Rev.2/22/2024

The Customer's Responsibility includes:

1. Proper setup of the network
2. American Business Phones will provide 1 hour of assistance to the IT Staff or IT Vendor regarding network settings. Any time over the 1 hour assisting the IT staff or Vendor will be billed at $115.00 per hour.

Please read this document carefully, if you have any questions about any of the information provided in this document, please email install@callabp.com

SWITCHVOX
NETWORK REQUIREMENTS
Some or all of the following settings will need to be programmed.

AMERICAN BUSINESS PHONES

Rev.2/22/2024

If needed; the following ports will need to be open and forwarded through the firewall.

It is **required that GEO-IP Blocking be used in your firewall** to limit the IP Address that can have access to the PBX to ONLY those originating from the United States.

Transport	Port(s)	Description
D-Series and P-Series Phones		
TCP/UDP	5060	SIP signaling port needed for phones outside your network
TCP/UDP	5061 (SIP TLS)	For SIP TLS port needed for phones outside your network, **only if** you are using TLS/SRTP transport option
UDP	10000-20000	RTP audio ports needed for phones outside your network
TCP	443	HTTPS port for API access
TCP	80	HTTP port for D-Series phone-firmware access
Sangoma Connect/Talk Mobile		
TCP/UDP	5060	SIP signaling ports needed for phones outside your network
TCP/UDP	5061 (SIP TLS)	TLS port needed for TLS/SRTP transport option
UDP	10000-20000	RTP audio ports needed for phones outside your network
TLS	5095	SIP signaling to cloud-fe.meet.sangoma.com to facilitate audio transfer from Switchvox to Meet when moving a call to the Meet service.
IP Whitelist		Sangoma Connect/Talk SIP Push/Register servers require SIP access to your PBX for the application to work correctly. If we have a restrictive firewall, please ensure that the following IPs can reach your PBX in TCP/UDP ports 5060 and 5061. For more information, please visit How to Set Up and Manage Switchvox for Sangoma Connect Mobile. 159.65.167.207 159.89.179.103 165.227.184.188 167.99.119.203 159.65.186.176 162.243.226.164 165.227.190.186 167.99.119.244 159.65.251.173 165.227.65.164 165.227.210.221 104.131.76.244 159.65.252.186 165.227.115.186 165.227.223.68 143.198.53.243 159.65.253.49 165.227.182.9 167.99.48.91 198.199.67.34

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AMERICAN
BUSINESS PHONES

Rev.2/22/2024

Desktop Softphones and Switchvox Chat for Mobile		
TCP	443	HTTPS port for Desktop client
UDP	10000-20000	RTP audio ports needed for phones outside your network
Web Portal for Admin or User		
TCP	80	HTTP port for remote web
TCP	443	HTTPS port for remote web admin, user and API access

Remote Provisioning Ports:

The following outgoing ports need to be open for some of your devices to allow for remote provisioning:

TCP 80, TLS 443, UDP 3478-3479, TLS 6514, TCP 443

Remote Phones and Softphones:

Ports used by Sangoma Mobile Connect and the IP Phones need to be open to any IP address and cannot be limited to a single IP address or address range due to the nature of how those devices are used. It is **required that GEO-IP Blocking** be used to limit the IP Address that can have access to the PBX to those originating from the United States.

Outgoing Connections:

Please note, the following ports are used by Switchvox for **OUTGOING CONNECTIONS**. Your firewall should allow connections to the Internet on these ports:

Transport	Port(s)	Description
UDP	1194	Must be open to outgoing traffic for Digium / Switchvox Technical Support VPN with allow jumbo frames enabled
ICMP	(any)	ICMP to confirm connectivity to Switchvox servers (In the event that Switchvox is unable to ping, it will report a connection issue.)
TCP	21	FTP when exporting recordings or backups
TCP	22	SFTP when exporting recordings or backups
TCP	25	SMTP when Switchvox sending emails

Rev.2/22/2024

What network ports do I need to forward to my PBX in order to use my SIP Station service?

We recommend forwarding ports UDP/5060 and UDP/10000-20000 for standard Asterisk-based installs. Optimal service will be obtained with the above-mentioned ports. You can lock down port UDP/5060 to the trunk1.freepbx.com and trunk2.freepbx.com FQDNs for additional security, but please note, we do from time to time change the IP addresses associated with these FQDNs. Therefore it is best to use the FQDN and not an IP Address. You **cannot lock down** UDP/10000-20000 to any specific IP address, since the media of a phone call can come from hundreds of different IP addresses.

