



SOHO KCCVoIP PBX GUIDE

Extract from the KCCVoIP Training for SOHO PBX configuration - last update July 2024

Before you plug it in - we need to do some basic design/planning as to how you are going to connect the appliance. If you have only one switch or one router, then you do not have much choice and will be connecting the appliance to the only available VLAN/network available. If you have multiple switches and VLAN configured, you may want to create a VLAN just for VoIP traffic to keep the voice traffic away from your video and Internet traffic. If you have limited bandwidth available, it is very wise to plan the network carefully and make sure you do not overload any part of your network otherwise you may have issues with voice quality. A voice conversation only requires 100kbps in each direction, but it does require low jitter and low packet loss to ensure good quality voice service. {Video calls require 350kbps}

If you have any questions, it is better to have them answered BEFORE you install and configure your voip appliance. Send any questions to support@kccvoip.com

Stages to consider

- Appliance location for power and network cabling
- Switch or router configuration to support the new appliance {what network address is available}
- Switch or router configuration options to support VoIP {SIP trunk to provider, QoS, extensions}
- If you are running a business –
 - remember if the power goes off then all of the phones are off
 - do you need a fallback system or UPS etc
 - do you need a backup system
- Dialplan design – how do you want to handle calls –
 - inbound route using DID {your external telephone numbers}
 - callerID {route calls to the right person using the callerID inbound}
 - IVR menu selection {give inbound callers the option to dial direct and/or leave message}
 - outbound route using DID {select the right outbound DID for each extension}
 - how to handle blacklisted calls
 - how to handle voicemail {and unified voicemail delivered to your email}
 - how to handle out of hours calls {set up working days/times etc}
 - how to setup 'follow-me' and forwarding to cell phones etc
- PSTN switch-off {do you have any telephones still using landlines}
 - You can port the landline numbers to a VoIP provider {ITSP} usually free of charge
 - Do you have any alarms, fax machines, franking machines that need porting
- Do you need remote access for telephone/smartphones etc. roaming outside the SOHO
- If you are integrating with Smart Home automation
 - Lights, alarm systems, heating and A/C, power and reporting system
 - Telephony caller ID display on your televisions, announce via Alexa, email status...
- If you are integrating with a CRM for a business
 - Brand and hooks required for database, agent screen population



above – important connections

All of the Asterisk/FreePBX SOHO appliances require only two connections – a power supply and a network. In the example above {Acer Revo M1-106} the ultra small form factor PCs require connection to their own 19v power supply. Other appliances either have dedicated power supplies or connect directly to the mains. Appliances with multiple interfaces and high availability server clusters are covered in separate documentation.

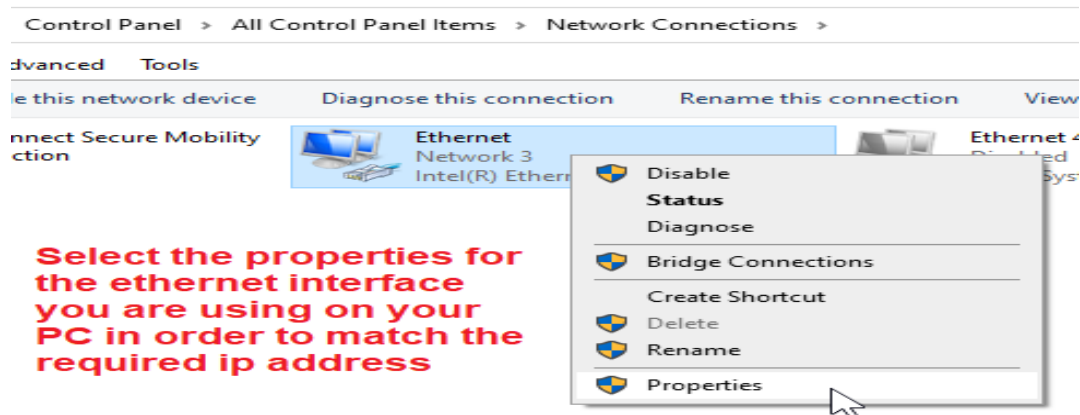
The appliances are supplied either as pre-configured to match your network OR configured with a fixed ip address. **If you have been supplied with an appliance with a pre-configured ip address to match your network, the following changes will not be necessary and you should be able to jump to SOHO basic configuration on page 7.** If not pre-configured for your network, in order to connect your browser to the appliance, you must temporarily change the network address on your PC before you can communicate with the appliance.

