



SOHO KCCVoIP PBX GUIDE

Extract from the KCCVoIP Training for SOHO PBX configuration - last update Jan 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}
- 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
- 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.

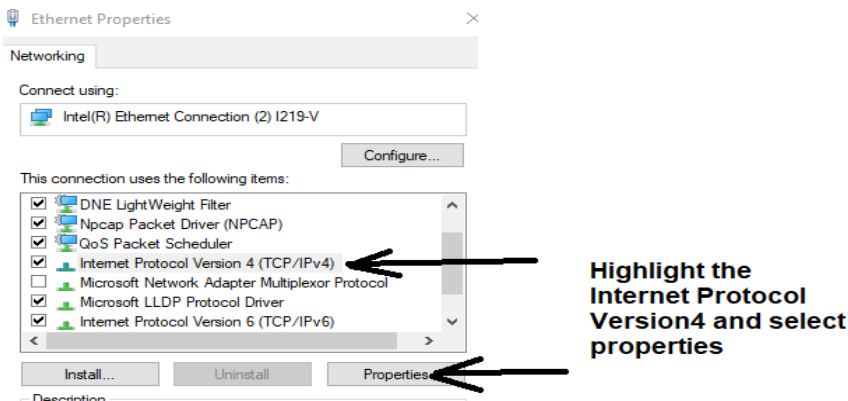
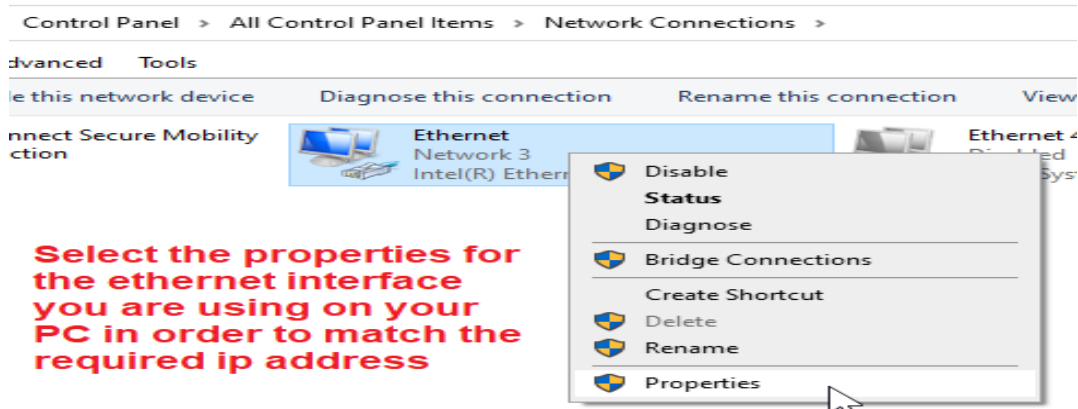
The appliances are supplied pre-configured and with a fixed ip address. To connect your browser to the appliance, you must match the network address on your PC before you can communicate with the appliance.

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.

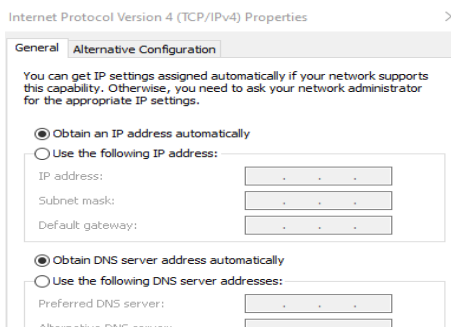
In order to communicate with the appliance, we need to know what address it is using. If the appliance had been set for DHCP an automatic address would have been allocated by your network. Unfortunately, we would have to interrogate your network switches and/or routers to find out what address had been 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



Usually you will find your PC is set to obtain an ip address from your router automatically using DHCP. This needs to be changed 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 ;





SOHO KCCVoIP PBX GUIDE

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 192.168.60.39

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.

