Hosted Cisco Best Practices

NETWORK PREP GUIDE

Version 1.0

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Contents

Foreword

This Guide is designed to give the CallTower customer administrator a reference for preparing their network for the use of CallTower Cloud based Cisco Unified Communications Manager. As a Unified Communications platform, Cisco Unified Communications Manager is a bandwidth intensive product that relies heavily on internal network setup and bandwidth to provide end users with a quality experience. While following these guidelines will better prepare your network for its use, CallTower cannot predict overall performance of the product across internal network connections, ports and firewalls due to the influence of user behavior on the network itself. Adhering to the guidance in this guide gives you and your organization a better opportunity to prepare and adjust your network to meet the needs of the product, allowing you to fully benefit from the robust feature set provided by Cisco Unified Communications Manager. CallTower is dedicated to be with our customers every step of the way and your resource for all your implementation needs!

Professional Implementations

Count on a seamless implementation and setup of Hosted Unified Communications and Collaboration Solutions. Our professional experts have deployed thousands of users and will ensure your success with a personal, dedicated Project Manager. Your Project Manager will navigate you through all phases of your implementation including porting, call flow set-up, CallTower device configuration, and training. They will offer you a clear understanding of the process so there is no confusion on duties and responsibilities. Our Project Managers work fast and efficient. With an implementation schedule under the industry average, we will get you up and running in no time.

Putting the Pieces Together

There is a lot of moving pieces when it comes to implementing a robust unified communications solution. You need circuits, equipment, software setup and much more. Your Project Manager works closely with our internal Logistics and Telecom teams to make sure your organization is receiving all of the necessary equipment and that circuits are getting installed on time.

Successful Projects Require the Right People

It is important to have the right people managing and providing input in a project. Comprehensive planning sets up a project for success from the start. All stakeholders should be on board during the planning process and always know in which direction the project is going to go. Clients do share a small portion of responsibility in the success of the project so it is also important they have right people assigned.

Understand Your Current Network

Cisco traditionally is more equipment centric and requires some basic knowledge of LAN infrastructure. If customers choose to provide their own switching infrastructure due to cost savings it is important that they have a good understanding of it. In order to avoid issues or risks it is recommended that the customer have an IT staff or 3rd party to assist with the migration or installation. (Nothing advance, just basic understanding and configuration of VLANs, DNS & Firewalls). CallTower can recommend a certified 3rd party installer at an additional cost. Their expertise would primarily be on the voice network. Any changes needed to the data network such as DNS or firewall setting would need to be done by the client's system administrator. For the majority of our customers, roughly 85%+, can self-install with full remote support from their CallTower project manager and their technical resources.

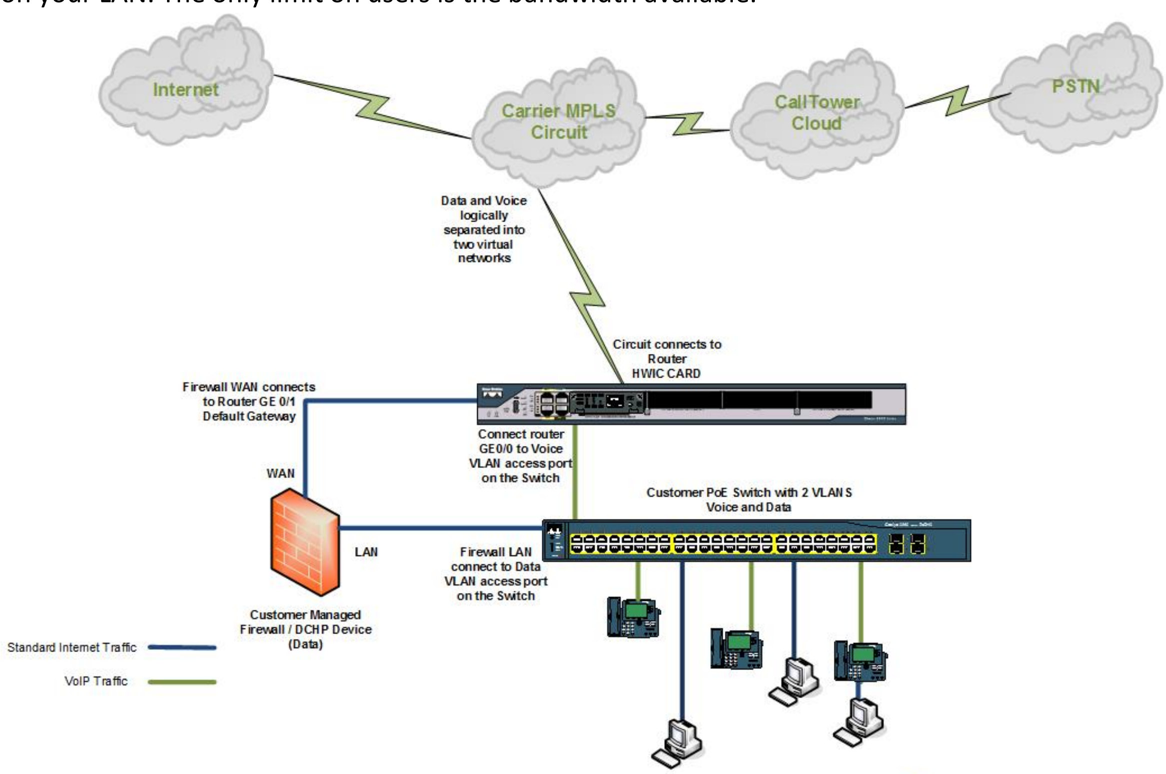
Network Design & Network Connectivity with CallTower

