Welcome to **Brightspeed Voice+** *with RingCentral*

The following guidelines describe how to configure your network to support an optimal experience with Brightspeed Voice+ with collaboration services.

Internet Bandwidth

1. Brightspeed Voice+ is a cloud-based voice service that requires an Internet connection. Brightspeed has multiple network services available to meet both your data and voice needs but can support for Brightspeed Voice+ over an alternate internet network provider, as an option.
2. Your Internet connection must be sized to support both your data bandwidth needs and the quantity of simultaneous voice or video calls needed by your business. Insufficient bandwidth can impact the quality of your audio or video communications. The following guideline should be used to determine bandwidth needs:
    a Voice call from a desktop IP Phone requires 100kbps of bandwidth.
    b Voice only calls using the RingCentral desktop or mobile application requires 512Kbps.
    c One to one video calls using the RingCentral application requires 3Mbps.
    d One to many video calls (meetings) requires 6Mbps bandwidth.

Local Area Network (LAN)

1. BV+ IP phones and Analog Terminal Adapters (ATAs) operate as a Session Initiation Protocol (SIP) device on the network and utilize a Local Area Network connection, just like a computer, versus traditional telephone jack wiring.
2. Power over Ethernet from the network switch can be used to power the IP Phone and a power brick and cord are also provided if Power over Ethernet is not available.
3. LAN bandwidth and switch capacity should be sized to support the quantity of IP Phones or Analog Terminal Adapters (ATA) ordered plus the data requirements of the business.
4. Ethernet/data network wall jacks should be available at locations where the IP Phones will be used. IP Phones can share the same network connection with a computer by utilizing the switch built into the phone, a practice called “daisy chaining.”
5. Dynamic Host Exchange Protocol (DHCP) should be available to provide a network IP Address to the phone.
6. Domain Name Service (DNS) or DNS Relay is necessary to allow the IP Phones to register to the cloud communication platform. DNS SRV and DNS A type records must be supported.
7. The firewall must allow SIP and RTP for IP Phones and ATA’s to place and receive calls.
8. Network switches that support Virtual Local Area Networks (VLAN’s) should be configured to provide a dedicated, separate path for voice versus data traffic as well as maintain quality of service (QoS) settings from the IP Phones.

Wide Area Network / Firewall

1. Quality of service (QoS) settings are used to prioritize voice and video over data applications to provide a better call quality experience. Customer router should be configured to honor and prioritize QOS markings from the LAN switch for Session Initiation Protocol (SIP) and Real Time Protocol (RTP) packets to and from the BVP platform.
2. Customer router/firewall must not manipulate the SIP or RTP packets at the application layer. If any CPE devices can function as a SIP Access Layer Gateway (ALG) or SIP Proxy, the feature must be disabled.
3. Customer router Network Address Translation (NAT) expiration timer should be set to 5 minutes to allow for IP Phone registration requests.
4. The following IP supernet address ranges should be reachable via Access Control Lists (ACL's) so the BV+ endpoints can register, obtain their configuration files and get firmware updates:
    a. RingCentral Supernet Addresses
        i. 66.81.240.0/20
        ii. 80.81.128.0/20
        iii. 103.44.68.0/22
        iv. 103.129.102.0/23
        v. 104.245.56.0/21
        vi. 185.23.248.0/22
        vii. 192.209.24.0/21
        viii. 199.68.212.0/22
        ix. 199.255.120.0/22
        x. 208.87.40.0/22

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

Purpose	Protocol	Domain Name/Address	Ports
Voice Signaling	SIP	IP supernets	TCP-5090, TCP-5099
	SIP	IP supernets	UDP-5090, UDP-5099
Voice Secure Signaling	SIP/TLS	IP supernets	TCP-5096, TCP-5098
Media/Secure Media	RTP/SRTP	IP supernets	UDP-20000-64999
Network Time Protocol	NTP	ntp1.ringcentral.com	UDP - 123
	NTP	ntp2.ringcentral.com	UDP - 123
Poly Phone Provisioning	HTTPS	pp.ringcentral.com	TCP - 443
	HTTPS	ztp.polycom.com	TCP - 443
Poly Firmware Update	HTTPS	pp.s3.ringcentral.com	TCP - 443