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Colt VoIP Access Service Guide

Note: This document is not legally binding

The information delivery platform for European business

1 Overview

Colt VoIP Access is suitable if you own and/or operate an IP Private Branch Exchange (IP PBX) on your premises and want to exploit the advantages of IP Telephony in the WAN and IP connectivity to the Public Switched Telephone Network (PSTN).

VoIP Access provides a connection to Colt's centrally located PSTN gateways which allow calls to pass between your IP PBX and the PSTN. We also provide you with a geographic number range for use with the service.

As you won't need to install your own on-site PSTN gateways, with Colt VoIP Access your organisation can benefit from significant cost savings in hardware requirements. Additionally, cost savings may also be realised in terms of management and administrative overheads.

2 Benefits

2.1 Operational efficiencies through voice and data convergence

Colt VoIP Access enables your organisation to converge its voice and data traffic onto a single end-to-end IP network. This allows you to minimise the Infrastructure required to carry your voice calls.

2.2 Control over costs

Colt VoIP Access allows voice traffic to be consolidated and delivered to the PSTN over high capacity Wide Area Network (WAN) links. This reduces the requirement for multiple dedicated ISDN access circuits from every site and reduces the cost of Primary Rate Interface (PRI) and/or Basic Rate Interface (BRI) hardware interfaces.

Colt VoIP Access provides competitive call rates to the PSTN and mobile networks, with access delivered across Colts multiple European PSTN switches.

2.3 Security and reliability

Colt VoIP Access infrastructure is housed in Colt network nodes, which provide a physically secure environment for the service. Network components are continuously monitored, and the service is backed by comprehensive Service Level Agreements (SLA). A number of options exist for providing resiliency in the WAN so you can have utmost confidence in service availability and reliability.

The service has been designed to perform at a target 99.999% availability (VoIP Access infrastructure only not including access elements).

2.4 Integration and management

Colt VoIP Access forms the basis for an integrated voice solution for your organisation. It does not require you to change your IP PBX hardware, and hence disruption to end users across your telephony estate is kept to a minimum.

3 Design Considerations

3.1 Pre-requisites

You will need to have an IP PBX or an IP enabled PBX in order to use the Colt VoIP Access service. Time Division Multiplex (TDM) PBX equipment is not currently supported on Colt VoIP Access but is on other Colt services.

Colt VoIP Access supports the native H.323 and SIP signalling protocols. Since many equipment vendors have introduced their own proprietary extensions to the H.323 and SIP protocol, we only support Colt VoIP Access with IP PBX's that have been compatibility tested by Colt. Colt has interoperability tested a number of popular IP PBXs and software releases and is constantly updating this list with a pro-active test programme. Details of IP PBXs which have been compatibility tested can be provided by your Colt sales representative.

Compatibility testing verifies basic interoperability for a given software version, it does not necessarily mean all features and functions can interoperate to the PSTN via VoIP Access.

PBX's and software versions not covered by our compatibility testing need to be assessed on a case by case basis–please contact your Colt representative to see how your requirements can be met.

The following table summarises important considerations which must be observed for successful configuration of IP PBX with the Colt VoIP Access service:

SIP based connection:

Dialing out	Enbloc operation only is supported
Fax	Fax Over IP is supported using the ITU T.38 standard but is subject to the limitations specified elsewhere in this document.
Codec Parameters	G.729 & G.729a-20ms sample rate G.711–20ms sample rate
DTMF	RFC2833
Other Bearer capabilities	Modem calls can be terminated but are subject to the limitations specified elsewhere in this document. 64Kbps Unrestricted Data is not supported

H.323 based connection:

Dialing out	Enbloc and overlap dialling operation only is supported
Mode of Operation	VoIP Access acts as Gateway–direct call signalling (no RAS) Customer Gatekeeper controlled (Inter zone: GK to GK) [Colt controlled Gatekeeper is not supported as standard]
Fax	Fax Over IP is supported using the ITU T.38 standard but is subject to the limitations specified elsewhere in this document.
Codec Parameters	Both G.711 and G.729/G729a use 20ms sample rate.
DTMF	H.323: H.245 (signalling + alphanumeric)
Other Bearer capabilities	Modem calls can be terminated but are subject to the limitations specified elsewhere in this document. 64Kbps Unrestricted Data is not supported

3.2 Interface Specification

3.2.1 Signalling Specification

Protocol Standard	Comments
SIP RFCs: 3261 [2543–now superceded]	These are the essential standards definitions of SIP. The default mode of operation is SIP over UDP. On an individual case basis SIP over TCP is supported. Please check the configuration guide for the PBX to be connected to VoIP Access for actual supported mode.
H.323 Version 2 & Version 4 (H.225 & H.245)	ITU standards implementations to maximise interoperability with H.323 enable IP PBXs and end-points.
H.323 Annex M.1	Implemented to provide interworking between TDM and VoIP. Not typically used in basic service.