Design considerations will greatly influence the overall success when integrating VoIP with an existing network. Network equipment will support virtual LANs (VLANs) and quality of service (QoS). Using VLANs, you can separate voice and data traffic but have them co-exists on the same medium. VLANs offer a simple way to provide a distinct level of service as well allowing the individual traffic types to be monitored. QoS is a method of distinguishing IP packets so they can be treated by distinct policies.

Your VoIP implementation should prioritize traffic to prevent existing data traffic from undermining voice communication integrity. This process requires matching specific fields within either the Layer 2 or Layer 3 headers inside the data units. Layer 2 QoS, for example 802.1p, allows for multiple levels of prioritization by tagging frames with certain fields. At the switch port level, QoS can then be provided to traffic entering the port. Therefore, as data and voice traffic simultaneously enter the device, voice traffic can be sent first and the data traffic queued for delivery. Connectivity to CallTower's hosted Unified Communications Manager service can be achieved in several ways.

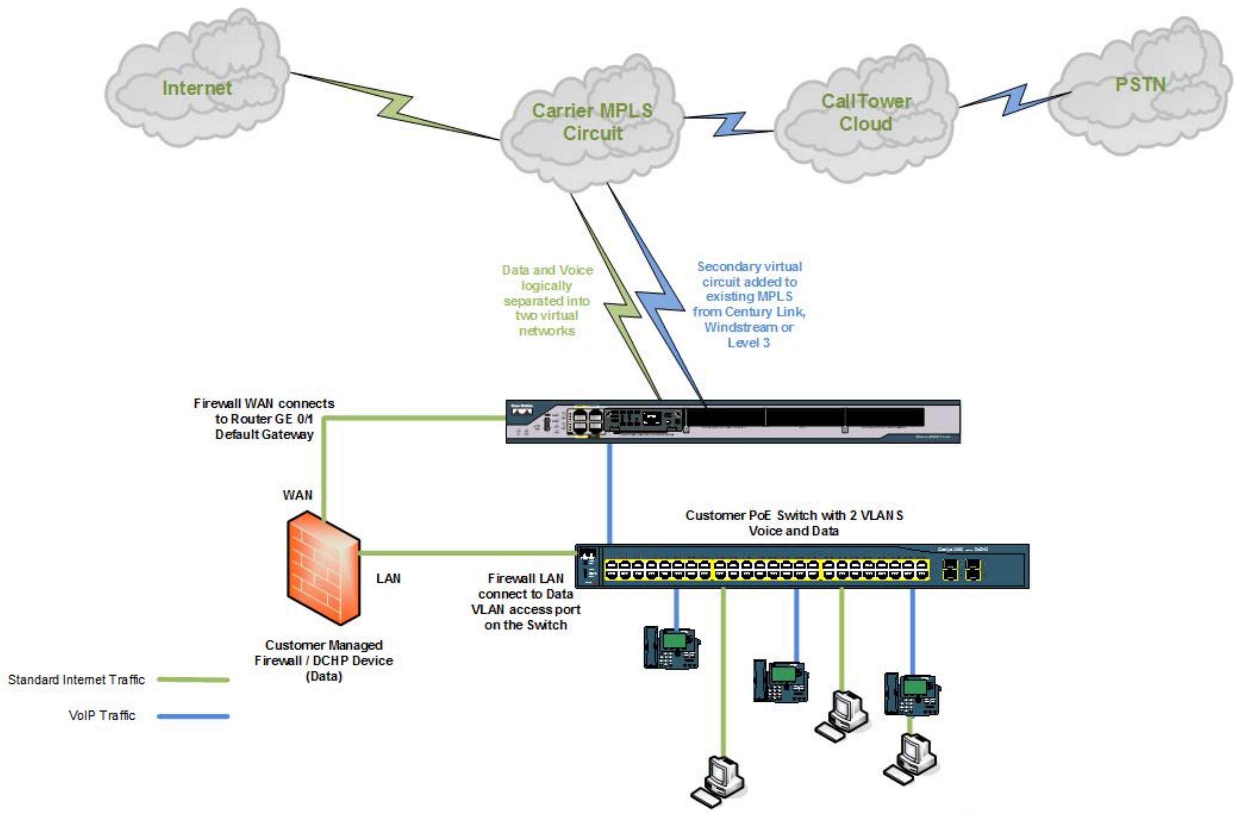
Connectivity Option #1

Dedicated MPLS from CallTower provides QoS and has been traditionally the best way to provide service. In this option a CallTower managed router is provided to terminate the MPLS circuit. The route provides connectivity, option 150 and DHCP to the phones. CallTower manages the network address of the phones on your LAN. The only limit on users is the bandwidth available.



