Network And Internet Management Services

VoIP Products Guide

Version 3.2

Contents

1. What is VoIP?

Voice over Internet Protocol (VoIP) is a blanket term describing the technology which delivers voice communications over the Internet, rather than the traditional telephone infrastructure.

Why choose VoIP?
Hosted VoIP moves your investment in telephony from a CapEx to an OpEx model. Hosted VoIP is significantly cheaper than conventional telephony; you don't need on-site Private Branch Exchange (PBX) equipment, saving space, power and equipment costs; the telephone handsets are usually cheaper and call rates are often cheaper too.

Calls between VoIP subscribers (termed "on-net" calls) are free of charge and the number of on-net calls that you can make per account is not limited.

Hosted VoIP solutions are also much more flexible than a traditional hardware PBX because they are driven through on-line portals. This means changes you make have an immediate effect and are much simpler to implement.

Unlike proprietary systems, there is no lock-in to handset hardware; you can select the options that are right for your needs.

The beauty of a hosted solution is that it scales with your business. It is easy to integrate home workers with the rest of your team. You can also enjoy the peace of mind of being supported 24/7 by a UK-based technical support team.

2. What VoIP products do we offer?

We have three products in our VoIP portfolio.

VoIP EasyAccess
VoIP EasyAccess connects you to the outside PSTN network from your own Internet Protocol (IP) server. It makes a cost-effective ISDN alternative and is ideal if you have built your own VoIP platform, or if you have invested in an IP PBX which you wish to retain.

A SIP (Session Initiation Protocol) trunk is a service that allows your IP enabled PBX to make calls to the Worldwide Public Switched Telephone Network (PSTN). The easy way to think of it is a wire within a wire; in the same way that ISDN provides up to 30 virtual channels through a single physical cable, a SIP trunk provides a number of virtual channels over an IP connection.

VoIP EasyAccess supports the Inter-Asterisk eXchange (IAX) protocol in addition to SIP.

The VoIP EasyAccess service provides call termination complete with full UK geographical (01 and 02 telephone numbers) and non-geographical (03, 0845) numbering to provide your own VoIP service.

There are two chargeable components for VoIP EasyAccess; you order one voice channel per concurrent duplex call you wish to make or receive, and order the number of Direct Dial-In numbers (DDIs) you require to receive incoming calls. These DDIs can also be the presenting telephone numbers from the trunk when making outgoing calls.

The number of channels and DDIs do not need to be the same. For example, your business of 10 users may want every user to be directly contactable by external customers so you would order 10 DDIs. If your business is heavily telephone-based and you wanted each user to be able to have a call waiting in the queue at the same time as they are making an outbound call, you would need 20 channels; but if you make few outgoing calls as a team you might decide 5 channels is plenty between you.

VoIP Express

VoIP Express is a pre-pay, hosted service ideal for single-user residential or SoHo customers. Each VoIP Express account includes one UK geographical (01/02) or non-geographical (0845/03) number with voicemail. You can choose an area code and telephone number at the point of sign up.

VoIP Enrich

VoIP Enrich is a hosted PBX solution for multiple users that provides the key business functionality, once only affordable by larger enterprises, to every size of business. You can start with a single licence and simply add user licences as your business grows.

A VoIP Enrich account includes one extension with a UK geographical (01/02) or non-geographical (0845/03) number with voicemail and the ability to have two simultaneous calls in progress. One single level Auto Attendant, Hunt Group and Call Queue are included. You can add additional instances of any of these items for a fee.

3. How is the VoIP platform built?

We have invested heavily in creating a resilient platform that offers a public IP voice service from a multi-tenanted central voice switch which provides handset extensions to your premises. Our media and database servers are stored in our secure data centres in London and Telford which are supported by backup generators and there is sufficient capacity in the system to failover in the unlikely event that one data centre were to have an outage. We also use diverse outbound suppliers for call termination, again for resilience.

To safeguard partners from the effects of fraud we proactively monitor dialling patterns. We also place monthly and daily call limits on accounts to mitigate any potential harm. The daily limit is defaulted from the monthly limit, but can be amended by contacting us; users are emailed at 90% of the limit and at 100% further usage is blocked.

