the 2nd legs of emergency calls



1. Briefing to CPs concerning OFCOM GC C6 and CLI



1. Glossary of terms & abbreviations

|  |  |
| --- | --- |
| **Term** | **Explanation** (for further details see Wikipedia or the relevant industry standards document such as RFC3261 [1] for SIP) |
| AES | Cisco’s Access Ethernet Switches deployed by BT in Telehouse North to terminate customer 1GB access circuits. Connectivity is then provided by a number of resilient 10GB backhaul circuits to BT’s 21C network, providing access to all of BT’s IPX/GIPX SBCs. |
| APRI | Address Presentation/Restriction Indicator – markers used to indicate whether CLIs can be displayed or passed on to subsequent networks or devices. |
| BIBO | Break In / Break Out service:  Break-in (BI): A call from BT’s PSTN/TDM/NGS layer to SDIN (e.g. IPX).  Break-out (BO): A call from SDIN (e.g. IPX) to the PSTN/TDM/NGS layer.  “On-net” covers the third type of call on SDIN/IPX which covers all IP to IP calls. |
| BGP | Border Gateway Protocol – not available as a standard access option for IPX/GIPX, and not available for connectivity through the SDIN London Telehouse NAP. |
| CAC | Call Admission Control – limits the number of sessions on an SBC for a customer, either by call count, bandwidth assigned to those calls, or both. |
| CDR | Call Detail Record, used for billing and reporting |
| CLI | Calling Line Identity – the identity (a phone number) of the line originating a voice session. On SDIN, there are normally two CLIs per call – the PAID (aka network CLI used for locating callers for various purposes) and the FROM (aka presentation CLI, what the caller wants displayed to the calling party to let them know who is calling, and who to call back if they want to). |
| CP | Communications Provider – used in this document as a generic term for customers, suppliers, internal services or any other “operator” provisioned on IPX or GIPX. |
| CPN | Called Party Number, also CdPN |
| CgPN | Calling Party Number, see also CLI. |
| CPS (CAPS) | Calls Per Second or (depending on context) Carrier Pre-Select.  The latter is similar to IDA, a prefix used for routing calls into IP Exchange from the PSTN which is retained as the calls pass through IP Exchange to the CP. Calls Per Second (CPS) can also more accurately be known as Call Attempts Per Second (CAPS), since it includes ineffective call attempts. |
| CRE | Core Routeing Engine, provides a routing database for the SBCs and MGCs to perform SIP redirect queries to determine where the call should be routed, primarily based on the dialled number. It normally provides a number of routing options, either to on-net destinations or to the PSTN. |
| CRF | Customer Requirements Form, stating how much capacity is provisioned, what access methods are used to connect the customer to BT, and what specific configuration settings are being used for provisioning the customer on the network. During provisioning, IP addresses (BT and customer) are added. The latest version of each CRF is held and maintained by the BT TAMs. |
| DA | Direct Access – connectivity between a customer and the SDIN network for IPX/GIPX service that involves a specially-provisioned physical link (normally 1GB) into either a BT 21C POP or the SDIN NAP in Telehouse North. |
| E.164 | The normal format for phone numbers used to determine where a call needs to be routed to, and where it has come from. It includes the country code at the front, excludes any leading zeros, and is the normal length for standard geographic or non-geographic numbers. 441442208902 would be an example of a geographic number in the UK. “+” is used at the front of E.164 numbers to signify that the next digit or digits are the country code. Non-E.164 numbers include those that start with country-specific prefixes, or short codes such as 999 for the UK emergency services. |
| FQDN | Fully Qualified Domain Name (with relation to DNS type responses) |
| GIPX | Global IPX - The name of the product variant of BT’s IPX service normally sold to customers outside the UK to reflect different feature sets, e.g. the UK emergency services are inappropriate for calls from other countries, therefore are not available. |
| GN | Generic Number – another variant of CLI sometimes present in addition to PAID and From CLIs. Also known as Additional Calling Party Number (ACgPN) |
| GVA | Generic Voice Access –the device present in each IPX/GIPX global POP for connecting customer access circuits and other circuits into the POP. |
| HA | High Availability – two SBCs running as master-standby, able to failover without dropping any call in progress and without losing any CDRs. The term also applies to other SDIN network components. |
