Voice over Internet Protocol (VoIP) Design Considerations

VoIP Bandwidth - Design Considerations:

When choosing your VoIP implementation the three main protocols groups are:
• **SIP**, **H.323**, **SCCP**, **IAX** or **MGCP**
• The Real-Time Transport Protocol (**RTP**)
• Skype **SILK** speech CODEC

Factors affecting the Voice quality of a VoIP call:

• Packet Loss (needs to be less than 1% loss)
• Delay
1. Ideal 150ms domestic
2. 200ms internationally
3. 400ms via satellite for the one way delay

• Jitter <10ms
• Bandwidth & **LFI** required on slow data/voice links (Calculations example in Table 3)

Most VoIP vendors support one or more of the following ITU standards CODEC:

• **G.711** often the default standard for IP PBX, IP phones and ATA vendors, as well as for the digital PSTN and trunks. This standard digitizes voice into 64 Kbps before encapsulation with no compression
• **G.729** is supported by many vendors for compressed voice operating at 8 Kbps, 8 to 1 compression. With quality just below that of G.711, although hard to detect the difference. There is a licensing fee associated with every G.729 active call stream in use. Ideal CODEC for WAN and low bandwidth circuits. To be avoided on links going through tandem encoding such as to GSM networks.
• **G.723.1** was once the recommended compression standard. It operates at 6.3 Kbps and 5.3 Kbps. Although this standard further reduces bandwidth consumption, voice is noticeably poorer than with G.729, so it is not very popular for VoIP.
• **G.722** operates at 64 Kbps, offering high-fidelity speech. Whereas the three previously described "G" standards deliver an analogue sound range of 3.4 kHz, G.722 delivers 7 kHz. High Definition Voice is the designation for G.722 SIP (patents have expired, so it is freely available).

VoIP Voice Quality:

The quality of a voice call is defined by the Mean Opinion Score (**MOS**) 0 to 5 (see Table 1). A score of 4.4 to 4.5 is considered to be toll quality. Voice compression affects the MOS as the sampling rate drops so does the MOS. An MOS below 3.5 will usually produce complaints from the users. Cell phone calls average about 3.8 to 4.0 for the MOS. In Table 1 below the most common VoIP protocols are summarized:

VoIP - Speed / Mean Opinion Score / Delay			**Table1**
Standard	Speed	MOS	Sampling Delay
G.711	64Kbps / 8000 Bytes per second	4.4	0.75ms
G.729	8Kbps / 1000 Bytes per second	4.2	10ms
G.723.1	6.3/5.3Kbps	4.0/3.5	30ms
G.722	64Kbps / 8000 Bytes per second	4.0	40ms

Bandwidth Consumption & Fax DTMF stability over VoIP:

Calculating bandwidth consumption for VoIP depends on several factors: the compression technology CODEC, packet overhead, network protocol used and whether silence suppression chosen (see Table 1). Also several factors affect the bandwidth, delay, jitter and losses within a network: bandwidth available, queuing strategy, QoS, reliability, ALG and resilience. All or some of these are controlled by the firewall **WAN/LAN** interface. Regardless of which broadband option chosen the Network Address Translation (**NAT**) needs to be limited to one translation from "A" class to class "B" IP range. This requires the modem to be placed in the **bridge mode** allowing the Class "A" address to reside on the firewall.

Not all of these factors are under the control of the end user. Tradeoffs and compromise are required between cost and broadband bandwidth. It is not always possible or cost effective to just add bandwidth to implement VoIP; QoS is not always available on the lowest cost broadband circuits DSL, ADSL, Cable Modem, and Wireless Point to Point. The enterprise will often have to accept that during times of busy voice activity the data network response will be slightly slower, if there are not enough funds to provide the luxury of ample bandwidth.

For example The choice of CODEC G.711/G.722@ 64kbps (8000 Bytes per sec) or G.729 @ 8kbps (1000 Bytes per sec) offers MOS of 4.1 or 3.94 which most people cannot differentiate, but G.729 will suffer if tandem encoding has to be used to reach a GSM network and drops the MOS to 3.15. In noisy environments a higher bit rate is also desirable as the noise levels are higher with the more complex CODECs (see Table 2):

• Choose a higher bit rate CODEC for noisy environments
• Do not use low bit rate CODECS for tandem encoding routes
• Use single CODEC frame per packet when using low bit rate CODECs on poor networks with suboptimal QoS
• If you plan to use a SIP CODEC for FAX remember the DTMF is sometimes distorted when compared to a POTS (Plain Old Telephone Service) and will require a ATA and the FAX limited to 9600 Baud rate if the older **T.4** standard is used on your fax rather than the newer **T.38** SIP standard (see Table 2).

DTMF Keypad frequencies				**Table 2**
	1208 Hz	**1336 Hz**	**1477 Hz**	**1633 Hz**
697 Hz	**1**	**2**	**3**	**A**
770 Hz	**4**	**5**	**6**	**B**
852 Hz	**7**	**8**	**9**	**C**
941 Hz	*****	**0**	**#**	**D**

DTMF Restrictions during SIP/VoIP transmission
Voice from a PSTN is compressed by VIP before sending across the IP network and then decompressed at the destination VIP. The VIP is designed for voice and not DTMF which produces distortion

Broadband Choices and Configurations:

T1: Use leased line circuit of 1.544 Mbps used to carry 24 x 64kbps channels. In some areas this is the only option and generally runs over $300/month (see conditions for DSL below).