Usually, internal home and office networks {not public addressed networks} will have an address configuration from the designated private address ranges {RFC1918} 192.168.x.x or 10.x.x.x or 172.16.x.x etc., but it is also possible that you are using a public address. This configuration routine can be used for any address you want to place on your new appliance for VoIP telephony.

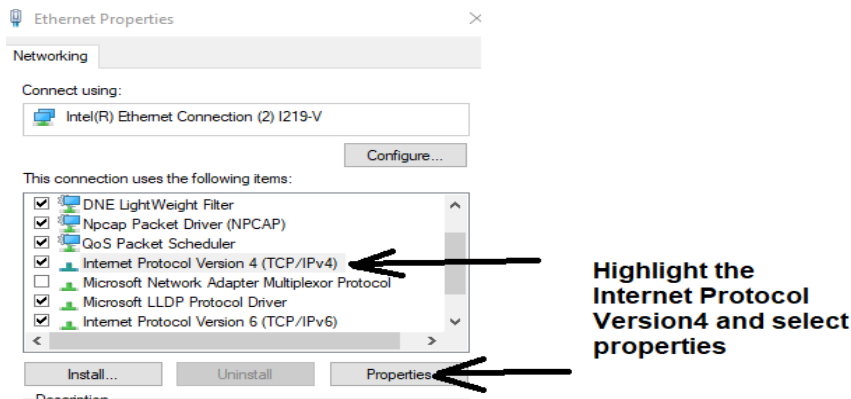
In order to communicate with the appliance, we need to know what address it is using. If the appliance had been set for DHCP an automatic unknown address would have been allocated by your network. In that case, we would have to interrogate your network switches and/or routers to find out what address had been allocated for the appliance. Instead, we allocate a known fixed ip address to the appliance so that you can easily connect to it and then change it to match your network requirements.

For example - Acer ast35 has default ip address 192.168.60.35/24
To connect to this appliance your PC must also have an address
on the 192.168.60.0/24 network – such as 192.168.60.1

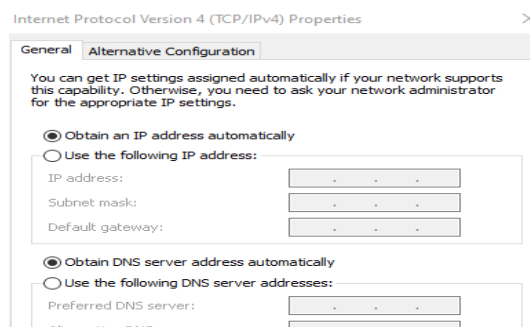
The configuration for a windows PC is shown below



Note – it is possible to make these temporary changes to a WiFi interface or switch, depending upon your local topology/configuration. **Alternatively, If you have been supplied with an appliance with a pre-configured ip address to match your network, these changes will not be necessary and you should be able to jump to SOHO basic configuration on page 7**



Usually you will find your PC is set to obtain an ip address from your router automatically using DHCP. This needs to be changed temporarily so that your PC will have a static address on the required network to communicate with the appliance. In the example below we see that the PC is set for DHCP ;





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In this example, we will set the PC ip address to the private address 192.168.60.1 to then allow us to connect to the appliance and configure the appliance with an address to match your network before returning your PC to the original configuration.

Internet Protocol Version 4 (TCP/IPv4) Properties

General

You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.

☐ Obtain an IP address automatically

☒ Use the following IP address:

IP address: 192 . 168 . 60 . 1

Subnet mask: 255 . 255 . 255 . 0

Default gateway: 192 . 168 . 60 . 35

☐ Obtain DNS server address automatically

☒ Use the following DNS server addresses:

Preferred DNS server: . . .

Alternative DNS server: . . .

So now we have a PC with network address 192.168.60.1 and the appliance with address 192.168.60.35

Notice – in this example we have set the gateway address to the address of the appliance. It is also acceptable to set the gateway address to a valid gateway address in your network if you are using the 192.168.60.0/24 network.

We can either connect directly to the appliance or we can connect the appliance to the same switch/router/VLAN as our PC to allow them to communicate.