FIREWALL AND ROUTER

When reviewing your firewall or router configuration, first make sure it is up-to-date (running the latest firmware version). Secondly, check to ensure the following features are **DISABLED**. After disabling a feature, we recommend restarting the router.

- SPI (Stateful Packet Inspection)
- SIP Transformations (SonicWall Firewalls)
- SIP ALG (SIP Application Layer Gateway)
- SIP FIXUP (Cisco Firewalls)
- ALG
- NAT Filtering
- SIP Inspection
- Smart Packet Detection
- **The following is a shortlist of specific 3rd party firewall settings for various makes/models that we know can cause issues with the Switchvox software:**

SWITCHVOX
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Rev.2/22/2024

Fortigate

- 1. Log in to the Fortigate CLI using your favorite method. (SSH or Dashboard)
- 2. At the CLI enter the following commands
-     a. config system settings
-     b. set sip-helper disable
-     c . set sip-nat-trace disable
-     d. end
-     e. config system session-helper
-     f. show (this command will give you a list of session helpers. find the one that says SIP)
-     g. delete <number of the SIP entry from previous>
-     h. end
-     i. config voip profile
-     j. edit default
-     k. config sip
-     l. set rtp disable
-     m. end
- Now reboot your firewall and you should be able to get audio now as long as your firewall rules and policies are correct.
- If this doesn't make calls work correctly then you will need to contact FortiGate support for help.

https://support.digium.com/community/s/article/Router-Firewall-Basic-Configuration-List

SWITCHVOX
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Rev.2/22/2024

LAN Settings

1. The phone system appliance (PBX) or Virtual installation of the Switchvox software will need (1) Static LAN IP Address. Each gateway will also need (1) Static LAN IP Address.

2. If the phone system is going to have Remote IP Phones, Desktop Softphone, or any Soft IP Phones (Sangoma Connect Mobile App), then the phone system will need to have an FQDN (Fully Qualified Domain Name) example: phones.abccompany.com, that is resolvable to the PBX from the LAN and the WAN. IF you are not using any of the types of phones listed above then this is not necessary but will make accessing the PBX for user logins easier.

3. The LAN subnet that the phones will be on will need to provide DHCP, none of the equipment being provided by American Business Phones provide DHCP.

4. If the phones will be using a Voice VLAN, then the switching equipment must be setup with LLDP-MED so the phones can discover the Voice VLAN without interaction from the installer. The switches should have 1 untagged port on the Voice VLAN for the PBX appliance and each gateway. The pass-through port on the phones (PC port) is set as a trunk port so it will pass to the device that is using the pass-through port the untagged or native VLAN of the switch port that the phone is plugged into. In most cases the other VLANs will need to have access to the PBX on the Voice VLAN so inter-VLAN routing will be required.

The PBX is setup to provide QoS. Settings are:

1. EF (expedited forwarding) which is DSCP value 46 for Voice packets and 26 for SIP packets.
2. 802.1p of Voice packets have a value of 6 and SIP 802.1p value of 3.
3. Each IP phone will need to plug in to a network data switch that provides PoE (Power over Ethernet) or will be required to have a power supply for each IP phone at an additional cost.

Rev.2/22/2024

Wi-Fi IP Phone Requirements:

Wi-Fi IP Phones both corded and cordless require a properly configured wireless network for them to operate correctly.

Supported wireless protocols are 802.11a/b/g/n/ac.

802.11e (WMM) should be enabled, 802.1p is also supported.

In a multi access point environment 802.11r needs to be enabled and 802.11k and 802.11v should also be used if possible.

Some Wi-Fi phone models need outgoing connections to the internet for setup and operation.

SSID's for Wi-Fi Phones should be on the same VLAN as Switchvox and inter-VLAN routing must be in place.

PoE consumption requirements for each phone model:

D60/62/65: 6Watts
D80: 10 Watts
P310: 2.5Watts
P315: 2.5Watts
P320/P325 4.5Watts, 9.5Watts with USB
P330: 8.2Watts, 5Watts per USB, 18.2Watts total
P370: 8.5Watts, 5Watts per USB, 18.5Watts total