3.2.2 Media Bearer Specification

The media stream conforms to RFC 1889 Real Time Protocol for the transport of voice and in-band bearer path data.

3.3 SIP Mode of Operation–Direct Signalling Trunk

SIP trunks from IP PBXs are typically configured as a direct connection between the Colt VoIP Access platform and your IP PBX. In that scenario no registration is supported but direct SIP Signalling messages are exchanged between the IP PBX and the Colt Session Border Controller SBC.

3.4 H.323 Mode of Operation

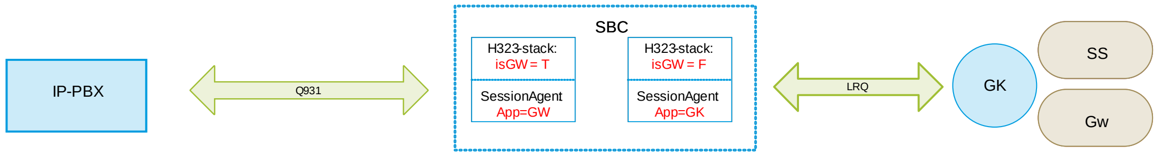
There 3 possible modes of operation for the interface between the customer IP PBX and the Colt service:

- Gateway-Direct call signalling to VoIP Access platform (No RAS)
- Customer gatekeeper to Colt Gatekeeper (Inter H.323 zone using LRQ)
- Colt controlled gatekeeper (PBX registers to Colt Gatekeeper using RRQ)–Bespoke orders only.

Option 1 is preferred method of operation. Option 2 can be supported but Option 3 is **not** a supported option and is described here for completeness.

3.4.1 Direct Call Signalling No RAS

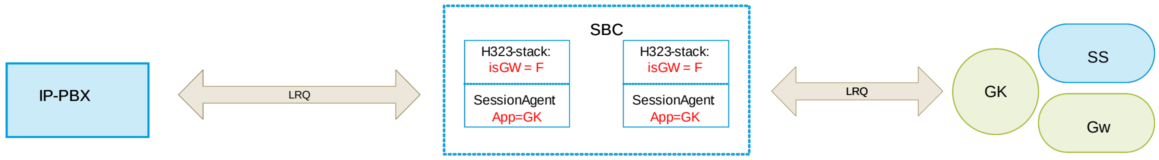
Some IP PBXs can be configured in a way to not use RAS messages with a gatekeeper but perform direct Q931 signalling. In that scenario no registration to a gatekeeper is done and no LRQ messages are sent to a gatekeeper but direct Q931 messages are exchanged between the IP PBX and the SBC.



3.4.2 Customer gatekeeper to Colt Gatekeeper (Inter H.323 zone using LRQ)

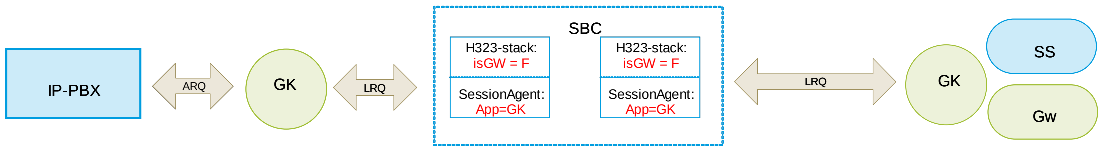
(a) Inter H323 Zone (GK to GK) using LRQ (IP PBX acts as GK)

In this scenario the IP PBX is configured in order to NOT register to Colt's gatekeeper. The IP PBX directly performs LRQ queries to initiate a call. The IP PBX also replies to LRQ messages from Colt gatekeeper for incoming PSTN calls.



