NETWORK READINESS

To successfully implement a Voice-Over-IP (VoIP) solution, the Customer's data Network must be able to support the demands of voice traffic concurrent with the data demands. Thus, a high performance network will need to be in place or established. ***Without the successful arrangement of a high performance network, a VoIP implementation may have undesirable performance.***

- RingCentral requires that each customer location test their broadband connection for quality and capacity. For voice quality, we recommend that you use a high-speed DSL, cable, or fiber-optic connection with dedicated upload and download bandwidth for voice of ***90Kbps*** or higher for each voice line you plan to run. Please use these Internet connection tests to verify your connection.
    - http://www.ringcentral.com/support/capacity.html
    - http://www.ringcentral.com/support/qos.html

You will also want to ensure the following port ranges are open on the firewall for your router. These ports in the firewall must be open for the phone system to work properly:

5060-5090 UDP

16384-16482 UDP

8000-8200 UDP

20000-65535 UDP

RingCentral also recommends that the following be in place in the Customer's Legacy Network to support a successful VoIP implementation:

- Switched media (no hubs)
- If your firewall or modem supports SIP ALG (Application-Level Gateway) or SPI (Stateful Packet Inspection) please disable these functions. This may be located in the NAT (Network Address Translation) options of your device.
- If the firewall supports the option, set NAT to “Open.”
- Minimum 2MB Ethernet LAN (no Token Ring)
- Category 5 or better cabling for all telephone stations
- Adequate bandwidth to support voice, video and data traffic volume demands over the network. Each VoIP call can consume approximately 90Kbps of bandwidth (2MB per 3 VoIP devices in a single voice and data network)
- Low delay to ensure a good quality voice conversation (< 125ms is recommended)
- Minimal packet loss must be one (1)% or less between endpoints to ensure parts of a conversation are not distorted or lost, especially during bursty data traffic flows
- Low jitter (less than 20ms) to ensure that the next IP packet can be played at the destination CODEC without requiring large jitter buffers.
- Separate VLAN for voice traffic is recommended for optimal quality of service, but not required
- Quality of Service (QoS) throughout the VoIP path by placing only voice in the highest priority queue to ensure voice gets the bandwidth and latency required for effective voice communication is strongly recommended.

It is the Customer's responsibility to make sure that both the Customer Network LAN and WAN infrastructure will meet and support VoIP specifications that provide acceptable VoIP quality. Network reconfiguration and/or upgrades of the data network (including LAN/WAN hardware/software) are the responsibility of the Customer and are outside the scope of this SOW.

RingCentral Network Requirements