Network Configurations L2
Network Configuration Specifications

In order to ensure the optimum Quality of Service (QoS) for your Cloud Communications Solution, we recommend adhering to the specifications listed in this document. Any deviation from these specifications can affect your Quality of Service (QoS). All configurations have been thoroughly tested and approved on our SwitchConnex platform and generally in any Hosted environment.

Cabling Requirements for Hosted PBX

- Cat5e (or higher) cable is required for all voice devices on the premise LAN.
- Splitters, Combiners, Extenders, RJ45 converters, or other modifications on cable runs to voice devices are not recommended. Use of cable modifiers can result in lower QoS.
- Hubs (repeaters) cannot be used on VoIP LAN segments.
- Wi-Fi VoIP devices are supported through Wi-Fi Access Points. Wi-Fi VoIP devices and WAP usually operate in standard Wi-Fi frequency IEEE 802.11 and operates in the 2.4, 3.6 and 5.0 GHz band.

Cabling Requirements for SIP Trunks

- Cabling and connections between the premise-based PBX and handsets can remain unchanged.
- SIP Trunk (IP PBX trunk port or TDM PBX VoIP Gateway) connects directly to a dedicated voice LAN segment (via switch, VLAN switch, edge device, or edge router LAN port).
- SIP Trunks typically connect to a port on the internet edge router (on converged networks).
- Do not connect SIP Trunks directly to the internet (via broadband).

Switch Requirements for VoIP (Physical of VLAN)

- PoE (Power over Ethernet) - Most SIP phones are only powered via data connections and must be connected to a PoE switch.
- PoE injectors or extenders are not recommended.
- SNMP (Simple Network Management Protocol) managed switches are recommended as they can be managed remotely (ex- reboot, reset, etc.).
- Do not mix IP and SIP devices on VLAN segments.
- Do not use unsupported ATAs (Analog Telephone Adapters). Contact ACC Telecom for a list of supported ATA's.

SOHO Networks for VoIP

- SOHO Networks experience relatively light traffic loads and VoIP performance is usually acceptable on a single segment on converged LAN/WAN. Voice and Data may not be divided into separate LAN segments with Switches or VLANs.
- QoS SOHO router is strongly recommended.

Router Requirements for VoIP

VLAN Support

- On converged networks, configure router to keep voice and data segments separated internally. Voice and data segments must be invisible to each other as they pass through the router.
- This is often accomplished by configuring an internal VLAN within the router. The router's internal VLAN requirement is not dependent on whether the LAN includes VLAN segments or not.
- A VLAN capable router is always required if VLAN switches are used.

Capacity

- An available port on the internet router is needed for the voice (VLAN or Physical) segment.
- If VLAN switches are used, this port is configured as a VLAN trunk.

IP Address

- A publically routable (static) IP address is required for the internet router.
- If the edge device is not integrated in the Internet router, it also requires a public IP address.
- Do not use a private IP address and NAT (Network Address Translation) on any routers that VoIP traffic will cross. NAT creates VoIP network problems because it cannot locate IP addresses embedded in SIP packets.
- VoIP packets should not traverse multiple routers.

Router Requirements for VoIP *Cont'd*

DHCP Options 66

- Routers should be configured as DHCP servers with Option 66 enabled. This allows devices on the voice segment to receive IP addresses and configuration files upon power up. This requires Transparent Proxy mode to be enabled on Edgemarc devices.
- There can only be one DHCP server per LAN segment.
- DHCP is not required for SIP Trunks.

WAN Link Redundancy

- If available, the feature uses multiple WAN ports to provide load balancing and failover.

Proxy Settings

- If available, Transparent Proxy mode should be enabled. This setting should always be enabled on EdgeMarc routers.

Firewall Requirements for VoIP

As VoIP uses SIP and RTP, firewalls must be configured to allow these protocols onto the LAN. The following firewall ports must be opened:

- UDP or UDP/TCP 5060/5061
- Required for SIP: UDP or UDP/TCP port range 10000-30000
- Required for RTP: Range can vary based on end user devices

Edge Device Requirements for VoIP

QoS functions are often integrated within other devices, such as routers, firewalls and switches. ACC Telecom refers to QoS devices as Edge Devices.

Recommended Implementation

- VoIP networks should use an integrated edge/router/firewall.
- On converged networks, it should be the primary internet access device for all voice and data traffic. This provides the best possible QoS (Quality of Service) because voice traffic gets prioritized and forwarded to the voice LAN segment as soon as it's received from the WAN.

Support for Existing Implementation

- Install the edge device outside the internet router/firewall (between the router/firewall and WAN).
- Both the edge device and internet/router/firewall should have publically routable, static IP addresses.
- Configure Bridged Proxy ARP on the edge device so it will be "invisible" to the existing router/firewall.

Edge Device Requirements

An edge device is required for converged and dedicated VoIP networks. The edge device should be installed as close to the "edge" of the network (LAN/WAN boundary) as possible.

- IP Address: A publically routable static IP address is required for the edge device. Do not use a private IP address and NAT (Network Address Translation) on the edge device. NAT creates VoIP network problems because it cannot locate an IP address embedded in SIP packets.
- Enable Syslog and SNMP
- Disable SSH or change the default password
- Enable transparent proxy mode on Edgemarc devices and if available on other make/models
- Router and firewall settings are also required for integrated edge/router/firewall devices:
  - DHCP Option 66
  - VLAN support
  - Open firewall ports for SIP and RTP

WAN Connection Requirements for VoIP

High speed Broadband connection (Cable or Fiber Optic) or T1 Lines is crucial to support time sensitive VoIP Service. Adequate amount of bandwidth is required to have an optimal Quality of Service (QoS).