| HD | High Definition in relation to voice sessions – using codecs that offer better voice quality than other codecs, without always consuming more bandwidth. |
| IDA or IA | Indirect Access code. Used to get a call to a particular transit operator from where either the remainder of the called number, or digits dialled in-band, are used to route the call to its destination. |
| IPX | IP Exchange – in this document, used to cover both the UK IPX and the Global IPX (GIPX) VOIP services supported by BT’s SDIN platform, carrying voice sessions, typically controlled using SIP. |
| ITFS | International Toll Free Service |
| ITU | The International Telecommunication Union – an agency of the United Nations that issue standards governing how telecommunications networks work, e.g. to help ensure that various devices across various networks can run SIP voice sessions successfully. |
| LCR | Least Cost Routeing - implemented in different ways different network components. |
| LDLI | Last Diverting Line Identity |
| MGC | Media Gateway Controllers, controlling signalling between TDM and IP, and controlling media gateways. Also provides signalling interworking (e.g. SIP to SIP-I), and involved in the call set-up for calls requiring transcoding or other interworking function. On SDIN, these are Ribbon C3s. |
| MGW | Media Gateways, converting between TDM and IP, or providing media interworking, e.g. transcoding. On SDIN, these are Ribbon G9s |
| MUA | Multi User Area – typically the basement of BT telephone exchanges. To make use of the IPX POP connect process (ordering a direct access 1GB link o buildings in the UK other than Telehouse North), CPs need to arrange for a circuit to be provisioned from their premises to one of the IPX POP connect buildings, terminating in a suitable cage in the MUA area. The POP connect process provides the onward connectivity from there to BT’s 21C access switch in the same building. |
| NAI | Nature of Address Indicator – whether a CLI (or other called or calling number) is in international format, national format or other format (e.g. an IDA code or number with porting prefix added). |
| NAP | Neutral Access Point – SDIN’s NAP is in London Telehouse (North). It is “neutral” because it is neither owned by BT nor the CP. |
| NAPTR | Naming Authority Pointer (with relation to DNS responses), normally a URI. |
| NP | Number Portability |
| NW | Network |
| OM rating | Optical Multi-mode - A means of identifying the usable bandwidth of a multi-mode fibre, e.g. OM1, OM2, OM3 etc, standard ISO 11801 refers. |
| PAID | P-asserted IDentity – the SIP equivalent of the network CLI. This must uniquely identify the physical site where the call originated, so no non-geo numbers are allowed. CPs sending PAIDs into IPX must “assert” the credibility of this information, since for example it is used to locate callers using the 999 or other emergency calls. |
| PDD | Post Dial Delay – the length of time between the last digit of a phone number being dialled and the commencement of ringing tone (or other appropriate tone) being played to the caller. |
| PNS | Personal Numbering Services – a type of number that can be bought by an individual, and then calls to that number are directed around the network to the individual’s location, whether that is on a mobile network, a standard fixed line or a VOIP network, etc. Charges for calls to such numbers may differ from other non-geographic or geographic calls. These typically start with 4470 (or 070 if dialled from the PSTN). |
| POP | Point Of Presence – a building that provides access to the IPX/GIPX service. It may have SBCs or other IPX components in it, or just an access switch that is linked to other IPX sites via BT’s 21C access layer. |
| POTS | Plain Old Telephone Service – a basic TDM PSTN phone line. This covers most conventional TDM phone lines that are not ISDN. |
| PSTN | Public Switched Telephone Network – used in this document to identify BT’s 20C telephone network. |
| PE router | Provider Edge router, in this context the routers in selected BT buildings that provides CPs with access to BT’s 21C layer, which in turn connects to every BT node containing IPX/GIPX SBCs. |
| Ptime | The packetisation interval for transmitting media. See “transrating”. |
| RIPE addresses | RIPE stands for Réseaux IP Européens (RIPE, French for "European IP Networks").  “RIPE addresses” means those assigned by RIPE for unique use to identify a specific device within an operator’s network when routing IP packets across network boundaries. |
| SAC | Service Address Code – prefix used for routeing within a network, often starts with 944. Also known as Network Address Code. |