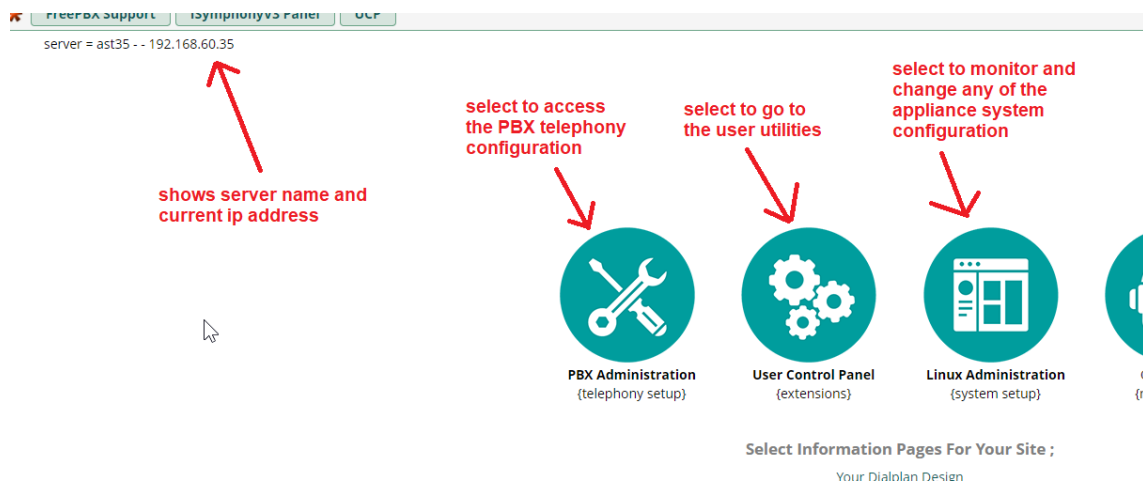
If you have been supplied with an appliance with a pre-configured ip address to match your network, the changes above will not be necessary and you should be able to jump to SOHO basic configuration on page 7

Send any questions to support@kccvoip.com

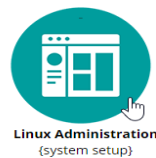


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Once your PC has an address configured on the same network as the appliance, you should be able to use a browser to connect and configure the appliance {https://192.168.60.35} - depending on the browser you are using you may see a security warning page when you connect {select advanced and proceed to site}

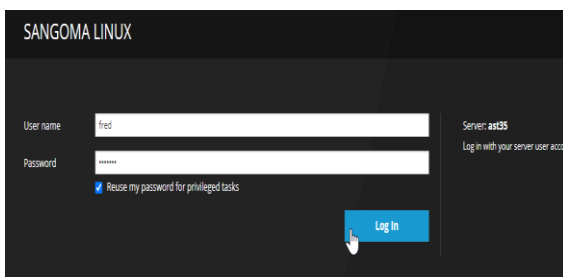


In this example we are going to change the ip address of the appliance from the default 192.168.60.35 to an address on our home network {192.168.0.35}.

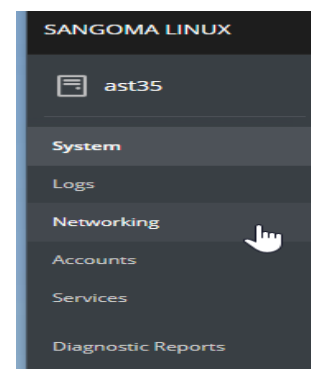


Select the Linux Administration from the menu

Then login with your default user name and password and check the box 'Reuse my password for privileged task' which allows you to make changes to the Linux system configuration.



Then select the networking option from the side menu





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For the installation of the telephony server, we need to change the appliance address to an address which will work on your networks.

Change the ip address of the appliance

Interfaces	
Name	IP Address
eth0	192.168.60.35/24

eth0 Realtek Semiconductor Co., Ltd. RTL8111/8168/8411 PCI Express Gigabit Ethernet Controller

Status 192.168.60.35/24, fe80:0:0:0:3919:6fd2:8ba0:31ac/64

Carrier 100 Mbps

General ☒ Connect automatically

IPv4 Address 192.168.60.35/24 via 192.168.60.254
DNS 192.168.66.254

IPv6 Automatic

MTU Automatic

IPv4 Settings

Addresses

Manual

192.168.0.35 24 192.168.0.254

DNS Automatic

Then click apply and the appliance will warn you that you are about to change the address and will need to change your PC address to reconnect ;

Connection will be lost

Changing the settings will break the connection to the server, and will make the administration UI unavailable.