Appendix 1 to this document sets out the security measures we recommend you take to protect yourself from fraud.
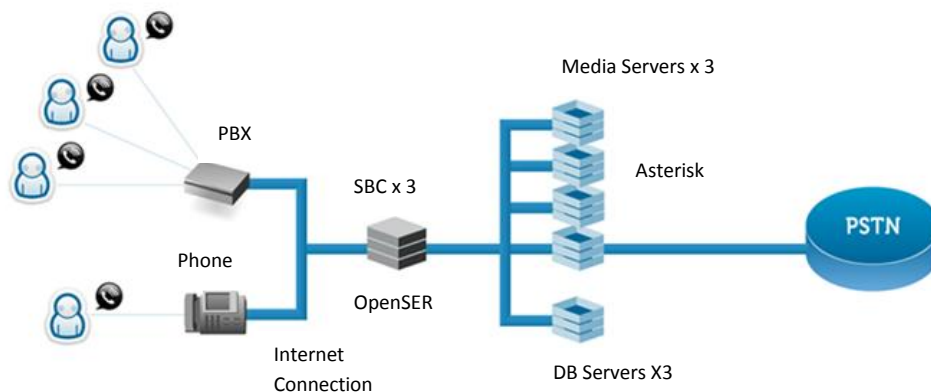
4. What devices can I use to make and receive calls?

One of the benefits of our VoIP services is their flexibility. Our services are Session Initiation Protocol ("SIP") based and can therefore be configured to any SIP device. There are three options:

1. Analogue Terminal Adaptor (ATA)

This is a device that you can purchase to enable VoIP calling from your existing standard (non-IP) telephone handset. An ATA is an adaptor that connects conventional telephone handsets to the customer's broadband router. When correctly configured, all outgoing and incoming calls made from your existing standard telephone handsets are routed over your VoIP account.

2. IP Telephone Handset

This is a device that replaces your existing standard (non-IP) telephone handset. The IP phone directly connects to your broadband router and can be configured to make/receive calls through their VoIP account. The IP phone offers a more user-friendly way to access certain call features such as voicemail, as it normally has function keys on the telephone keypad.

3. Software Phones

If you do not wish to invest in an IP device, a PC-based software phone can be downloaded free of charge from various web sites. However, this is not a recommended method as there are many factors which could affect the call quality that are outside of our control. For example, your PC's sound card quality, the PC microphone sound quality etc.

How do I configure a hardware device?

The SIP configuration details that are required for your IT department to configure a SIP device are set out in Appendix 2. Our support team will gladly assist you with these; but there is such a range of hardware available we will not advise on hardware specific issues.

5. Can I transfer my existing telephone numbers to the VoIP platform?

Yes, if you have working BT telephone numbers that you wish to retain, we can support the number porting from BT to our VoIP service. No other providers are supported. This process typically takes at least 15 working days and regrettably there is no guaranteed lead-time. Should you wish to port a BT number over to us, we can supply a Number Porting Request Form.

This is a chargeable service and the cost varies depending on which of these options is chosen:

- Move a single analogue number
- Move a BT Multi-line[1] number or range
- Move a range of less than 10 DDI numbers
- Move a range of more than 10 DDI numbers

A fee is still payable if a porting request fails.

One of our numbers will be used to provision a working account prior to the port and this number will remain on the account (and form the username for the SIP credentials). If the port is successful the ported number will be provisioned as an additional number on the account.

6. How are the services charged?

VoIP Express
VoIP Express is free to set-up. VoIP Express orders are activated immediately upon completion of the VoIP Express order process.

There is a monthly charge for the use of the service, paid in advance from your MasterCard or Visa credit/debit card or by Direct Debit.

VoIP Express is a pre-pay service to give you reassurance that you are in control of your call costs at all times. You may either manually top-up their account on an ad-hoc basis with immediate effect through the user portal by adding an amount of credit, which is depleted as calls are made, from your MasterCard or Visa credit or debit card, or you can set a credit level for automated top ups.

Where automated top ups are set, once the call credit equals or falls below the credit level set by you, a 'Top-up Amount' specified by you in pounds will be debited from your credit/debit card and credited to your VoIP account.

Call credit on VoIP Express accounts is valid for 90 days only, from the date it is added to the account. Any call credit unused after 90 days will expire and is not refundable.

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[1] BT Multi-line was a service allowing two analogue telephone lines to be connected to one number, also offered as BT Broadband Talk Multi-line with broadband service and more than one number. It is no longer available for new supply.

VoIP Enrich

VoIP Enrich has a small charge for setting-up the service. VoIP Enrich orders usually require 48 hours to be activated and are subject to you passing a credit check.