(b) Inter H323 Zone (GK to GK) using LRQ (Customer owned separate GK)

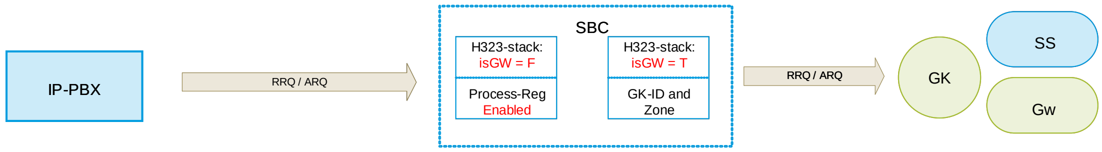
This scenario is used when the customer use their own gatekeeper. From the SCB configuration perspective, this scenario is identical as above however for troubleshooting it will be different (signalling).



3.4.3 IP PBX Registration on Colt Gatekeeper using RRQ

This is not a standard option and is provided for reference only.

Some IP PBX's don't support direct Gatekeeper mode based on LRQ messages. These IP PBXs would require registering to Colt VoIP Access's gatekeeper.



3.5 WAN connectivity and determining bandwidth requirements

3.5.1 WAN Access options

A suitable WAN or Internet Access service is required in order to connect to Colt VoIP Access-the following access types are available:

- "Dedicated access" for VoIP Access
- Access via a Colt IP Virtual Private Network (IP VPN)
- Access via public Internet or Colt IP Access (Colt's Internet access product). Note this is for SIP trunks only.

For option 1 (Dedicated connections) Colt provides the VoIP Access service up to and including the Customer Edge router. The link is used for VoIP traffic only; and must not be used for sending other traffic. For option 2 (via a Colt IP VPN), Colt provides connectivity between your existing IP VPN and the VoIP Access service (no **additional** physical access line or Customer edge router is provided). Option 3 provides the ability to connect a SIP-based IP PBX to the VoIP Access service via the Internet (Internet access service provided by Colt or a 3rd party).

You are responsible for the specification, provision and maintenance of your own Local Area Network (LAN) infrastructure, including ethernet cabling between the Colt CPE (if present), and your IP PBX. This infrastructure must be of a suitable standard for passing high quality real-time VoIP calls.

The link between your premises and Colt needs to be dimensioned and configured correctly in order to maintain a good quality of service. The maximum required number of simultaneous calls during the busiest time of the day (the "busy hour") between your sites and the PSTN will determine the bandwidth required. Colt can provide assistance in calculating an appropriate bandwidth for use with the service; please speak to your Colt representative.

3.5.1.1 Dedicated Access

If you do not have an existing access service in place and require a connection for voice only, then 'generic' IP connectivity can be provided as part of the VoIP Access service. This connection is a dedicated MPLS based connection capable of carrying only VoIP traffic. All the connection bandwidth (minus 10% for signalling and management) may be used to carry Colt VoIP Access traffic.

The following bandwidth options are available:

- Colt 'Fibre to the premises' connection (referred to as On-net)
- 3rd Party leased line access (referred to as OLO tail)
- Colt DSL (where available Colt deliver over Unbundled Local Loop)
- 3rd party DSL (Wholesale DSL) – where Colt use DSL access from a wholesale for access. Only uncontended options (often labelled as 1:1 contention ratio) are used to ensure performance quality of the service.

3.5.1.2 Colt IP Corporate Plus (IP VPN)

This is a private IP network provided by Colt which is primarily delivered over MPLS infrastructure. VoIP Access is delivered over the MPLS variant as standard (Colt IP Corporate Plus). All bandwidth options of 1Mbps and above for IP VPN corporate are available. On DSL based access you should only use un-contended options (often labelled as 1:1 contention ratio) to ensure performance quality of the service.

If you are planning on converging voice and data on a Colt IP Corporate Plus network, a maximum of 30% of the total bandwidth can be allocated to voice.

3.5.1.3 Colt IP Access and Public Internet (SIP trunks only)

Colt VoIP Access is available over public IP networks which do not restrict any of the port or application types used for VoIP. In general using public IP based access should be approached with caution as Quality of Service is not available which may lead to degraded VoIP quality.

For "Non-Colt" (public) internet access then the following guidelines must be followed and noted:

The Internet connection must **only** be used for VoIP Access traffic, no Internet traffic may be sent. This is to minimise the impact of contention between voice and data on the access link.

Due to the nature of the Internet, it is impossible to configure "Quality of service" mechanisms, which may mean that there will be performance degradation between the 3rd party ISP and Colt's VoIP Access platform. Using VoIP Access via an Internet connection is therefore provided on a 'best effort' basis, and is subject to a reduced SLA. Colt is unable to provide any guarantees of speech quality where an internet connection is used to connect the IP PBX.