Change the settings Keep connection

You should then be able to configure your PC back to the original DHCP settings and connect to your VoIP appliance using your own network. In this example the appliance is now on 192.168.0.35 and your PC will obtain a DHCP automatic address in the 192.168.0.0 network

Send any questions to support@kccvoip.com



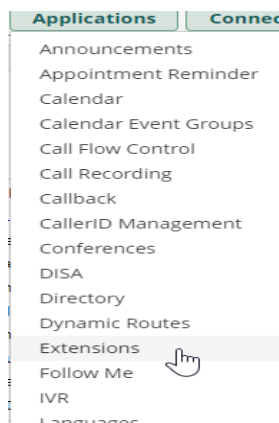
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SOHO PBX BASIC Telephony Configuration

Login to your PBX Administration GUI from your browser




You should then see the dashboard screen which shows the status of the VoIP appliance.



To add/edit/remove extensions, select the Extensions menu item from the Applications tab

From there you should see the list of extensions that have been created. You can edit, add and delete as required. To add extensions quickly without the need to know the SIP configuration details you can also use the **Quick Create Extension** tab from within 'Extensions'

NOTICE - there is a help prompt  next to each variable, so you can click to understand what each of the fields are used for.

Asterisk supports most VoIP protocols and can have very flexible mixed vendor connectivity. At default we have configured support for generic SIP which is used by a vast range of manufacturers {Siemens, Mitel, Avaya, Cisco, Digium, Sangoma, PolyCom, Grandstream, Snom...}. We also have included Cisco SCCP {skinny protocol – used by the large corporate call manager systems} which allows connection of Cisco telephones and devices which have a vast array of features. IAX, H323, MGCP and interfaces with proprietary manufacturer protocols are also available {NEC, Siemens, Panasonic, Nortel, Mitel, Avaya, PRI/BRI....}



Display name	=	name seen as callerID for this extension and name seen on any telephone display
Outbound CID	=	the caller ID seen on calls from this extension {usual format = user name <extension number>}
Emergency CID	=	the caller ID seen on emergency calls {if used, needs to be a public number that corresponds to an actual address that the emergency services will use}
Secret	=	the password used by any telephones using this extension

Voicemail has been enabled for these default extensions and you can delete/edit/add as required for your environment.

Send any questions to support@kccvoip.com

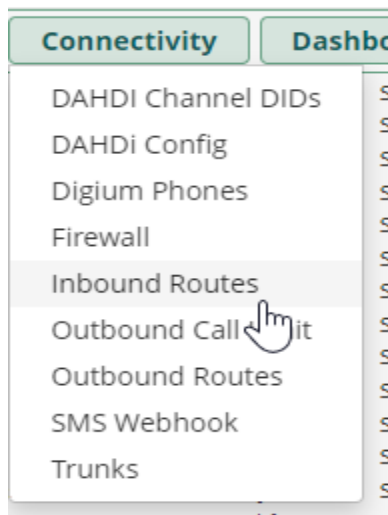


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NOTE on inbound call routing - there are many ways you can identify an inbound call and route it to the desired extension or to a routine in your VoIP PBX {inbound caller ID, DID the number the caller dialed, the time of day of the call, via ring group of extensions and/or external numbers to reach you, via IVR menu to let the caller select. You can also either blacklist calls you do not want or whitelist calls you want to bypass any menus and further security...}. Now that most telcos {except BT as yet} are using STIR/SHAKEN to identify bogus and unwanted calls, you can use the caller ID on an inbound call to decide where to send it within your system and know if the caller is trusted or tagged as possible spam etc. Important calls can be forwarded to your mobile number or voicemail if you are not available.