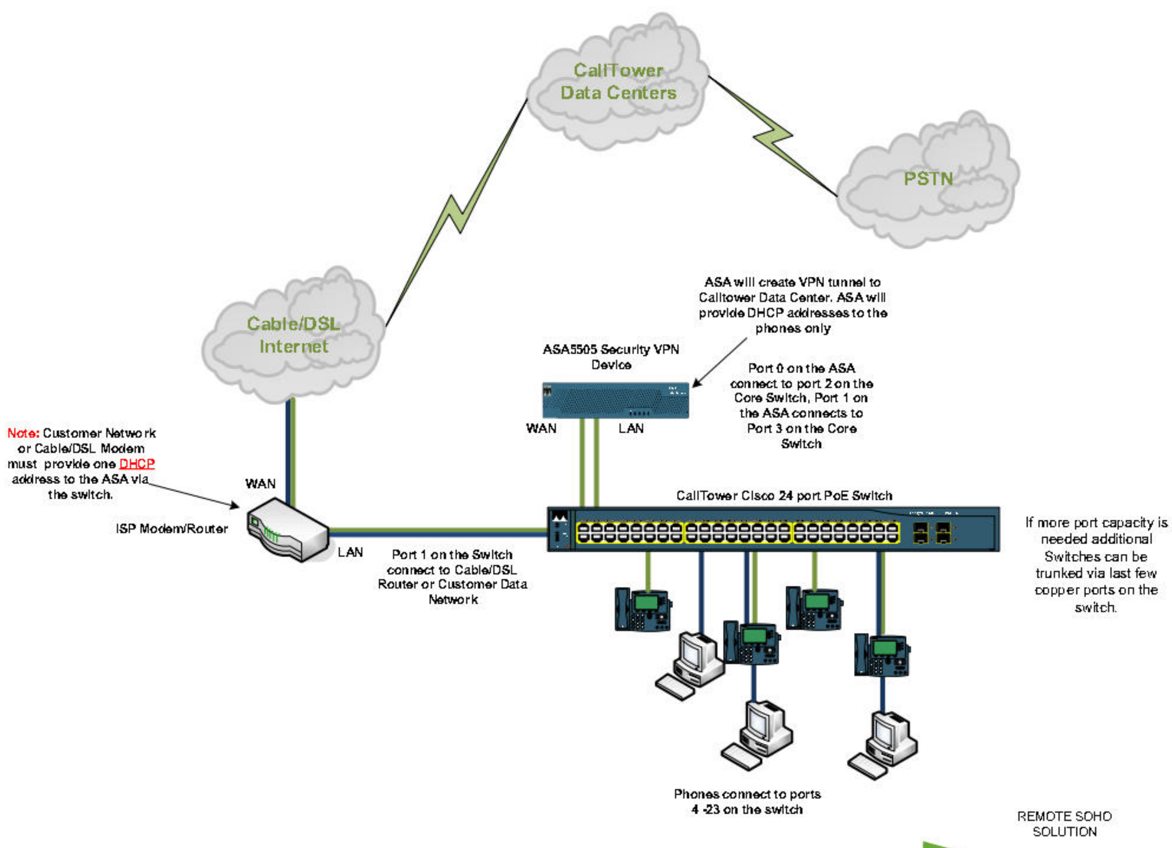
Connectivity Option #2

Customer existing MPLS from one of three MPLS carriers (Century Link, Windstream, or Level 3). A VRF/CUG is ordered through your current MPLS provider. Your service provider will provision access to the CallTower network on a layer 3 VRF or layer 2 VLAN. CallTower & your IT staff work with your provider to have the DCHP and option 150 configured on the router. The only limit on users is the bandwidth available.