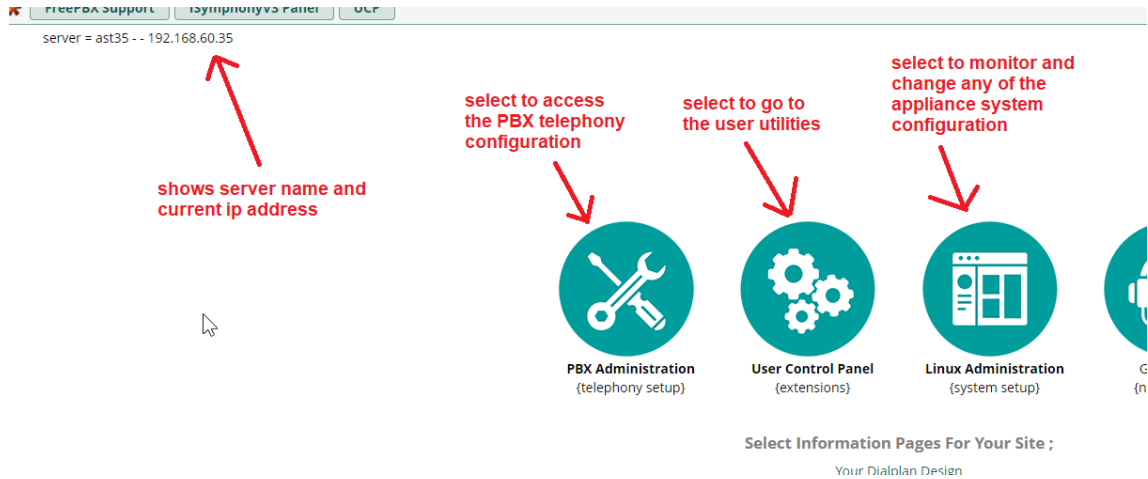
We can either connect directly to the appliance or we can connect the appliance to the same switch/router/VLAN as our PC to allow them to communicate.

Send any questions to support@kccvoip.com

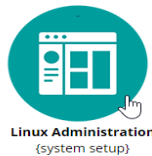


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Once your PC has an address configured on the same network as the appliance, you should be able to use a browser to connect and configure the appliance {https://192.168.60.35} - depending on the browser you are using you may see a security warning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

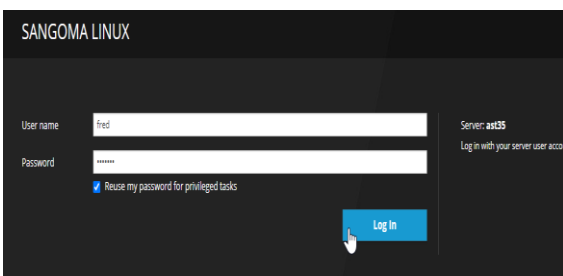


In this example we are going to change the ip address of the appliance from the default 192.168.60.35 to an address on our home network {192.168.0.35}.

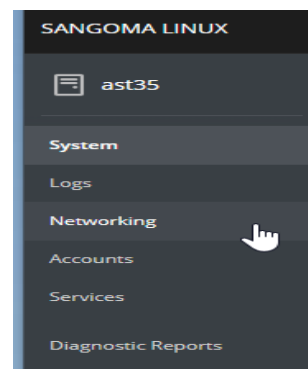


Select the Linux Administration from the menu

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, fe80:0:0:3919:6fd2:8ba0:31ac: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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The default configuration of the SOHO appliance is to have extensions 202 – 205 ready configured as home phones and 212 – 214 as office phones. The usernames can be change as required or if you prefer you can create a different range of extension numbers.