We will invoice you the monthly charge for the use of the service, billed in advance.

To protect you from fraud an Estimated Monthly Usage amount must be set at the time of order, with a £10 minimum (£50 max without credit check). Daily usage limits are also set by us.

VoIP EasyAccess

The VoIP EasyAccess monthly service rental is charged in advance and all chargeable calls are billed in arrears in a single bill. VoIP EasyAccess orders usually require 48 hours to be activated and are subject to you passing a credit check.

7. Is there a minimum contract?

There are no minimum contract terms on the VoIP Express, VoIP Enrich or VoIP EasyAccess services.

To cease a VoIP connection, please email us stating your cease requirement and the VoIP telephone number. Please be aware that a cancellation period of 30 days applies. Any unused call credit in a VoIP Express/Enrich account is <u>not</u> refundable.

8. How are calls charged?

Calls from any your VoIP account to any other VoIP users on the VoIP network are classified as 'on-net' calls. All on-net calls are free of charge (subject to SIP trunk channels being available). We do not limit the number of on-net calls per account that you can make.

Calls to any numbers outside of Entanet's VoIP network are classified as 'off-net' calls. For example, calls to PSTN numbers, calls to mobile numbers or calls to other VoIP providers' VoIP telephone numbers. Off-net calls are chargeable on a per second basis and are rounded up to the nearest penny unless otherwise agreed in your tariff (with the exception of any calls with connection charges or fixed fee calls which become chargeable the moment the call connects). Please refer to Entanet's VoIP Price Guide or contact your Entanet account manager for more information.

Charging for a call commences from when an answer signal is received indicating the media path is complete and ceases when a release signal is received indicating that the call has been disconnected.

9. VoIP Express and Enrich features compared

Feature	Express	Enrich
Web-based hosted management portal	✔	✔
View call history on portal	✔	✔
Included geographic/non-geo number(s)	✔	✔
Voicemail - delivered to you by email or telephone	✔	✔
Call forwarding option[†]	✔	✔
Withhold caller ID option	✔	✔
Screen incoming calls - black list	✔	✔
Do Not Disturb with override white list	✔	✔
Play received SMS	✔	✔
Check tariff on portal before calling	✔	✘
Send SMS from portal[†]	✔	✘
Reject anonymous calls option	✔	✘
Screen incoming calls - black list	✔	✘
Do Not Disturb with override white list	✔	✘
Call recording[†]	✘	✔
Follow me - try calling multiple numbers to contact me	✘	✔
SMS Voicemail notifications[†]	✘	✔
Send calls to this number to extension	✘	✔
Send calls to this number to hunt group	✘	✔
Pick up calls within your hunt group using *3 on your telephone	✘	✔
Send calls to this number to call queue with routing strategies	✘	✔
Send all calls to this number to voicemail	✘	✔
Auto-attendant routing	✘	✔
Time based routing behaviour for auto-attendant	✘	✔
Upload auto-attendant voice prompts as .WAV or .MP3 files	✘	✔
Bar outgoing calls by extension	✘	✔
Bar mobile numbers by extension	✘	✔
Bar international numbers by extension	✘	✔
Bar premium rate numbers by extension	✘	✔
Telephone directory on portal	✘	✔

Features marked with [†] are additional-cost options.

10. VoIP Express and Enrich features explained

Web-based hosted management portals

The User Portals https://www.voipuserportal.co.uk/ for VoIP Express customers and https://www.managevoip.co.uk/ for VoIP Enrich customers are easy to use yet comprehensive portals which enable you to manage their account online. All call features can be edited via the portal and all changes made to the system are updated immediately.

The following features are available to both VoIP Express and VoIP Enrich customers (features marked with a † are additional cost options):

View call history on portal

All call information is available within the "My Call History" section of the User Portal for VoIP Express and "Reports" in real time for information and monitoring purposes, with download to .CSV options.

Included geographic/non-geo number(s)

Each VoIP Express and VoIP Enrich account includes one UK geographical (01/02/03) or non-geographical (0845) number. You can choose an area code and telephone number at the point of sign up.

Voicemail - delivered to you by email or telephone

One voicemail box is included as standard. You can retrieve voicemail from your telephone, by email (voicemail will be delivered to a designated email address as a WAV file) or for VoIP Express customers from the User Portal where you can listen to and download the voicemail file. All incoming calls can be set to go to voicemail automatically during multiple time periods each day for VoIP Express customers and during the hours the business is closed or on special days such as bank holidays for VoIP Enrich customers.