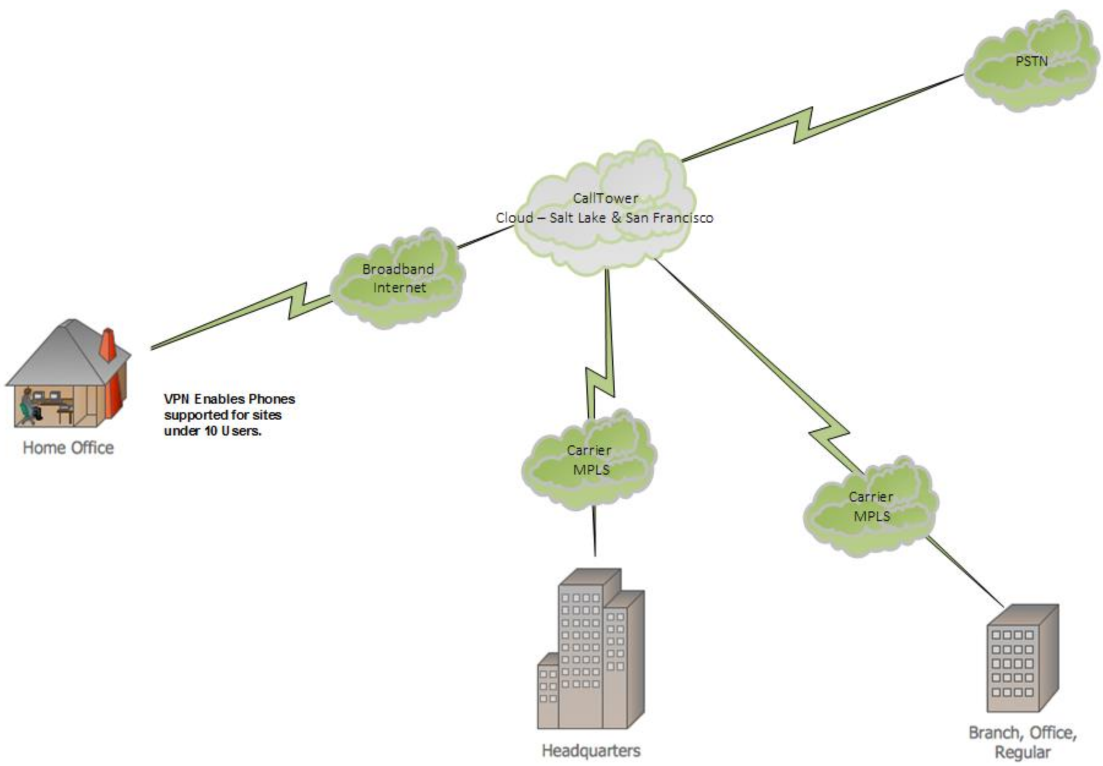
Connectivity Option #3

Customers with Internet circuits from alternate carriers that are not mentioned above can connect "over the top" or through their DIA, also referred as BYOB (Bring Your Own Bandwidth). CallTower provides and manages a gateway device (Cisco ASA) which creates a secure connection back to our data centers for connectivity. The ASA provides connectivity, option 150 and DHCP to the phones. CallTower manages the network address of the phones on your LAN. This configuration is supported for sites under 240 users. If there is a need to support more users and additional ASA can be added, which would require a secondary Voice VLAN for this device and its phones. QoS is not provided though a customer's carrier may provide a SLA on circuit.



Connectivity Option #4

For small offices with less than 10 phones we recommend VPN enabled phones or Edge enabled phones. These eliminate the need for additional equipment and does not require VLAN setup. Simple to use and setup. These phones connect over any carrier or circuit above 1.5MB. No additional equipment needed except for a phone power brick. The phone sits inside your LAN with your other data devices and does not require VLANs on your switch. All other solutions above require that you have a separate voice VLAN for the phones. QoS is not provided though a customer's carrier can provide a SLA on circuit. There are caveats that need to be discussed with these two solution, so please get with your solution engineer to understand them.