For example

Call Routing Examples



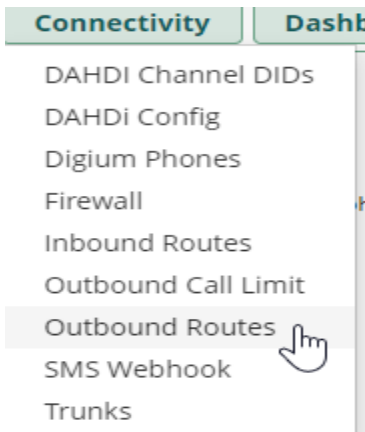
Inbound routing can be configured using the **Inbound Routes** menu item from the **Connectivity** tab

Use this to set the handling of inbound calls into your VoIP PBX from the outside world. You can set caller ID information to route calls with known numbers to route to the desired extension numbers or menus within your VoIP PBX

NOTE on outbound call routing - much the same as inbound routing, you can decide where you want to send outbound calls based upon the extension used or time of day or prefix dialed etc.. You can keep all home calls via a specific external telephone number, business calls to your business numbers and/or route to your smartphone etc. You can dedicate external numbers to particular functions such as a teenager in the house can have their own number etc. This way your customers see that you are calling from your business numbers and your friends and family see the calls as from your home numbers. You can run reports to see who, where when calls are made and received.

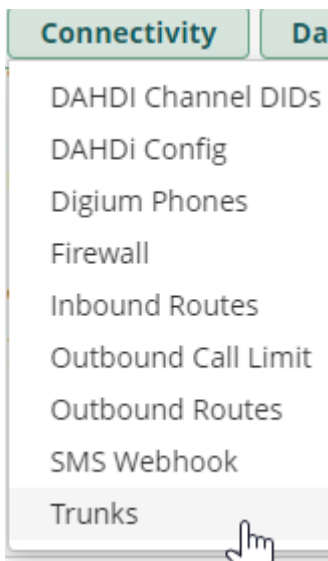


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Outbound routing can be configured using the **Outbound Routes** menu item from the **Connectivity** tab

Use this to set the handling of outbound calls from your VoIP PBX to the outside world. You can set extension and outside number matching information to route calls from extension numbers or menus within your VoIP PBX to outside world using the desired trunk and DID {your outside telephone number}



Trunks can be configured using the **Trunks** menu item from the **Connectivity** tab

Use this to configure all of your links to the outside world. Here you configure your SIP trunk to your VoIP provider where your DID numbers are terminated.

At default you will find there is an implementation trunk configured which allows test calls to/from KCCVoIP. This allows support from KCCVoIP and is also used as a training example as to how to configure a SIP trunk.

NOTE – you may have to add an entry to your firewall to allow SIP and RTP traffic through from the outside world to your VoIP appliance. Map UDP port 5060 and UDP/RTP port range 10000-20000 through to the appliance address. Outbound firewall should automatically allow outbound traffic from the appliance. If you have only the one VoIP server, it is safe to disable ALG, but ALG is required if more than one VoIP server is behind your firewall. If you have a support agreement or consultation arrangement, you may also need to allow the support ports through your firewall – details will have been supplied with your support agreement.



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Here is an extract of some of the basic applications within the PBX you may need for SOHO

System Recordings (menu item from the Admin tab)

Allows you to setup announcements and messages used within the system for providing announcements and menus to callers into the system. At default we have created the announcement 'Welcome' which is used for the IVR auto-attendant which consists of the English GB male voice saying 'if you know the extension you require, please dial it now'

You can also record your own announcements, adverts, music and messages or use any combinations of the thousands of pre-recorded announcements and messages in the system recordings directories. You can also upload your own announcements etc. for your own IVRs and adverts etc

IVR (menu from the Applications tab)

Here you can setup menus for your callers using the system recordings from above to prompt and announce to callers. The default example 'welcome' has been created which will show you how this works.

Ring Groups (menu from the Applications tab)

This allows you to configure groups of extensions and phone numbers to be called from a single number. The ring group can be configured to ring all numbers within the group at once or one at a time or in a priority order... etc. You can set announcements and music or adverts to play until the call is answered. You can configure call recording and final route of traffic when no numbers within the group are available etc.