If voice is critical to a customer's business operation, Colt recommends using an access method which has Quality of Service enabled, and offers un-contended bandwidth. Colt's IP VPN (IP Corporate Plus) and VoIP Access Dedicated Access both offer such a connection.

Note that only SIP based trunks are supported over public IP (Internet) networks; H.323 connections are not supported.

The following recommendations are provided to minimise risk of speech quality impairment on Internet based connections:

For connections via Colt IP Access (Colt Internet):

- Access bandwidth should not exceed 80% peak utilisation
- VoIP traffic bandwidth should not exceed 5% of overall bandwidth (eg100Kbps on a 2Mbps link-approx 4 calls)
- Access bandwidth should be non-contended and 2Mbps (bi-directional) or greater

For connections via public (3rd party provided) Internet connections;

- Connections should be used for VoIP Access (voice) traffic only. No internet data (http, ftp etc) should be passed across the link. This is to eliminate contention between voice and data traffic on the access link.
- Access bandwidths should be non-contended and 2Mbps (bi-directional) or greater

3.5.2 Bandwidth Requirements

It is very important to ensure that the bandwidth allocated between your IP Telephony end-points and the Colt VoIP Access platform is sufficient for the peak number of calls required. If this is not carefully dimensioned then voice quality will degrade. Typically the access connection to a site is the limitation of bandwidth. The following table can be used as an approximation to the amount of bandwidth used per call (**Note:** variations in layer 1 and layer 2 overhead can make figures vary):

Codec	Bit rate Payload (Kbps)	Sampling Rate (ms)	Leased Line Bandwidth per Call (Kbps)	Approx. ATM (DSL) Bandwidth per Call (Kbps)
G.711	64	20	84	106
G.729a	8	30	21	28
G.729a	8	20	28	43

The following formula provides the peak number of calls that can be carried for a given access bandwidth (10% of bandwidth must be reserved for signalling and management):

Peak Number of Calls = [Available access bandwidth]/[bandwidth per call x 1.1]

3.6 IP Design

3.6.1 Private IP Addressing (Dedicated connection and IP VPN)

The service is only supported with direct connection across private networks. This means that all addressing is provided by the customer in their private network range.

3.6.1.1 Reserved IP Addresses

In order for us to manage the customer located equipment as well as provide WAN connectivity across the network, we will use the following addresses for management. It is not possible for you to use any of these addresses on your own network, you will need to re-address if they are currently in use.

Excluded LAN Addresses
10.82.0.0/22

3.6.2 IP Addressing Options for Internet based customers

3.6.2.1 Overview

There are three IP Addressing Options for Internet based customers:

- Option 1: NAT with Port Forwarding
- Option 2: No NAT with single public IP address
- Option 3: No NAT with dual public IP addresses

These options are described in more detail below:

3.6.2.2 NAT with Port Forwarding

The Colt VoIP Access platform has public IP addresses for connection over the Internet. However, it is more normal to have the IP PBX and endpoints within a privately addressed network, which is connected to the Internet via dynamic Network Address Translation (NAT) or Port Address Translation (PAT). Such cases require a special setup to ensure the Colt VoIP Access platform can communicate with the IP PBX behind the NAT device.

Internet access terminated on a device which has ‘N:1 NAT’, also called PAT (or overload NAT), is configured to provide connectivity for internal devices to the public Internet via a shared public IP address. This NAT method implies that initial request is made from internal side to the internet (eg a PC connects to a web-server). For a VoIP Access to IP PBX calls it is necessary to configure a static NAT entry on the CPE router which points to the IPPBX. You should either request this from your service provider if the CPE performing NAT is managed or implement the following configuration on your own router performing the NAT. The CPE router is configured to forward traffic received on a given port on the public WAN interface to the IP Address od the IPPBX on the internal side. For VoIP Access SIP trunk this means that ideally UDP port 5060 (standard SIP port) should be used on the WAN side to identify packets which must be translated to the IP address of IP PBX. (Other ports could be used as well but since Colt need to know the port number for VoIP Access platform configuration it must be provided on the order form (in addition to the IP PBX address).

Example:

```
[Colt] -> [CPE Public Address] -> [IP-PBX Private Address]

 80.80.80.1:5060 -> 10.10.10.10:5060
```

Also if a NAT on CPE cannot port forward based on UDP port 5060 then a different can be used, for example, using port 6000 on the WAN side would give the following mapping:

```
[Colt] -> [CPE Public Address] -> [IP-PBX Private Address]
                               80.80.80.1:6000 -> 10.10.10.10:5060

(port 6000 on CPE forwarded to SIP port 5060 on IPPBX)
```

The NAT function on the CPE must not be 'SIP aware', it must not change any SIP signalling data.