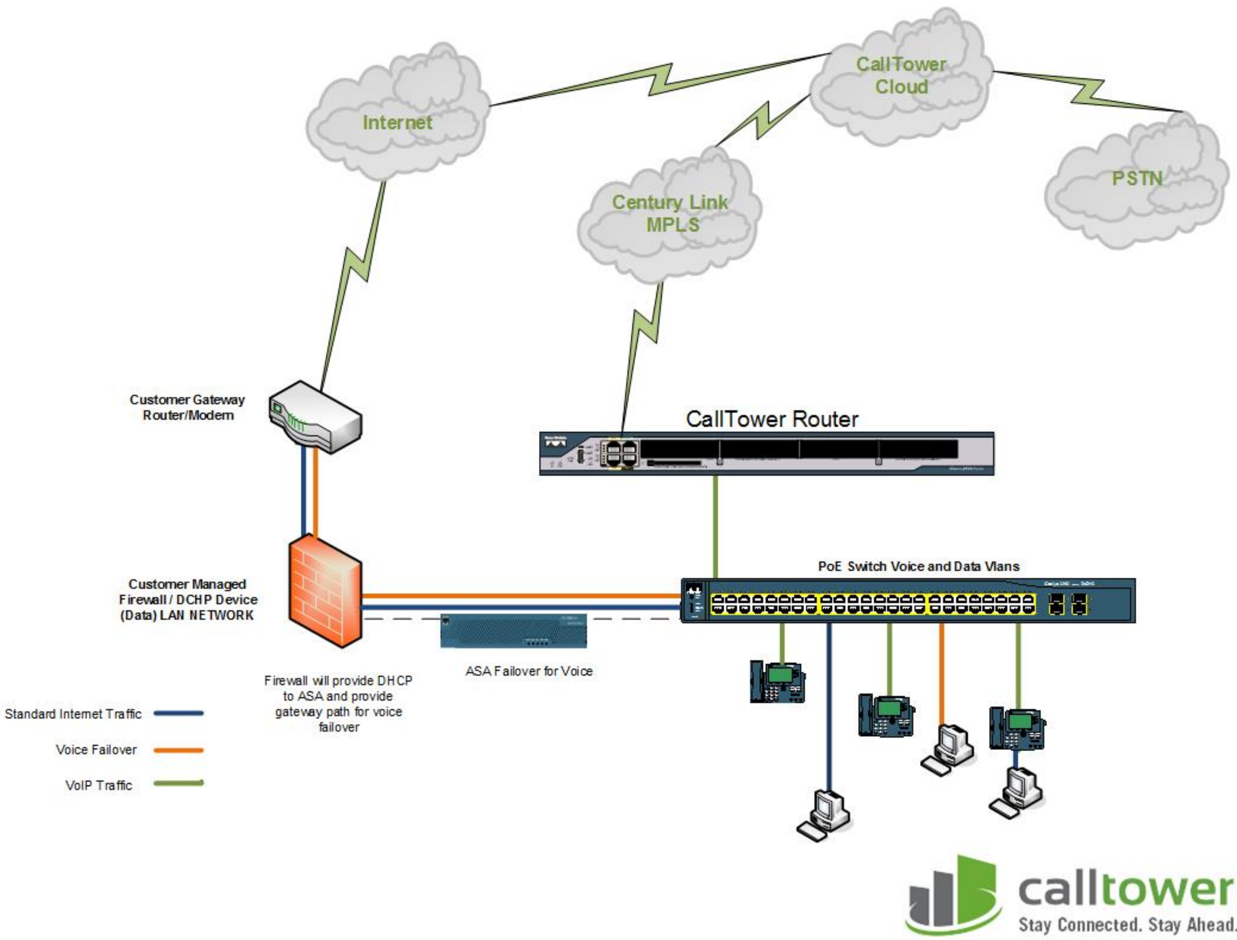
Redundancy

Redundancy is a critical success factor for a cloud based model. A successful VoIP solution requires a resilient and redundant network. When possible, the network should contain redundant hardware, redundant links, and adequate power backups. To meet the needs of a VoIP network, core network and VoIP hardware should run off UPS backups. All VoIP solution uses Power over Ethernet (PoE), the local network switches will also need power backups. These switches transmit power to the individual telephones, which will need power and connectivity during an electrical outage.