Call forwarding option[†]

You can forward incoming calls to a different telephone number via the User Portal. Standard call charges apply when a call is forwarded to a chargeable outgoing telephone number.

Withhold caller ID option

You can choose to hide your VoIP telephone number for your outgoing calls ensuring the recipient of the call does not see your VoIP telephone number.

Play received SMS

If an SMS is sent to your VoIP number your phone will ring and the message will be played back to you.

VoIP Express customers have access to these additional features which are designed for personal users:

Check tariff on portal before calling

VoIP Express Customers can check the tariff for a destination before placing the call by entering the number into the tariff checker on the User Portal.

Send SMS from portal[†]

You can also send SMS messages to mobile numbers via the VoIP Express User Portal. Please note this is a chargeable service.

Reject anonymous calls option

You can choose to reject all calls received without any CLI presentation.

Screen incoming calls - black list

A black list is a call screening facility that allows you to block unwanted incoming calls. An incoming call from a number on your black list will be blocked or diverted to voicemail.

Do Not Disturb with override white list

A white list is an advanced Do-Not-Disturb feature. If the white list is activated and has no numbers entered it will block all incoming numbers. If a telephone number is added to the white list the system will allow calls from that specific number through to the customer.

VoIP Enrich customers have access to the following additional business features:

Call recording[†]

For complete security and peace of mind the administrator can set all incoming calls to a specified extension, hunt group or call queue to be recorded via the User Portal. These call recordings can be retrieved from the "Reports" section of the User Portal. An additional charge applies for call recording.

Follow me - try calling multiple numbers to contact me

Extension users can forward incoming calls to up to 5 different telephone numbers (which can be either internal or external numbers) via the User Portal.

SMS Voicemail notifications[†]

Via the User Portal customers can set at extension level the mobile number to be notified by SMS when a voicemail is recorded.

Send calls to this number to extension

Each user licence is allocated an extension number. Telephone numbers can be allocated to an extension to allow external callers to ring the extension directly. Your customers' internal users can call internally just by dialling the extension number.

Send calls to this number to hunt group/ pick up calls within hunt group

A hunt group allows you to pre-define a group of extensions/users and route incoming calls to this group for example, to the sales department. Members of a hunt group can pick up each other's calls by pressing *3 on their telephone and the administrator can set all incoming calls to either ring each phone within the group consecutively or all at the same time via the user portal.

Send calls to this number to call queue with routing strategies

A call queue enables you to retain all incoming calls by placing them in a queue until an appropriate extension is free to receive each call. Call queues provide several routing options beyond those available with a Hunt Group. In addition to 'Ring All' and 'Round Robin', you can choose 'Least Recent' (to always ring the user who has gone the longest time without receiving a call), 'Fewest Calls' (to always ring the user who receives the fewest incoming calls) and 'Random' (to ring randomly between users). A call queue also provides a 'music on hold' option while the caller is waiting in addition to the traditional ringing tone to present a more professional image to callers.

Send all calls to this number to voicemail

This option allows incoming calls to go directly to voicemail so the user is not disturbed.

Auto-attendant

This is an automated answering system that routes calls to the appropriate extension, hunt group, call queue or voicemail on the caller's responses to the voice prompts that your customers have defined. Auto Attendant replaces a human operator and hence is considered to be a key telephone system feature that reduces operational costs and increases efficiency.

Time based routing behaviour for auto-attendant

Time based routing is used to set the auto attendant messages which the caller hears during different times of the day or night. You can set messages for times when the business is open, closed and for special times, for example bank holidays. The times are set via the User Portal.

Upload auto-attendant voice prompts as .WAV or .MP3 files

You can record your own prompts for use on your auto attendant menus and upload them to the system via the User Portal. The system supports WAV and MP3 formats.

Bar outgoing calls: any of; mobile numbers, international numbers, premium rate numbers

Your administrator, in addition to setting permissions for configuring specific call features at extension level (voicemail, for example), can bar all or certain types of outgoing calls from each extension.

Telephone directory on portal

You can maintain a centralised phone book of internal and external contacts. You can add new contacts, edit or delete existing contacts and search for contacts within the phone book.

11. Can customers make calls to the 999 Emergency Services number?

We support 999 Emergency Service access from its VoIP service. When the access is granted, the call is made at no charge.