3.6.2.3 No NAT with single public IP address

In this case the IP PBX has a public address and all the media (RTP) is channelled through the IP PBX (signalling & media use same Public IP address). This option should be selected on the order form and public address recorded for the IP PBX.

3.6.2.4 No NAT with dual public IP addresses

In this case the IP PBX has a public address. The media (RTP) is channelled through a separate entity (media concentrator) which presents all RTP from a single public IP address different to that of the IP PBX.

3.6.3 Quality of Service

It is important to ensure voice quality that overall IP quality of service parameters are not exceed between End-points to VoIP Access Platform. The following table forms a guide to follow:

Parameter	Maximum value
Latency (Round trip)	150ms
Jitter	20ms
Packet loss	1 in 10E3 (or better)

The Colt network should typically not add more than the following to the overall end to end performance budget for Dedicated Access and IP VPN (figures do not apply to DSL based access):

Parameter	Value
Latency (Round trip)	30ms
Jitter	10ms
Packet loss	1 in 10E6

3.7 Bearer capabilities

3.7.1 Voice CODEC

You choose the CODEC which is used between the PBX and the Colt VoIP Access service (the choice of CODEC determines the extent to which speech is compressed within a VoIP network). The CODEC recommended by Colt for use with VoIP Access is G.729a with 30 millisecond sampling rate.

Supported Codecs are:

Codec	Recommended sampling period
G.729/G.729a/G.711	20ms (fixed) for both SIP and H.323 trunks

3.7.2 Fax

The ITU standard of T.38 for the carriage of fax over IP is the only recommended method for use with Colt VoIP Access. Even then testing has shown that faxing can be unpredictable. It is important to maximise access performance to increase success rate, but even then it has been found that different types of fax machines give different results.

Although the latest technology is used by Colt to convey fax, reliable transmission cannot be guaranteed. If fax is a critical part of your business we recommend the use of a traditional PSTN analogue line

3.7.3 Modem

Modems are rarely used on IP Telephony systems and are not recommended for use with the service. The only method currently available is to use G.711 codec and carry modem within the voice codec, however, this will only have a limited connect speed which may be unsatisfactory for the application.

If modem transmission is a critical part of your business we recommend the use of a traditional PSTN analogue line.

3.7.4 Digital Data Transmission

No standard for digital data transmission is supported including 64Kbps Unrestricted Data (URD).

If you require such capability then Colt recommends using a separate ISDN line.

3.7.5 DTMF

The following methods are supported to carry DTMF tones:

DTMF Method	Description
H.245 Alphanumeric	DTMF transferred out-of-band in H.245 signalling UserInputIndication message. In Alphanumeric mode only a single event signal is sent. This results in the regeneration of a fixed-duration DTMF signal (usually 200 milliseconds) by the far-end VoIP gateway. In this mode, the length of the regenerated digit is unrelated to how long you press the keypad button on the phone.
H.245 Signal	As above but includes information about timing and duration of DTMF (using SignalUpdate message). In signal mode, two events are sent: one to indicate the start of the digit and one to indicate the end of the digit.
RFC2833	For SIP DTMF transferred as Named Telephony Events in RTP payload

4 Supplementary Services

4.1 Direct Dial In (DDI)

Incoming calls to any of the allocated numbers against the PBX trunk are typically delivered in the following format. The default format for Called and Calling Party Number are specified in Annex B.

4.2 Direct Dial Out (DDO)

For SIP trunks numbers must be sent 'Enbloc'

For H.323 trunks 'Enbloc' and Overlap working is currently supported.

For SIP trunks numbers must be sent 'Enbloc'

For H.323 trunks 'Enbloc' and Overlap working is currently supported.

The default format for Called and Calling Party Number are specified in Annex B.

If multiple sites (within the same country) are supported over a single VoIP trunk, the customer must ensure that they send a full "national format" Calling Party Number on all calls. This because in this scenario the source IP address alone is not enough to route the call to the correct "city specific" dial plan. Also, a VoIP Access trunk can only support multiple number ranges for a given country. Support for multiple countries on a single IP PBX requires a trunk per country although multiple trunks to one PBX can only be supported over dedicated access and IP VPN, and not over Internet connections.