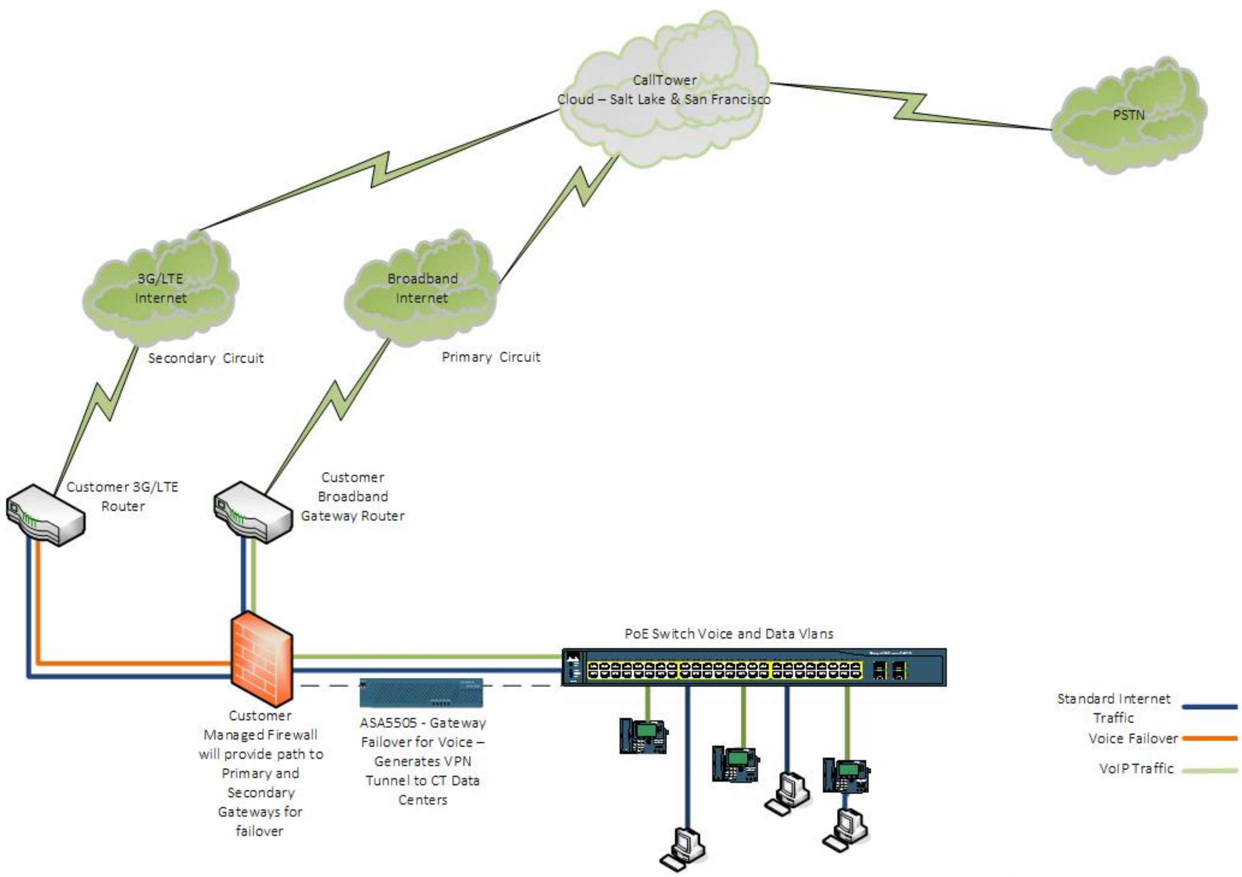
Circuit Redundancy

Implementing redundancy at the VoIP and network infrastructure levels is an integral part of VoIP communication, ensuring call and business continuity. Best practice for redundancy is using 2 different providers. Based on the connectivity scenarios provided above customers have several common redundancy options. Alternate failover designs can be approved by your CallTower Sales Engineer.

CT MPLS or CT VRF MPLS Failover to Broadband "BYOB"



Dual Carrier "Over the Top" or "BYOB" Failover



BYOB Circuit Assessment & Running a CallTower VoIP Test

During the presales & implementation phase of a VoIP project, information on the network is required in order to understand the potential VoIP environment. Reviewing the results from an assessment tool(s) can help decide if you need to update the voice and network infrastructure, create Quality of Service plans, and figure out the impact of VoIP on the IT organization. It is important to analyze your LAN network segments and WAN circuits to determine volume and type of traffic in order to assess the need for upgrades and to create a snapshot of current application performance.

Your phone system is the lifeline between your company, your customers, and your employees. That's why reliability and quality of service are essential parts of your communications network. There are certain aspects of your current network that need to be considered to ensure that you are capable of delivering high-quality voice, video, and UC features to your office. Below you will find tips and tools including a comprehensive IP SLA VoIP Network Test, to help you understand your network and determine whether or not it's fully ready to take advantage of a cloud hosted system. It is recommended that you run this test several times per location and at different times of the day. This will give you an overall snap shot of your network health.

http://www.calltower.com/resources/tools/voip-test/

How Do I Read the VoIP Test Results?

Here are the top 4 factors in determining your VoIP network readiness. A good result on all of these tests does not guarantee your network will be reliable, but it can quickly detect an inadequate network. These are the required ranges are below. If you receive a result outside these ranges, keep reading for potential tips on how to address them:

- *Latency (Maximum Round Trip Time) < 150ms*
- *Jitter < 50 ms*
- *Packet Loss < 0.5%*
- *MOS ≥ 4.0*

Upload and Download Speeds:

Bandwidth speeds can vary widely based on your LAN, your Internet connection (cable modem, DSL, T1, etc.) or your Internet providers network. Verify you are receiving the Internet service purchased from your provider. In order to carry voice traffic over your Internet connection it may require you to upgrade your connection or change providers in some circumstances.

Latency (Delay):

Delay occurs when there is a higher than normal latency on your network. Latency is the amount of time it takes between one person saying something and the other person hearing what was said. Latency doesn't typically cause audio quality issues, but when the latency is over about 150ms, the delay is noticeable to your users. If you have a latency issue, simple network tuning can solve the issue.

Packet Loss:

Packet loss is the failure of one or more transmitted packets to arrive at their destination. This event can cause noticeable effects in all types of digital communications. With audio calls, packet loss can provide jitter and silent gaps in communication.

Jitter:

Jitter occurs when voice packets arrive with varying delays. This is typically caused with changes in network traffic, and can usually be fixed by using QoS, or reducing the amount of traffic on your network equipment is handling. Your users will often report jitter as poor audio quality. If jitter is too high, you'll have to fix the network underneath to improve the call quality.

MOS Score:

In voice and video communication, quality usually dictates whether the experience is a good or bad one. Besides the qualitative description we hear, like 'quite good' or 'very bad', there is a numerical method of expressing voice and video quality. It is called Mean Opinion Score (MOS). MOS gives a numerical indication of the perceived quality of the media received after being transmitted. MOS is expressed in one number, from 1 to 5, 1 being the worst and 5 the best. MOS is quite subjective, as it is based figures that result from what is perceived by people during tests. However, there are software applications that measure MOS on networks, as we see above.

