VoIP / SIP Planning and Disclosure

Voice over internet protocol (VoIP) and session initiation protocol (SIP) technologies are the telecommunication industry's leading commodity due to its cost savings and premium feature capabilities. VoIP technology is the transmission of (traditionally circuit-switched) person-to-person voice conversations over IP based networks. Similarly, SIP trunking technology is a quality version of VoIP providing the transmission of voice paths across the internet for business customers. This is the same technology which Verizon, Google, and AT&T employ in their main CO locations, while now bringing it straight to your business. Such services include basic telephony services and unified communications services, provided to the customer by an IP branch exchange (IP PBX).

Preparing for voice and video over IP requires understanding current network traffic, determining network policies for traffic, and applying quality of service (QoS) controls over the wide-area network (WAN).

For IP networks supporting voice, video, and data applications, the service objective is to preserve both the mission-critical data in the presence of voice and video, and to preserve the voice and video quality in the presence of burstable and / or peak data traffic. Accordingly, certain times during the day may result in higher than normal levels of Internet data traffic, therefore possibly impairing call quality.

Organizations wishing to maintain management control of their networks when adding new applications, including voice over IP (VoIP) and SIP Trunking, usually plan for and follow some variation of a process, including: **1) perform an assessment of the current network**, **2) determine the bandwidth and performance characteristics required for the new applications**, **3) implement control procedures**, and **4) monitor and report network behavior**.

This study will gather information about the condition of the network hardware, configuration and congestion, determine your network readiness, and suggest needed changes and recommended network improvements.

If the requirements stated in the Network Assessment document(s) are not met before the implementation of VoIP on your network, telephone calls across the network may experience poor voice quality such as static or latency, as well as call connection and disconnection problems.

Minimum Network Requirements of VoIP / SIP Implementation:

- Routers and Switches in the Local Area Network must support 100Mbps transmission.
- The LAN must employ Layer 2 Data Switches that support Class of Service (CoS), especially if IP phones and devices are deployed at the location.
- Routers connected to the WAN must support prioritization based on Quality of Service (QoS) tags.
- The LAN portion of the network should be a “managed network”. (The Internet is not considered a managed network).

Please be prepared to assign responsibility for bringing the various aspects of the network into compliance to properly support VoIP / SIP, as well as for monitoring the different parts of your network in post installation. It is important that these responsibilities be assigned prior to implementing VoIP / SIP technology; if problems arise it will be clear who is responsible for solving them.

Please realize that network environments change constantly and the capacity requirements of the business applications running throughout them are in flux. As a result, administrators must monitor and report on network performance to ensure that bandwidth is utilized appropriately.

The transmission requirements and network parameters given in the following requirements and recommendations are for the network after VoIP has been implemented.

- End-to-End packet delay must be 150 milliseconds or less during peak loads.
- Packet delay should be 80 milliseconds or less for toll quality voice.
- For good voice quality, end-to-end packet jitter should be 20 milliseconds or less.
- End-to-end packet loss must be 2% or less for acceptable voice quality.
- Packet loss should be 1% or less for good voice quality.
- The Bandwidth Utilization on the WAN must be 70% or less.

Telecommunications Relay Service (TRS)

Telecommunications Relay Service (TRS) is a telephone service that allows persons with hearing and speech disabilities to place and receive telephone calls through the assistance of a Communication Assistant (CA). The CA will place the call and type the spoken words for the text telephone users to read. In addition, the CA will also read messages the teletypewriter (TTY) user sends back. All conversations are confidential without censorship. There is no charge to TRS users for this service; however standard phone charges do apply. TRS services allow for 24 hours, seven days a week calling without time limits.

Though VoIP solutions do not directly support TTY service, a cloud extension would be optimized for repeated 711 / state specific toll free relay calling. Should a caller need access to TRS / TTY service, ACC Telecom recommends visiting http://www.access-able.com/relay.html to process their call. This link provides access to US States' TTY, Voice, ASCII, and Spanish direct dial toll free relay numbers.

911 / E911 Disclosure

The rules of the Federal Communications Commission (FCC) require ACC Telecom, like all Voice over Internet Protocol (VoIP) service providers, to inform its customers of any differences between the 911 and E911 access capabilities available with VoIP / SIP service as compared to 911 and E911 access capabilities available with traditional wire line telephone service.

By signing the 911 / E911 acknowledgement, you are affirmatively acknowledging that you understand that you may not be able to contact emergency services by dialing 911 in the event of power failure or disruption, and 911 / E911 will not operate if your broadband connection is disrupted.

You must provide ACC Telecom with your correct service address or calls using your IP device to 911 / E911 may be routed to emergency personnel who will not be able to assist you. Dialing 911 using a VoIP / SIP service, your call is routed to your local emergency operator designated for the address that you listed at time of activation (your Requested Address).

VoIP / SIP 911 / E911 calls may not complete or may be routed to emergency personnel who will not be able to assist you if you disable, damage, or move the equipment to a location other than the service address you provided when service was initiated. You acknowledge and understand that 911 / E911 does not function if you move your device to a different street address, unless you have notified ACC Telecom of any such change in your Registered Address. You also acknowledge that it may take up to 72 hours for any change in address to be processed. Accordingly, you should notify ACC Telecom in advance of any and all changes to your Registered Address. Failure to provide the current and correct physical address and location of your device may result in any 911 call you make being routed to the incorrect local emergency service provider and emergency personnel being dispatched to the incorrect location.

Emergency personnel may not be able to identify your phone number in order to call you back if the call cannot be completed, is dropped or disconnected, or if you are physically unable to tell them your phone number, and / or your IP service is not operational for any reason other than suspension of service.

Emergency personnel may not be able to identify your address if you use your equipment at an address other than your registered address. You acknowledge and understand that Public Safety Answering Point (PSAP) and emergency personnel will not be able to find your location if the call cannot be completed, is dropped or disconnected, if you are physically unable to tell them your location, or if the service is not operational for any reason other than suspension of service.

Please note that VoIP / SIP E911 / 911 service calls may be delayed or dropped due to network architecture. You understand and acknowledge that due to technical constraints, there is a greater possibility of network congestion and / or reduced speed in the routing of a 911 call made utilizing your IP equipment as compared to traditional 911 dialing over traditional telephone networks. You acknowledge and understand that there is a greater possibility that the general telephone number for the local service provider will produce a busy signal or will take longer to answer, as compared to those 911 calls routed to the 911 dispatcher(s) who are specifically designated to receive incoming 911 calls using traditional 911 dialing. For more information on E911 /911 IP services please see the FCC order: www.fcc.gov/cgb/voip911order.pdf.

I understand that if the minimum network requirements stated in this document are not met and maintained, VoIP / SIP telephone communications across the network may be impaired. These requirements are generally the same for most all IP vendors and most types of IP telephone equipment.

I acknowledge and understand the impact of moving or disabling my IP device when contacting emergency personnel and am fully aware of notifying ACC Telecom with any changes to my Registered Address.

I recognize that ACC Telecom cannot be held responsible or liable for the condition of voice signals transmitted across an IP network over which they have no control, and will not be liable for problems resulting from poor voice quality if recommendations above are not met.

Your signature on your ACC Telecom Sales Agreement indicates that you have received, read, and fully understand the contents of this document.

Optional: **Contact Information for Network Service Company if outsourced**

Company Name: ____

Company Contact: ____

Service Company: ____

Address: ____

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City: ____________________________________ State: _______ Zip Code:_________________

Telephone: _______________________________ Fax: ___________________________________

Contact Telephone Number: ___________________________________ Extension: ___________

Email: ____