Lumen® Cloud Voice Minimum Network Requirements

Follow these network configuration guidelines to optimize your experience with Lumen Cloud Voice (LCV) and collaboration services.

Internet connection

Lumen Cloud Voice (LCV) is a cloud-based voice service that requires an internet connection. This can be provided through Lumen network services or another provider.

Speed

Your internet connection must be sized to allow for both your data and voice or video calls. Typically, each package ordered requires an additional 100-300 Kbps of bandwidth for the LCC features. Insufficient bandwidth can cause inferior call quality.

Electrical outlets for devices

A power adapter, or power brick, is provided for each Lumen supplied telephone device and will require an electrical outlet..

Power over Ethernet (PoE)

PoE is supported by all phones sold or leased through Lumen. A PoE switch is required. ATAs do not support PoE.

Network configuration

SIP ALG

Disable. Protocols that modify SIP packets can cause problems, specifically with inbound calls. Most routers have SIP ALG enabled by default and will need to be disabled. The router/firewall must not manipulate SIP or RTP packets.

UDP timeout

Disable or set to a minimum of 24 hours.

NAT binding timer

Set to a minimum of 30 seconds or longer.

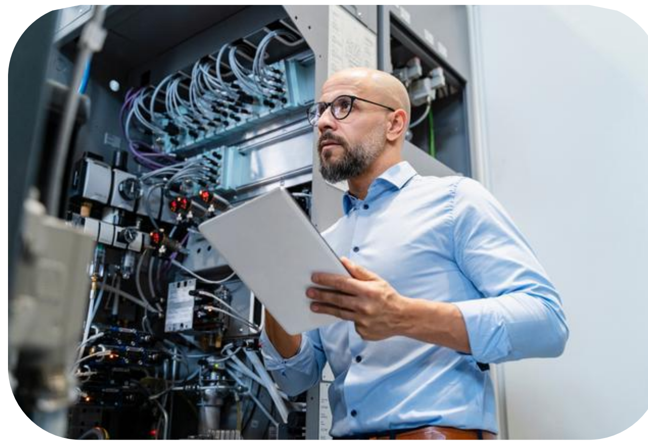
MTU

Set to 1400 to avoid fragmentation.

Ports

The following Internet Protocol communication ports and protocols should be allowed access through the firewall for routing to the internet:

- **RTP** UDP 20000-40000
- **SIP** UDP 5060-5065
- **Provisioning** TCP 80,443



QoS requirements

For better call quality, prioritize voice and video over data in your QoS (quality of service) settings. Configure QoS to mark session initiation protocol (SIP) and real-time protocol (RTP) packets to and from the LCC platform as high priority using Differentiate Services Code Point (DSCP) to prioritize these packets over lower priority data and web traffic.

- Voice traffic must be prioritized (DSCP EF/46)
- Separate VLAN for voice traffic (if recommended)
- Assume 100 Kbps (up/down) per call
- Jitter buffer configuration

IP addresses

The following IP addresses should be reachable:

Network protocol
216.239.35.0/32
216.239.35.4/32
216.239.35.8/32
216.239.35.12/32

Confinguration files
18.217.97.178/32
18.219.216.7/32
173.244.45.14/32
198.204.63.27/32
52.15.237.101/32

Call routing
198.204.63.0/24
173.244.45.0/24
208.93.8.0/24
208.93.9.0/24
65.75.218.0/24
65.75.219.0/24
2620:91:c005::/64
2620:91:c001::/64

Common issues and solutions

Device does not power on

- Check that the power adapter is plugged in and cables are seated fully.
- If using PoE, ensure power is enabled on the switch port and the ethernet cable is working and PoE compliant.
- Reminder: ATAs do not support PoE.

VoIP phone is asking for a password

Equipment will arrive pre-provisioned and must be connected to the internet to fully download its configuration. If the phone asks for a password and does not perform an initial configuration pull, it usually indicates a lack of internet connectivity.

- Ensure that your network is connected to the network port on the VoIP phone via an ethernet cable and that it is physically connected to a port on your router.
- Do not use the PC port on the back of the phone, as it is only intended for passing internet (IP address) to the computer.
- If needed, a wireless dongle can be purchased separately for some devices to enable Wi-Fi connectivity. Due to Wi-Fi bandwidth restrictions, it is not recommended to use more than two lines when connecting wirelessly.

Calls drop after 15 minutes

Around 15 minutes into a phone call, a SIP re-invite is sent to ensure the call remains active. If the SIP re-invite is not acknowledged, the call will drop or lose two-way audio. This issue is typically caused by SIP ALG being enabled and/or the UDP timeout being too short, causing frequent port registration changes.

- Ensure that SIP ALG is disabled in your router settings.
- Confirm the UDP timeout is either disabled or set to at least 24 hours in your router settings.

One-way audio

One way audio can happen when RTP packets are lost. When the router fails to utilize NAT/PAT appropriately, the RTP packets can be dropped, or sent to the incorrect device.

- Ensure that SIP ALG is disabled in your router settings.
- Confirm the UDP timeout is either disabled or set to at least 24 hours in your router settings.
- Verify the ports necessary for RTP are open. The port range for RTP for this service is UDP 20000 – 40000.

Device has dial tone but cannot receive incoming calls

- Confirm that do not disturb (DND) is disabled on the device.
- Ensure that SIP ALG is disabled in your router settings.
- Confirm the UDP timeout is either disabled or set to at least 24 hours in your router settings.

Calls stay connected after call ends

Several factors may cause this issue. If the problem persists, please contact support. The CPC loop current interruption signal is disabled by default on the ATA. Support can enable this setting to improve compatibility with some PBXs. While this setting can be changed locally through the LAN, the cloud configuration for the device also needs to be updated by support.

Latency concerns

Your phone system is registered to the Lumen network through one of two connection points. The connection closest to your business may result in less latency.

Ashburn

- Hostname: ashprdsla01.alianza.com
- New IP: 198.204.63.129

Sacramento

- Hostname: sacprdsla01.alianza.com
- New IP: 173.244.45.129

Why Lumen?

Lumen provides a secure solution across its owned and operated network, with a single relationship of subject matter experts who combine a tailored, comprehensive onboarding experience with meticulous monitoring, support and ongoing management.

866-352-0291 I lumen.com I info@lumen.com

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