Calls are billed on a per-second basis, and may be itemized on the bill if requested on the Order Form. **Note:** Offline Billing. In Germany, some telephone numbers, including 010, 012, 018, 0191, 0192, 0193, 0194, 0900, 118, will be billed by a 3rd party, and not by Colt (note that exceptions exist within these ranges). You should expect to receive a separate invoice for calls made to these numbers.

4.2.1 Emergency calls

In some countries emergency calls are routed to local emergency centres defined by the geographical significance of the Calling Party Number of the call (which must pass the screening against the geographical ranges allocated to the trunk).

Important Note: It is the customers responsibility to ensure that calls sent to the VoIP Access service have the correct geographically-significant A number to represent the location from where they originate. This will ensure correct routing in countries where emergency calls are handled locally.

4.2.1.1 Calling Line Identifier Presentation (CLIP)

Colt VoIP Access service screens the CgPN (Calling Party Number) to ensure it conforms to the PSTN numbers allocated to the service connection. Should the number sent not exist within the allocated number range or a number parameter is not present then Colt shall insert a default number (the "main number" as specified on the Order Form) relating to the allocated range. Note that where local regulations allow, you may request CLIP (no screening); which will allow you to populate the CLI field with a number other than that allocated to the IP trunk by Colt. You must be permitted to use the number which you populate in the CLI field.

4.2.1.2 Calling Line Identifier Restriction (CLIR)

A CgPN may be marked as restricted to prevent presentation to destination parties. This can either be done by the following methods:

- customer IP PBX sending CgPN marked as 'presentation restricted'.
- the customer IP PBX sending (via user dialling) a country specific prefix (where available) : '141' in the UK, '069' in Spain.

4.3 Other Supplementary Services

In general VoIP Access should be transparent to supplementary services implemented on the customer IP PBX.

Supplementary services requiring interaction with the network are not supported at this time. Call forward and diversion may populate the Redirection number parameter (in SIP this is called the diversion header) to show the number of the call diverting, however for this to pass network screening the full national significant number should be sent.

5 Resilience

5.1 Inbound Call Rerouting

Colt offers an additional chargeable feature with VoIP Access known as ‘Inbound Call Re-routing’. When loss of connectivity occurs between the IP PBX and the Colt SBC, all calls are redirected to a single E.164 number specified by the customer. This E.164 number must be specified in advance, via the product Order Form. Note that all inbound calls to the customer’s existing DDI range will be redirected to a single telephone number, and no provision is made for DDI functionality with this feature.

Note: the redirection number is not populated with the original Called party number as it is used for an internal purpose.

6 Service Delivery

6.1 Fast Track

Fast Track installation is available for all On-Net sites in all Colt geographic locations. If Fast Track is possible, Colt will provide a Fast Track delivery date based on technical feasibility, which customers can accept or reject. If accepted, the Fast Track option incurs an extra charge in addition to the standard installation fee.

6.1.1 Dedicated Access Variant

Fast Track allows customers to expedite delivery of their order and receive their service on a date specified by the customer (generally prior to the standard lead time), or, if this date is not possible, on the earliest possible date.

Consult a Colt Account Executive for more information.

6.2 Modifying an existing service

Modifying an existing service consists of the subsequent enabling or disabling of service features, functions and interfaces as well as service changes following initial installation, which are chargeable items.

7 Service management

Colt VoIP Access is a managed service which comprehensively covers service installation and configuration, fault handling and service restoration, performance reporting and call detail record access.

Where service is delivered over a dedicated connection Colt is responsible for the delivery and maintenance of the VoIP Access service from the customer premise termination point to the Public Switched Telephone Network (PSTN). Where the Colt VoIP Access service is delivered via a converged network, such as Colt IP VPN or Internet, the service demarcation is the interface between the data network and the Colt VoIP Access platform. The service management of the converged network should be addressed via the methods defined for the converged network itself. For example performance reporting on Colt IP VPN is provided by it's own online performance portal and faults pertaining to the IP VPN should be raised against the IP VPN contract number.

7.1 Fault Management

The Colt VoIP Access service includes a fault reporting number and access to a Performance Verification Tool (PVT) which may help self-diagnosis of quality issues (see Annex A for more details).

7.2 Performance Reporting

For dedicated access there is an online performance portal providing throughput, delay and packet loss. This can also be accessed from the central portal.

7.3 Service Level Agreement

The Service Level Agreement (SLA) describes the target for service delivery, restoration and quality for Colt VoIP Access.