Initial call routing configuration

Inbound calls → If from callerID 743 → route calls to home phones {ring group 201}

Inbound calls → if from callerID 735 → route calls to work phones {ring group 211}

Inbound calls → if from another caller ID → route calls to IVR {ask them to dial extension}

Outbound calls → if dialed number 7XX → route calls out through trunk to KCCVoIP

Send any questions to support@kccvoip.com



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Connecting telephones to your appliance

For most telephones you configure the SIP server address {192.168.0.35 in this example}, the extension number(s) required for this telephone/softphone/smartphone, the password assigned to this extension number {as configured on the appliance}. Every other configuration option provides refinements and additional features – these can be added once you have the extensions up and working.

Auto-provisioning – is where the telephone automatically obtains an ip address, full telephony configuration, video capabilities and firmware updates from the appliance. Full support for the Cisco telephone features are also installed ready for use on all of the KCCVoIP PBX systems. These features are free of charge and ready to be configured on your appliance, but beyond the scope of this document.

NOTE – if you are using smartphones, they will work using the private addresses when you are connected via WiFi, but may need additional configuration to work outside the home and via 4G/5G networks. VPN, DDNS or static address {from your Internet provider} would be required to allow connectivity outside the home via any other Internet connectivity and 4G/5G etc. anywhere else in the world.... Be aware, that if you are using 4G/5G type connectivity for your main home network - most 4G/5G providers do not allow inbound connectivity without additional charge – this will limit your VoIP connectivity from outside the home into your appliance. You can also forward calls to your cell phone numbers if the 4G/5G proves difficult.

In our example – the ip address of the appliance is now 192.168.0.35

For softphones/deskphones/smartphones we configure to point at the appliance address ;

Server	User	Network	Codecs	Certificate	Sound
Proxy/Registrar		STUN server			
<input type="text" value="192.168.0.35"/>		<input type="text"/>			
<input checked="" type="checkbox"/> Registration		<input checked="" type="checkbox"/> MWI			
registration interval		Domain/Realm			
<input type="text" value="600"/>		<input type="text"/>			

Set the SIP proxy and registration server to your appliance ip address

Notice the selection of 'Registration' tells the softphone to contact the server/appliance to let it know you want calls to be sent to this softphone for the extension configured. MWI {message waiting indication} when selected, allows voicemail and lets you know if you have unread messages. STUN is used to resolve public addresses when you configure for outside the home using some 4G/5G and other public addressing {internet cafes, external WiFi etc} when VPN is not an option.



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Server	User	Network	Codecs	Certificate	Sound
User name					
<input type="text" value="212"/>		<input checked="" type="checkbox"/>		displayed name	
				<input type="text" value="KCCVoIP-8-8"/>	
				voicemail number	
				<input type="text" value="*97"/>	
password					
<input type="password" value="....."/>		authentication name		phone number	
		<input type="text" value="212"/>		<input type="text" value="212"/>	
				<input type="checkbox"/> From Header	

Here we see we have the softphone configured to use extension number 212 with the password set to agree with the configured extension secret/password on the VoIP server/appliance

If there are options for encryption, UDP/TCP, multicast DNS, auto answer / intercom, noise and silence cancelling etc. you should initially setup for non-encrypted, UDP with no multicast and no silence suppression until you fully understand the requirements. Then you can refine the configuration and make sure that the VoIP server configuration always matches the extension/telephone configuration.

Configuration of smartphones, deskphones and DECT VoIP are much the same as above. You only need to set the appliance {VoIP PBX server} address or hostname, the extension number and the password to enable the extension and you should then be able to make and receive calls.

Setting the CODEC {the quality of voice in use}, the voicemail options {quick dial button to access voicemail and indication that voicemail is waiting}, quick dial buttons, contacts lists and phone logs etc. are all options to refine and improve the telephony service. At default most phones are configured to support G711 {similar to old PSTN quality} and G722 {HD voice quality} for voice and H264/H265 for video.

When you have an extension connected and registered, you should be able to dial 411 and hear the directory service or dial another extension, dial a ring group such as 201 or 211, dial the IVR or dial *97 to check voicemail etc. You may have been given access to a test trunk into kccvoip – there you can also test dial over the SIP trunk to one of the kccvoip servers and beyond. Dial 300 should route over a test trunk to a kccvoip menu.

Send any questions to support@kccvoip.com

More detailed documentation is available on the web site such as [basics on the PSTN switch off](#), the [general guide to telephony](#), [low cost solutions guide to PSTN switch off](#), extract from training for [H323 and SIP basics](#) and [ALG and SIP](#) extract from training