| SBC | Session Border Controller. On SDIN, these are Ribbon S3s (Q20 or Q21s). These can be customer-facing (CP-SBCs - separating BT’s core from the UK or non-UK access network), Global Borders (separating BT’s core from BT’s global MPLS network) or proxy SBCs (used for CRE lookup within SDIN). They can provide a number of features including CDR generation, hiding the structure of one network from being visible to another, setting limits to the capacity available to each CP (CACs), etc. |
| SCTP | Stream Control Transmission Protocol (RFC 4960 [21]) is a transport-layer protocol, serving in a similar role to the popular protocols TCP and UDP, and is not supported on IPX/GIPX. |
| SDIN | Session Distribution & Interworking Network, largely comprised of Ribbon voice components built on the 21C core network, handling VOIP sessions. |
| Service codes & Short codes | Telephone numbers that are much shorter than standard geographic or non-geographic numbers, and connect the caller to a specific service, e.g. 123 for the speaking clock, 118xxx for directory enquiry services etc. They are normally only relevant/available for calls originating in the UK. |
| SIP, SIP-I | Session Initiation Protocol – used to set up, clear down, and manage in-call features for VOIP sessions. SIP-I is an extension of SIP that handles additional features commonly used in ISDN calls by embedding the ISDN-specific signalling within SIP. The vast majority of CPs on IPX/GIPX use SIP, only a small majority use SIP-I. SDIN handles interworking for calls between SIP and SIP-I CPs, but of course the ISDN features are lost. (ISDN = Integrated Services Digital Network, a type of TDM phone line that offers more lines and more features than a standard single line.) |
| TAM | Technical Account Manager – the team of people within BT that handle provisioning (including CRFs) and other customer comms. Their group email address is [interconnect.team@bt.com](mailto:interconnect.team@bt.com) |
| TCP | Transmission Control Protocol (TCP) is one of the main protocols of the Internet protocol suite, and provides reliable, ordered, and error-checked delivery of a stream of octets between applications running on hosts communicating by an IP network. (source: Wikipedia) It is not available for Signalling transport on IPX/GIPX. |
| TDM | Time Division Multiplexing – used in the PSTN for circuit mode communication with a fixed number of channels and constant bandwidth per channel. |
| TLS | Transport Layer Security - a cryptographic protocol that provides communications security over a computer network. |
| TNOR | Terminating Network Operator Reference – used for billing, CRE routing and a number of other purposes in SDIN for IPX and GIPX CPs. |
| Transcoding | A means of Translating a call that uses one codec (the means by which the voice or other sound in a phone call is translated into 1s and 0s) into another codec, for example to ensure that a call between a CP that only uses G.711 codec can be relayed to another that only uses G.729. SDIN does transcoding either proactively – when it realises that the originating CP and the terminating CP have no codec in common – or reactively for example when the terminating CP sends back a 488 SIP response. |
| Transrating | Similar to Transcoding except where the originating and terminating CP uses different packetisation – the means by which the data for a voice session is split up into packets that contain data for a given small slice of the call, for example if both CPs are using G711 but that codec is packetized at 10ms intervals by one CP and 20ms for another. The parameter that sets the interval (10ms, 20ms etc.) is called “ptime”. |
| UAC | User Agent Client – in this context, the CP’s SIP device, either an SBC or equivalent device that handles SIP requests and responses when setting up, clearing down or managing SIP voice sessions to/from IPX/GIPX. |
| UDP | User Datagram Protocol (UDP), a transport protocol which provides a connectionless datagram service that is used by IPX/GIPX for reduced latency. |
| URI | Uniform Resource Identifier |
| VOIP | Voice Over IP (Internet Protocol) – a means of transmitting phone calls over an IP network. |
| WES | Wholesale Ethernet Service transmission links providing Gigabit Ethernet connectivity between two sites. |

1. [↑](#footnote-ref-1)
2. Support for a /31 subnet as described in RFC 3021 [22] might be possible – please confirm with your technical account manager. [↑](#footnote-ref-2)
3. Support for a /31 subnet as described in RFC 3021 [22] might be possible – please confirm with your technical account manager. [↑](#footnote-ref-3)