Please note that the VoIP service is dependent on the availability of power and internet connectivity to you, as well as on the availability of the on-premise VoIP equipment being used. Please note that you are not able to make/receive calls (including to 999 emergency services) under (but not limited to) any of the circumstances below:

1. When there is no power – unlike a traditional telephony service (such as PSTN) that does not rely on power to supply the service, a VoIP service is dependent on devices (such as PCs, ATAs or IP phones) which require a power supply or powered switch. A VoIP service will stop working (and therefore emergency services cannot be accessed) if the power fails.
2. When there is no internet access – A VoIP service fails when the internet service fails.
3. When the handset or other equipment that enables your connection to the VoIP service fails.

These limitations are not unique to our VoIP service and it is essential that you advise customers of their possible inability to make 999 emergency calls. It is strongly recommended that you have an alternative method of making emergency calls (such as PSTN or mobile) and such alternative calling methods are available on the premises where the VoIP is installed.

The 112 number for Emergency Service is not currently accessible from our VoIP service.

> Because VoIP is not tied to a geographic location it is essential that you maintain an up to date record of their location on the system, to allow the emergency services to reach them if they cannot be heard on a 999 call. This can be done quickly and easily at https://www.999-location.co.uk/.

12. Support Information

What do I do if I have a technical problem?
If you are experiencing technical difficulties please contact our Technical Support team on 0333 101 0600.

All our VoIP services are supported by 24/7 UK based technical support. Calls to our support team from a VoIP account are completely free of charge.

When is technical support available?

Our technical support team are available via phone or email 24/7/365 to help with any technical issues you may be experiencing.

What if I have a provisioning enquiry?

If you have any enquiries regarding the provisioning of the VoIP service you should contact our customer services team on 0333 101 0600.

What if I have a billing or accounts enquiry?

Please contact us.

Appendix 1 – Security Recommendations

VoIP Express and VoIP Enrich Customers

We recommend that you always use strong passwords for your accounts and avoid words commonly found in the dictionary or names of people or pets. A random list of alphanumeric characters with mixed case and added punctuation marks is most effective. VoIP Enrich has a feature to generate such passwords for you.

VoIP EasyAccess Customers

Anti-Virus

It is possible for a PC to become infected by a virus that would attempt to attack a Session Initiation Protocol (SIP) PBX on the local LAN. Therefore it is imperative that you keep all PCs updated with virus checking software and other security patches. This also applies to any PCs connected to the local LAN via a VPN connection.

It is good practice to separate the phone network from the rest of the LAN through the use of VLAN technology.

Firewall Settings

If you are directly connecting your PBX to a routable IP address you need to make sure you are firewalling that connection. Below we list the IP addresses and ports that we use which should be allowed through the firewall.

```
87.127.240.98
87.127.240.101                          87.127.234.34
87.127.240.104                          87.127.234.38
87.127.240.105                          87.127.234.39

87.127.215.242                          188.39.62.242
87.127.215.243                          188.39.62.243
87.127.215.244
87.127.215.245                          Ports 5060 TCP & UDP (For
87.127.215.246                          Signalling)
87.127.215.247
87.127.215.251                          Ports 10000 – 20000 UDP (For
87.127.215.252                          Media)
87.127.215.254
```

Asterisk Users

We recommend in addition to the above recommendations that a separate Linux IPtables firewall is run on the same machine as an Asterisk or Asterisk based installation.

Appendix 2 – Configuring Session Initiation Protocol client devices

Due to the numerous softphones and SIP devices available, we only give general advice regarding configuration. The following details are sufficient for most installations.

Credentials

Proxy Address: proxy.entacall.com (for VoIP Express and VoIP EasyAccess customers. VoIP Enrich customers view the Proxy Address on the extensions tab of their portal)

Username: As advised

Password: As advised

Codecs

A codec is a technology for compressing and decompressing data. We support the following codecs, which can be selected on the telephony hardware if it supports them:

- G.711 A or U law
- G722
- G723
- G726
- G729
- GSM
- H263
- H264

Network Address Translation (NAT) is often deployed in small business environments. It provides for an internal network infrastructure to be deployed using private IP addresses and connections out of the local network are translated to globally routed public IP addresses by a network router or firewall. NAT Traversal is supported by our VoIP solutions. However SIP expects an end to end IP address and if you do experience problems with one way audio or call drops after a number of seconds then the problem is almost certainly a firewall/NAT issue. Try specifying a Session Traversal Utilities for NAT server such as stun.counterpath.net. Also, some Analogue Telephone Adaptors and Internet Protocol phones will need to know the routable IP address of the NAT router. Check firewall logs to make sure both signalling (port 5060) and media (ports 10000:20000) are passing traffic both ways.