- Determine existing bandwidth
- Determine how much bandwidth is required for Voice & Data traffic together.
  Consider:
  - Concurrent Active VoIP phone lines
  - Codec to be used G711 (full Quality Audio), G722 (HD Quality Audio), or G729 (Compressed Audio)
  - Data traffic on the network

We recommend to prioritize Voice traffic over Data traffic and to implement a post installation procedure to continually monitor bandwidth utilization.

Supported Router/Firewall Devices

A wide variety of edge router/firewall devices are available; however, we recommend the devices listed below for dedicated and converged VoIP networks. These devices have been tested in our lab. We provide set-up instructions as well as configuration and troubleshooting assistance.

- Edgewater Networks – EM 4550, 4750, 4601 (PRI)
- Adtran – TA908e (PRI and Hosted PBX)
- Peplink – Balance 20, Balance One, and Pepwave SOHO

Soft/Desk Phone Requirements

Soft Phones

- All Soft Phones and IP Devices must be SIP compliant and enabled.
- Supported Codecs: G.729 (recommended), G.711 and G.722.

System Requirements (Bria Softphone)

- Processor Minimum: Pentium 4® 2.4 GHz or equivalent
- Optimal: Intel Core Duo or equivalent, Video Card with DirectX 9.0c support
- Memory Minimum: 1 GB RAM, Optimal: 2 GB RAM
- Hard Disk Space 125MB
- Operating system: Microsoft Windows XP Service Pack 2, Windows Vista (32-64 bit), Windows 7
- Connection: IP network connection (broadband, LAN, wireless); constant internet connection
- Sound Card Full-duplex, 16-bit or use USB headset

Desk Phones

The recommended SIP telephones have been extensively tested and are known to work well with our services. Our recommended models are officially supported and each release is tested to ensure continued compatibility. Supported phone manufacturers include, but are not limited to: Cisco, Grandstream, Polycom, and Yealink. Please contact ACC Telecom for a complete list of supported phone manufacturers and models.

On a Gigabit Network, all devices must support Gigabit functionality. If a slower device joins a gigabit ready hub, transfer speeds will crawl only when you access that particular device which may result in low QoS. End users could daisy chain the computer off of the VoIP phone but on a data-heavy work environment (ex- graphic designers, video editors, etc.) this could potentially slow down the computer's data transfer. In this case you may consider running a second line for the phone so it is kept separate from the computer's gigabit connection to the router/switch/server.

Browser Compatibility

While the portal minimizes browser dependencies due to our user interface (UI) design, the following browsers have been the target of our testing. Other browsers may work, but the extent of testing is with the following browsers based on the wide scale adoption and use.

Supported Browsers

- Mozilla® Firefox® version – 42.0 (32 & 64 bit)
- Google Chrome™ version –46.0.2490.86 (32 & 64 bit)
- Apple® Safari® version 7 onwards
- Microsoft® Internet Explorer® 9, 10, 11

Power

In order to ensure uninterrupted service in the case of a power outage, the end user may want to implement power backup equipment.

Power over Ethernet (PoE)
PoE provides power to Voice over IP telephones over the Ethernet cable. PoE provides both data and power connections in one cable, so equipment doesn't require a separate cable for each need. All phones can be powered by single source depending upon the Switch. ACC Telecom recommends Managed Switches as they provide more control over LAN traffic.

- **IEE802.3af** - PoE standard provides up to 12.95W of power
- **IEE802.3at –** PoE standard (PoE+/PoE plus) provides up to 25.5W of power

Uninterruptible Power Supply (UPS)
UPS are electrical apparatus that provide power to equipment through built-in batteries which allow your network devices to remain operational during a power outage.

ACC Telecom recommends implementing a Managed PoE Switch to power all phones and utilizing an uninterruptible power supply (UPS) to prevent any loss in service.

Local Number Portability (LNP) Requests

The following LNP requests are supported:

- Local or domestic numbers (Local Number Port, or LNP)
- Toll free numbers (Responsible Organizations, or RespOrg)
- vFax (Virtual Fax)

The following LNP requests are not supported:

- International phone numbers

To help guarantee a successful port request, we recommend the following:

- Obtain a Customer Service Record (CSR) and/or Invoice from the losing Carrier. Customer's information on the port request must match the information as it appears on the Customer Service Record (CSR)/Invoice exactly.
- Customer must complete a Letter of Agency (LOA) within 30 days of port submission. LOA must be typed and free of errors.
- CSR, Invoice & LOA must be in PDF format with size less than 4MB each.
- Customer must account for all active telephone numbers at the customer's premise when porting (i.e.- must indicate the numbers that are to be ported and the numbers that will not be ported).

Requirements for Advanced Features

Communications Client

- Browsers with WebRTC Support
  - Mozilla® Firefox® version - 42.0 (32 & 64 bit) or higher
  - Google Chrome™ version – 46.0.2490.86 (32 & 64 bit) or higher
- Display: 1280x800 minimum supported resolution
- Processor Minimum: Intel Core i3 2.5 GHz, AMD Athlon X2 2.0 GHz or equivalent
- Memory Minimum: 1 GB RAM
- Sound Card Full-Duplex, 16-bit or use USB headset
- Web Camera

Outlook Plugin

- Microsoft Office version 2010 or 2013
- Net Version 451 or higher

Click Connex Plugin

- Browsers with WebRTC Support
  - Mozilla® Firefox® version - 42.0 (32 & 64 bit) or higher
  - Google Chrome™ version – 46.0.2490.86 (32 & 64 bit) or higher