The score is quite easy to grade.

- 5 - Perfect. Like face-to-face conversation or radio reception.
- 4 - Fair. Imperfections can be perceived, but sound still clear. This is (supposedly) the range for cell phones.
- 3 - Annoying.
- 2 - Very annoying. Nearly impossible to communicate.
- 1 - Impossible to communicate

Filling out the CallTower SE Questionnaire

Your sales representative will provide you with access to the SE questionnaire. This questionnaire is required to help our solutions engineers better understand your current network environment and unique needs. Providing as much information as possible on you network, facilities, equipment and integrations are critical to the success of the project. Failing to disclose integrations, special equipment, or critical must have features may cause delay or put the project at risk. While filling out the questionnaire does not eliminate risk completely, it does help substantially in mitigating risk and/or delays during the project. If there is anything that you feel is not being asked or needs to be addressed in this questionnaire it should be noted in the questionnaire or directly discussed with the solutions engineer.

Cisco Jabber IM& Presence

Cisco Jabber Hardware Requirements for Desktop Jabber Clients

Requirement	Cisco Jabber for Windows	Cisco Jabber for Mac
Installed RAM	2-GB RAM on Microsoft Windows 7 and Windows 8	2-GB RAM
Free physical memory	128 MB	1 GB
Free disk space	256 MB	300 MB
CPU speed and type	AMD Mobile Sempron Processor 3600+ 2 GHz Intel Core 2 Duo Processor T7400 @ 2. 16 GHz	Intel Core 2 Duo or later processors in any of the following Apple hardware: • Mac Pro • MacBook Pro (including Retina Display model) • MacBook • MacBook Air • iMac • Mac Mini
GPU	DirectX11 on Microsoft Windows 7	N/A
I/O ports	USB 2.0 for USB camera and audio devices.	USB 2.0 for USB camera and audio devices

Operating Systems for Cisco Jabber for Windows

You can install Cisco Jabber for Windows on the following operating systems:

- Microsoft Windows 10 (desktop mode)
- Microsoft Windows 8.1 (desktop mode)
- Microsoft Windows 8 (desktop mode)
- Microsoft Windows 7

Operating Systems for Cisco Jabber for Mac

You can install Cisco Jabber for Mac on the following operating systems:

- Apple OS X Yosemite 10.10 (or later)
- Apple OS X Mavericks 10.9 (or later)
- Apple OS X Mountain Lion 10.8.1 (or later)

Hardware Requirements for Cisco Jabber for iPhone and iPad

The following Apple devices are supported for Cisco Jabber for iPhone and iPad on iOS 8 and later:

Apple Device	Generation
iPod touch	5
iPhone	4S, 5, 5c, 5s, 6, 6 Plus,
iPad	Second, third, fourth, and
iPad mini	Mini 1, mini 2, mini 3, and mini 4
iPad Air	Air1 and Air 2

Hardware Requirements for Cisco Jabber for Android

Note Cisco Jabber for Android is tested with the Android devices listed here. Although other Android devices are not officially supported, you might be able to use Cisco Jabber for Android on other Android devices.

The minimum CPU and display requirements for the Android devices are:

- Chipset—Android devices that are based on an Intel chipset are not supported.
- Android Operating System—4.1.2 or later.
- CPU—1.5 GHz dual-core, 1.2 GHz quad-core or higher (quad-core recommended).
- Display—For two-way video, the minimum display resolution requirement is 480 x 800 or higher.

Note

- Cisco Jabber for Android does not support the Tegra 2 chipset.
- Due to an Android kernel issue, Cisco Jabber cannot register to the Cisco Unified Communications Manager on some Android devices. If this problem occurs, see the Troubleshooting chapter of the *Cisco Jabber for Android User Guide*.

Cisco Jabber for Android supports IM only mode on the Android devices that meet the following minimum specifications:

- Chipset — Android devices that are based on an Intel chipset are not supported.
- Android OS — 4.1.2 or higher
- CPU — 1.5 GHz dual-core, 1.2 GHz quad-core or higher (quad-core recommended).
- Display — 320 x 480 or higher

Cisco Jabber for Android supports Audio and Video Enabled mode in the following devices with respective version of Operating System provided in the table:

Device	Device Model	Operating System
Cisco DX	70	10.2.x version
	80	10.2.x version
	650	10.2.x version
HTC	M7	Android OS 4.4.2 or later
	M8	Android OS 4.4.2 or later
	One Max	Android OS 4.4.2 or later
Google Nexus	5	Android OS 4.4 or later
	6	Android OS 5.0.2 or later
	7	Android OS 4.4 or later
	9	Android OS 5.0.2 or later
	10	Android OS 4.4 or later
LG	G2	Android OS 4.2.2 or later
	G3	Android OS 4.4.2 or later
Motorola	Moto G	Android OS 4.4.2 or later
	MC40	Android OS 4.1.1[1]
Samsung Galaxy	Note II	Android OS 4.2 or later
	Note III	Android OS 4.3 or later
	Note IV	Android OS 4.4.4 or later
	Note Edge	Android OS 4.4.4 or later
	Note Pro 12.2	Android OS 4.4.2 or later
	Rugby Pro	Android OS 4.2.2 or later
	SII	Android OS 4.1.2 or later