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

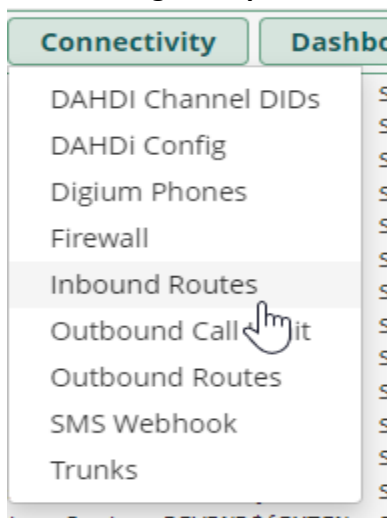
As you create new extensions and allocate user names etc. introductory emails will automatically be sent to the user emails with details on their new extension and how to use voicemail.

Send any questions to support@kccvoip.com

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 days of the call, a ring group of extensions and 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

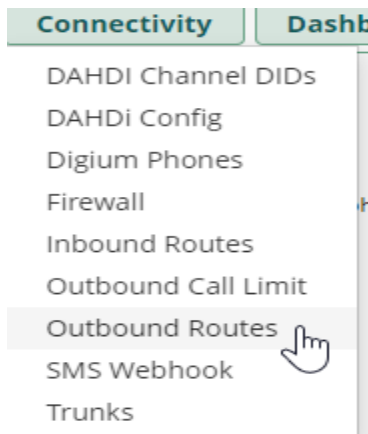
Use this to set the handling of inbound calls into your VoIP PBX from the outside world. You can set caller ID information to route calls with known numbers to route to the desired extension numbers or menus within your VoIP PBX

Send any questions to support@kccvoip.com



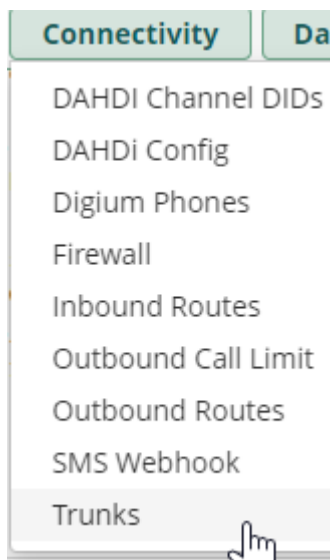
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NOTE on outbound call routing - much the same as inbound routing, you can decide where you want to send outbound calls based upon the extension used or time of day or prefix dialed etc.. You can keep all home calls via a specific external telephone number and business calls to your business numbers. 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.



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



SOHO KCCVoIP PBX GUIDE

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.

Here is an extract 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 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



SOHO KCCVoIP PBX GUIDE

Connecting telephones to your appliance

On most telephones you need to configure the SIP server address {192.168.0.35 in this example}, the extension number(s) required for this telephone/softphone, 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.

NOTE – if you are using a smartphone it will work using the private addresses when you are connected via WiFi, but will need additional configuration to work outside the home and via 4G/5G networks. DDNS or static addressing for your ISP connection is the usual method to ensure connectivity inside the home via WiFi and via 4G/5G any where else in the world.

For our example – the ip address of the appliance is 192.168.0.35

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

The screenshot shows a configuration window with tabs for Server, User, Network, Codecs, Certificate, and Sound. The 'Server' tab is active. It contains the following fields and options:

- Proxy/Registrar: 192.168.0.35
- STUN server: (empty)
- Registration
- MWI
- registration interval: 600
- Domain/Realm: (empty)

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

Check registration tells the softphone to contact the server to let it know you want calls to be sent to this softphone for the extension configured below. MWI {message waiting indication} checks for voicemail and lets you know if you have unread messages. STUN is used to resolve public addresses when you configure for outside 4G/5G and other public addressing {internet cafes, external WiFi etc}

The screenshot shows a configuration window with tabs for Server, User, Network, Codecs, Certificate, and Sound. The 'User' tab is active. It contains the following fields and options:

- User name: 212
- displayed name: KCCVoIP-8-8
- voicemail number: *97
- (checked)
- password: (masked with dots)
- authentication name: 212
- phone number: 212
- 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 on the VoIP server



SOHO KCCVoIP PBX GUIDE

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 is much the same as above. You only need to set the appliance {VoIP PBX server} address, 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 another extension, dial a ring group or dial *97 to check voicemail etc.

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