Network Preparation for SIP Traffic

Considerations

Internet Connection

- Phones must be connected using a reliable high speed business class Internet connection and should have the following:
  - Public IP address required – Static IP preferable; DHCP is supported.
  - Latency less than 150ms. (<100 ideal) – measures the amount of time it takes for a packet of data to get from one designated point to another.
  - Jitter less than 20ms – jitter is the variation in the time between packets arriving, caused by network congestion, timing drift, or routing changes.
  - IP Fragmentation is acceptable.
  - DNS resolution – process to resolve a hostname to an IP address.
  - Packet Loss should be 0% - Packet loss occurs when one or more packets of data traversing across the network or Internet fail to reach their destination.
- Download Speed – insufficient download speeds may impact inbound traffic to the VoIP user, including the ability to hear callers. Each call utilizes approximately 85Kbps of download bandwidth.
- Upload Speed – insufficient upload speeds may impact outgoing traffic, including the ability for callers to hear you. Each call utilizes approximately 85Kbps of upload bandwidth.
- Home networks may not be designed to manage Quality of Service (QoS), users should limit functions that utilize large amounts of bandwidth usage while on a call such as large file downloads, Internet television, and streaming of audio & video.
- For Businesses using Dual WAN Connections/Providers, you will need to ensure that the phones continue to send SIP traffic via the same Public connection IP with which they registered. If the phones try to send traffic via a different IP from which they registered, then phones will experience problems. Using routing rules or VLANs will be needed to force traffic bound for Dobson IPs/Domains out a single connection. This is most likely to be needed if you are load balancing Internet traffic between different carriers. Primary/Secondary failovers will usually work because if the primary fails, the secondary connection will be used and the phones will register using the new connection, even if a phone reboot is required.

Physical Network Wiring

Depending on the phone model, IP phones may come with a built-in network switch. The LAN port should plug directly into the Internet router or public switch. Wireless networking should be avoided to take advantage of built-in QoS on the IP Phone and to limit outside Radio Frequency (RF) interference. While IP Phones do have ports that allow other network devices plugged into the back, there are limitations to the expected performance of the phone and PC when used in this way and therefore, this is not the recommended deployment. Problems resulting from this configuration usually cannot be resolved without separation of the network elements. For best performance the phone and PC should have separate connections directly to the ethernet switch.

Software Based Phone Recommended Requirements

- Business class desktop or PC system with ample processing and memory to run all required applications.
- Business class headsets such as Plantronics (www.headsets.com)
- Updated BiOS, Firmware, Drivers, Operating System Patches and Antivirus.
- Utilize physical network Ethernet cable and disable wireless networking.

Firewall configuration

- Make sure you can bridge your router to your provider’s modem. Routers that are not bridged can cause problems with voice over IP installations.
- Never use more than one router or (Network Address Translation) NAT gateway on the network at a time as this will cause problems for IP Phones when they attempt to do NAT.
- Built-in ALG (Application Layer Gateway) may need to be disabled. See SIP ALG Section.
- Update router/firewall firmware to resolve any known device issues.

If you experience problems with phone registrations, one way audio, or phones not ringing on inbound or outbound calls check the local router's firewall configuration settings.
The most likely issue relates to the SIP ALG service being enabled.

Depending on router configuration, also allow (whitelist) all traffic from our IPs:

SIP Traffic:

- 209.183.174.0/24 (new in 2021)
- 72.18.40.0/24 (new in 2021)
- 209.183.160.0/24
- 162.244.96.0/21
- 63.209.193.0 /24
- 208.85.134.0/24
- 208.67.14.0/24
- 208.67.15.0/24
- 204.16.54.0/24
- 204.16.55.0/24
- 209.55.25.32/27

Messenger

For Messenger, please ensure the following ports are opened:

http:	80
https:	443
XMPP:	522, 1081
SIP:	5060-5080 (both TCP and UDP)
RTP:	0-65535
SOCKS:	52644-52645

Web Services

You should allow firewall access to the following domains:

- portal.dobson.net
- voipa.dobson.net
- voipb.dobson.net
- voipc.dobson.net

Please be advised if you are using app.mymtm.us or ews.altvoip.com (for client services such as Unity, BW Anywhere, or Receptionist) and only allowing http/https/tftp traffic to the specific IP addresses, you will need to allow traffic to the new IP addresses below:

- 63.209.193.0/24
- 204.16.52.0/24
- 162.244.96.0/21
- 208.67.14.0/24
- 208.67.15.0/24

If this is not possible, ensure that these ports and port ranges are allowed to pass through:

TCP 1720-1728
TCP 5060 - 5080
UDP 5060 - 5080 or
UDP 1024 - 65535

As for other services which could be used for phone configuration and firmware upgrades, those can and will change and are dependent upon the endpoints. We currently use:

FTP
HTTP
HTTPS
TFTP

Of these protocols, the only one which is likely to be blocked by default is TFTP.

Note:

In cases where you have insufficient bandwidth or there is jitter/packet-loss created by your Internet connection, upgrading and properly configuring your router will not resolve all issues. These cases require that you to make changes to your Internet connection and/or upgrading to a higher-speed Internet connection. Audio quality issues can also be caused by other issue such as heavy Internet traffic involving the use of video over the Internet. This can impact the bandwidth requirements of your Internet connection far more than voice usage and needs to be factored in.

SIP ALG

Technical Tip – SIP Fixup (SIP ALG)

Dobson would like to remind our Partners to make sure that your premise routers do not have “SIP Fixup” (or equivalent) enabled. Not all routers support this but may do and they are defaulted as turned on which will lead to phone failures. Different manufactures may name the functionality differently depending on the make and model of your device. Please refer to your router documentation or contact your vendor for specific help.

On Cisco devices it is referred to as "SIP Fixup" and it is enabled by default on both routers and PIX devices. Because this is a default setting, no indication of it being on or off is visible in the configuration. “SIP Fixup” causes many problems and, if enabled, will cease to work on the Momentum network as we continue to upgrade our equipment with newer releases.

There are limited options to work around an active SIP ALG, so disabling that function in your router is of upmost importance.

For some common models and helps turning off SIP ALGs, please visit our online references:

https://www.dobson.net/sipalg