It is important to note that the Service Level Agreement does not apply to customers who are connecting to the Colt VoIP Access service over the public Internet. This is because the quality and reliability of the service cannot be guaranteed.

8 Charges and Billing

Colt VoIP Access is charged on the following basis:

- Installation/configuration fee (one-off, dependent on access type)
- Service rental fee, where applicable.
- Voice tariff–usage based minutes charges (based upon in-country tariffs)
- Additional features and Moves Adds and Changes

Please note that orders for sites in different countries generate separate invoices for each site.

9 Glossary

CdPN - Called Party Number

CgPN – Calling Party Number

DTMF – Dual Tone Multi Frequency

IP PBX – IP Private Branch Exchange

LRQ – Location Request

PSTN – Public Switched Telephone Network

RRQ – Registration Request

RAS - Registration, Admission, and Status

Appendix A Performance Verfication Tool (PVT)

The Performance Verification Tool (PVT) provides a simple, web-based tool for the customer to verify VoIP Access network connection and call quality. It operates from a browser application on a PC. Probes are connected in various areas of the Colt IP network and the PVT application tests connection and call quality between one of these probes and a PC at the customer premises.

PVT over Dedicated and Via Colt IP VPN connections

For VoIP Access dedicated access and IP VPN Corporate based access a special configuration is required on the CPE router to support use of a private address for a PC to connect to the router at the site under test. Details of this configuration can be obtained from your Colt pre-sales representative.

PVT over Internet based access

For Internet based access the PVT can be accessed via the central portal (http://portal.Colt.net – see handover document for your logon details). It is necessary to ensure the PC used for running the tests will route internet traffic via the internet access used to carry the VoIP Access connection, and not via any alternative access which the customer may have in their network.

Running a PVT Test

The PVT is produced by Brix Networks hence references to Brix in this section.

There are 4 stages to run a test:

- Connect a PC to the LAN segment of the Colt CPE router interface.
- Using a Brower (Internet Explorer 6 or later) go to the PVT portal (labelled “Brix Care portal”) and select the appropriate test, the application will then downloads the required information to the PC.
- run the test
- display results

The following sections explain in more detail how to execute these steps for each access type.

- Connect a PC to the ethernet port of the CPE router or to the LAN segment associated with that port.
- Configure an IP address on the PC, set the default gateway of the PC to the IP Address of the ethernet port of the router.
- Open IE and browse to the following url http://brix0.lon.oss.colt.net:9090/index.html
- Follow the directions given in ‘Running the PVT application’

Running the PVT Application

The initial screen looks like:

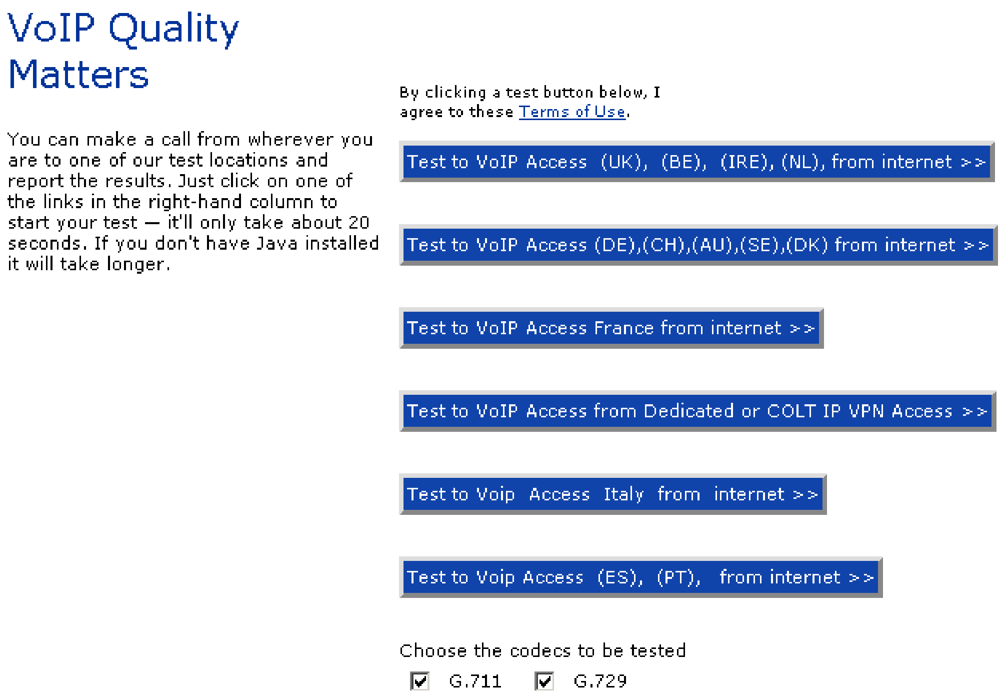


Figure 1: PVT test tabs

For Internet connections please select one of the following according to which country the site resides from which the test is being run:

- "Test to VoIP Access GB,BE,IRL,NL from Internet" – from UK, Belgium, Ireland and Netherlands
- "Test to VoIP Access DE,CH,AU,SE,DK from Internet" – from Germany, Switzerland, Austria, Sweden and Denmark
- "Test to VoIP Access France from Internet" – from France
- "Test to VoIP Access Italy from Internet" – from Italy
- "Test to VoIP Access Es, PT from Internet" – from Spain and Portugal

When a test completes, a basic results page is displayed for the BrixCare Agent user. The basic results page illustrates the test result using a bar chart or more detailed results can be displayed.

Appendix B Signalling Number Formats

The formats defined here address the default formats for national, and international numbers where appropriate, for the Called and Calling Party Numbers contained in signalling to and from the VoIP Access platform. Special numbers (eg. Emergency numbers) are exceptions and follow local formats.

DDI Formats

	SIP Trunks		H.323 Trunks			
Country	Called #	Calling #	Called #	ToN	Calling #	ToN
Austria	NSN	NSN CC+NSN	NSN	Nat	NSN CC+NSN	Nat INat
Belgium	NSN	Variable	NSN	Nat	Variable	Var
Denmark	NSN	NSN CC+NSN	NSN	Nat	NSN CC+NSN	Nat INat
France	0+NSN	0+NSN CC+NSN	0+NSN	Nat	0+NSN CC+NSN	Ukn INat
Germany	NSN	0+NSN CC+NSN	NSN	Nat	NSN CC+NSN	Nat INat
Ireland	NSN	NSN CC+NSN	NSN	Nat	NSN CC+NSN	Nat INat
Italy	NSN	0+NSN CC+NSN	NSN	Nat	0+NSN CC+NSN	Ukn INat
Netherlands	NSN	NSN CC+NSN	NSN	Ukn	NSN CC+NSN	Nat INat
Portugal	NSN	NSN CC+NSN	NSN	Nat	NSN CC+NSN	Nat INat
Spain	NSN	NSN CC+NSN	NSN	Nat	NSN CC+NSN	Nat INat
Sweden	NSN	NSN CC+NSN	NSN	Nat	NSN CC+NSN	Nat INat
Switzerland	NSN	0+NSN 00+CC+NSN	NSN	Nat	0+NSN 00+CC+NSN	Ukn Ukn
UK	0+NSN	NSN CC+NSN	0+NSN	Ukn	NSN CC+NSN	Nat INat

DDO Formats

	SIP Trunks		H.323 Trunks			
Country	Called #	Calling #	Called #	ToN	Calling #	ToN
Austria	0+NSN 00+CC+NSN	0+NSN	0+NSN 00+CC+NSN	Ukn	0+NSN	Ukn
Belgium	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN CC+NSN	Nat INat
Denmark	NSN 00+CC+NSN	NSN	NSN	Ukn	NSN CC+NSN	Nat INat
France	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	0+NSN	Ukn
Germany	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN CC+NSN	Nat INat
Ireland	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN CC+NSN	Nat INat
Italy	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN CC+NSN	Nat INat
Netherlands	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN CC+NSN	Nat INat
Portugal	NSN 00+CC+NSN	NSN	NSN 00+CC+NSN	Ukn	NSN	Nat
Spain	NSN 00+CC+NSN	NSN	NSN 00+CC+NSN	Ukn	NSN	Nat
Sweden	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN CC+NSN	Nat INat
Switzerland	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN	Nat
UK	0+NSN 00+CC+NSN	NSN	0+NSN 00+CC+NSN	Ukn	NSN	Nat

Key:

NSN – National Significant Number : significant digits without leading zero (eg For dialled number: 0170996465; the NSN is : 170996465 and 0+NSN is: 0170996465).

CC – Country Code : digits defining destination or origin country of the call (eg for dialled number 0033170996465; the CC+ NSN is: 33170996465 and 00+CC+NSN= 0033170996465)

ToN - Type of Number : defines the format of the number in H.323:

Nat – National

INat – International

Ukn - Unknown