How much bandwidth do VoIP calls use?

Normally VoIP calls take up around 8k – 60k bandwidth each way depending on the Codec they use. For example, G.711 uses 85Kbps in each direction over Ethernet and 115kbps in each direction over ADSL; G729 uses 25-30KKbps in each direction.

We therefore recommend having no more than 5 VoIP users on a single broadband connection. We also recommend having a dedicated Internet connection such as Annex M broadband or EFM/leased line for the exclusive use of your VoIP service (i.e. not shared with data users).

Example Minimum Asterisk Configuration

sip.conf

`register => 44123456789:`mypass`@proxy.entacall.com/`myexten	Replace the username, password and extension examples with your own
`[entacall]`	
`type=friend`	Create channel for incoming and outgoing calls
`username=`44123456789	Replace with the Username provided
`secret=`mypass	Replace with the Password provided
`host=proxy.entacall.com`	
`nat=no`	Set to no or yes depending on whether NAT is in use
`authuser=`44123456789	Replace with the Username provided
`dtmfmode=rfc2833`	Generate a signal when you press the keys on the handset
`context=default`	The context of the extensions.conf file
`insecure=very`	Allow registered hosts to call without re-authenticating
`canreinvite=no`	Once a connection is set up, stop asking
`allow=ulaw`	Allow the ulaw

codec

extensions.conf

This file controls how incoming and outgoing calls are handled and routed.

```
exten => _9.,1,Dial(SIP/${EXTEN:1}@entacall|120|t)
```

Testing your account

Once you have successfully registered with us, you will see the name of the User Agent on the portal
https://www.voipuserportal.co.uk/

An echo service is available by calling 121. If you can hear your voice echoed back to you then both signalling and media are working correctly. This is also a useful check for latency issues. Additionally, call your VoIP number from the PSTN/Mobile network to make sure incoming calls are accepted.

Useful Numbers

- Voicemail   1571
- Current Credit   122
- Echo Test   121
- Telephone number reminder   123

Appendix 3 – Understanding your Call Detail Record ("CDR")

When do I receive a CDR?

We email CDR files to support your monthly statement. The subject line of the email is your VoIP account reference and the files are text files with the extension .txt in ASCII text comma-separated value (CSV) format. Each file has a unique name composed of PP and then an incremented number. The first row is a header row bearing the names explained below. Each row thereafter is for one outbound call. No field is ever blank.

What does my CDR show?

Column	Name	Description	Sample Value
1	UniqueRef	Internal hexadecimal reference	f1ffb4e865b1588348d76eaf138cfae2
2	StartTime	The time the call was answered by the dialled party in (UK) DD/MM/YYYY (24hr) hh:mm:ss format	02/08/2012 10:08:35
3	Account	VoIP account number	441930420304
4	CLI	The caller line identification of the calling party in Canonical without plus format. This is not necessarily one of the inbound DDI (direct dial-in) numbers assigned to that SIP account; a customer may nominate any telephone number which they own to appear as their outbound CLI. It may also be blank or contain only white-space.	443331050670
5	Destination	The number dialled by the end user, in the format in which they dialled it (see table below)	01604070605
6	Duration	Call duration in seconds	871

7	BillableSeconds	Billable seconds	849
8	Callcost	Cost of call in GBP (£) to three decimal places ,rounded up where the third decimal place is =>5	0.123 [i.e. 12p]
9	Network	Description of call destination	UK Geo

Possible formats for the values in column 5:

Description	Alternative name	Example
National without zero	Local without zero	2074020304
National	Local	02074200304
Canonical without plus	International without leading zeros	442074020304
Canonical with plus	E.164	+442074020304
International		00442074020304

What does my CDR not show?

Inbound calls are not shown on the CDR. Calls between VoIP enrich extensions are not shown on the CDR.

VoIP Express users may download their call history as a CSV file from https://www.voipuserportal.co.uk/.

This adopts a simpler format of the following columns: Call [Start] Time, Call End [Time], [Number] Called, Duration [Seconds], Cost [£], Destination