Device	Device Model	Operating System
	SIII	Android OS 4.2.2 or later
	S4	Android OS 4.2.2 or later
	S4 mini	Android OS 4.2.2 or later
	S5	Android OS 4.2.2 or later
	S5 mini	Android OS 4.2.2 or later
	Tab 3 8-inch	Android OS 4.4 or later
	S6	Android OS 5.0.2 or later
	S6 Edge	Android OS 5.0.2 or later
	Tab 4 7-inch, 8-inch, and 10.1-inch	Android OS 4.4.2 or later
	Tab PRO 8.4-inch and 10.1-inch	Android OS 4.4.2 or later
	Tab S 8.4-inch & 10.5-inch	Android OS 4.4.2 or later
	Note 10.1-inch 2014 Edition	Android OS 4.4.2 or later
Sony Xperia	ZR/A	Android OS 4.1.2 or later
	M2	Android OS 4.3 or later
	Z1	Android OS 4.2 or later
	Z2	Android OS 4.4.2 or later
	Z2 tablet	Android OS 4.4.2 or later
	Z3	Android OS 4.4.2 or later
	Z3 Tablet Compact	Android OS 4.4.4 or later
Huawei	G6	Android OS 4.2.2 or later
	Mate 7	Android OS 4.4 or later
Sonim	XP7	Android OS 4.4.4
Xiaomi	Mi 4	Android OS 4.4 or later

Ports and Protocols for Desktop Clients

The following table lists outbound ports and protocols that Cisco Jabber uses.

Port	Protocol	Description
443	TCP (Extensible Messaging and Presence Protocol [XMPP] and HTTPS)	XMPP traffic to the WebEx Messenger service. The client sends XMPP through this port in cloud-based deployments only. If port 443 is blocked, the client falls back to port 5222. **Note** Cisco Jabber can also use this port for: • HTTPS traffic to Cisco Unity Connection and Cisco WebEx Meetings Server. • Saving chats to the Microsoft Exchange server.
389	UDP/TCP	Lightweight Directory Access Protocol (LDAP) directory server.
2748	TCP	Computer Telephony Interface (CTI) used for desk phone control.
3268	TCP	Global Catalog server.
5070 to 6070	UDP	Binary Floor Control Protocol (BFCP) for video desktop sharing capabilities.
5222	TCP (XMPP)	XMPP traffic to Cisco Unified Presence or Cisco Unified Communications Manager IM and Presence Service.
8443	TCP (HTTPS)	Traffic to Cisco Unified Communications Manager and Cisco Unified Communications Manager IM and Presence Service.
7080	TCP (HTTPS)	Cisco Unity Connection for notifications of voice messages (new message, message update, and message deletion).
53	UDP/TCP	Domain Name System (DNS) traffic.
37200	SOCKS5 Bytestreams	Peer-to-peer file transfers. In on-premises deployments, the client also uses this port to send screen captures.
5060	UDP/TCP	Session Initiation Protocol (SIP) call signaling.
5061	TCP	Secure SIP call signaling.
49152 to 65535	TCP	IM-only screen share. The client randomly selects a port from the range.

Port	Protocol	Description
		The actual range may vary. To find the real range, enter the netsh interface ipv4 show dynamicportrange tcp command. You can use the SharePortRangeStart and SharePortRangeSize parameters to narrow the range used for IM screen share. For more information on these parameters, see the section on Common Policies parameters in the *Deployment and Installation Guide*.

Bandwidth Requirements for Cisco Jabber Desktop Clients

Cisco Jabber for Mac separates the bit rate for audio and then divides the remaining bandwidth equally between interactive video and presentation video. The following table provides information to help you understand what performance you should be able to achieve per bandwidth:

Bandwidth	Audio	Audio + Interactive video (Main video)
125 kbps under VPN	At bandwidth threshold for g.711. Sufficient bandwidth for g.729a and g.722.1.	Insufficient bandwidth for video.
384 kbps in an enterprise network	Sufficient bandwidth for any audio codec.	w288p (512 x 288) at 30 fps
1000 kbps / 1MB	Sufficient bandwidth for any audio codec.	w576p (1024 x 576) at 30 fps
2000 kbps / 2MB	Sufficient bandwidth for any audio codec.	w720p30 (1280 x 720) at 30 fps

Understanding the Porting Process and Timeline

https://www.uc.solutions/CallTower_Internal/Implementation/Porting_Timeline_and_Process