DSL: Standard provides up to 6Mbps downstream and 768Kbps upstream within 1.8/9504 Miles/Ft of the DSLAM. The existence of a bridge tap or other line abnormality will reduce these sects as will the distance from the **DSLAM**. Call your Telco and have them preform line conditioning on all DSL & ADSL circuits (if the line defects are outside the service call is free). Remember to install a UPS at all points of failure to insure phone service in brownout or blackout proof (ie. Modem, firewall, switches and handsets).

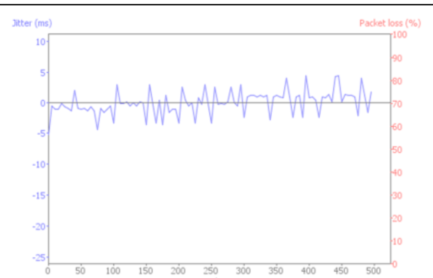
ADSL2: Standard provides up to 12Mbps downstream and 1Mbps upstream within 1.8/9504 Miles/Ft of the **DSLAM**. When the subscriber is at the distance limit Interleaving (**LFI**) is an option which counters the effects of burst noise on the telephone line. An interleaved ADSL line has a depth, usually 8 to 64, which describes how many Reed–Solomon codewords are accumulated before they are sent. As they can all is sent together, their forward error correction codes can be made more resilient. Interleaving adds latency as all the packets have to first be gathered (or replaced by empty packets) and they, of course, all take time to transmit. 8 frame interleaving adds 5ms round-trip-time, while 64 deep interleaving adds 25ms. Other possible depths are 16 and 32. Caution must be taken in mounting the DSL/ADSL Modem at the MPOE, maximize the twists from the punch down with the shortest distance to the modem. Utilize a CAT 5 Ethernet wire to connect to the Firewall. This will reduce RFI and improve line stability (see conditions for DSL above).

ADSL2+: Improvement which provide up to 24Mbps downstream and 1Mbps upstream within 1.8/9504 Miles/Ft of the DSLAM (see conditions for DSL & ADSL above).

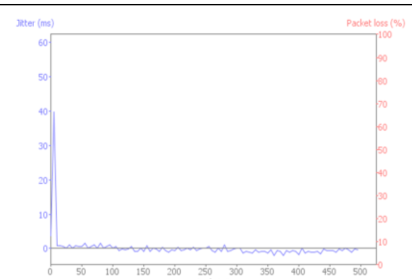
Cable Modem: When a cable company offers Internet access over the cable, Internet information can use the same cables because the cable modem system puts downstream data sent from the Internet to an individual computer into a 6-MHz channel. On the cable, the data looks just like a TV channel. So Internet downstream data takes up the same amount of cable space as any single channel of programming. Upstream data information sent from an individual back to the Internet requires even less of the cable's bandwidth, just 2 MHz, since the assumption is that most people download far more information than they upload. Putting both upstream and downstream data on the cable television system requires two types of equipment: a cable modem on the customer end and a cable modem termination system (**CMTS**) at the cable provider's end. Between these two types of equipment, all the computer networking, security and management of Internet access over cable television is put into place. One of the advantages of cable is the low latency and low jitter (excellent isolation from RFI). Speeds vary depending on the CMTS capacity and configurations in your area; expect 10Mbps/4Mbps or more. Don't be afraid to call your cable provider if the speed and latency isn't up to spec, they too may need to filter and do other line conditioning procedures as Telco's do (see conditions for DSL & ADSL above).

Firewall and Broadband Bandwidth management: Low end Firewalls offer inadequate QoS or ALG implementations. QoS is a requirement for any bottle-neck within the network. The bottle-necks are usually found at the WAN/Firewall interface. Remember it is important to preserve the VoIP requirements of little or no packet loss, <200ms delay, <10ms jitter and enough bandwidth for the call (see Table 3). That will not be achieved if you WAN is fed with a Firewall that has no QoS and allows an **FTP** transfer to send several 1500 Byte packets onto the WAN during a VoIP call causing the VoIP audio stream to wait more than 10ms before it is sent, the result would be broken audio. Therefore it is import to configure the switches and Firewalls inside and at the WAN/LAN interface to mark and classify the packets to enable the queue and prioritize the VoIP packets and ensure minimum delays to the voice packets. The Cisco ASA 5505 is an ideal interface at the WAN/Firewall boundary. It comes with a SIP compliant ALG and QoS implementation. It will require some testing to customize it for your particular CODEC. Newer firewalls may work it is all trial and error.

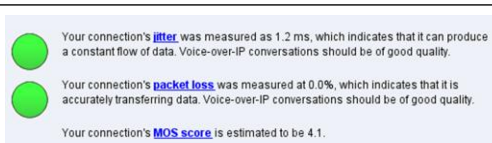
VoIP: Jitter, Packet Loss, MOS, Bandwidth Calculation & Seed test Table3



Upstream Jitter and Packet Loss



Downstream Jitter and Packet Loss



Jitter at 1.2ms, Packet loss at 0.0% and MOS Score of 4.1
http://myspeed.visualware.com G.711/G.722 64Kbps

Bandwidth with ADSL with interleave 2.5Mbps down 800Kbps Up In a 5 SIP phone G.722 office. Plan on 320Kbps up @40% at the up bottle neck. Remember if a QoS is not implemented a large FTP upload will take out the VoIP circuit. The down load is ness problematic with only 12.5% of the bandwidth consumed by VoIP. For speed testing go to http://speedtest.comcast.net

Glossary:

VoIP (Voice over Internet Protocol)
Skinny Call Control Protocol (SCCP)
Inter-Asterisk eXchange (IAX)
Media Gateway Control Protocol (MGCP)
Voice CODEC (COder/DECoder)
SIP (Session Initiation Protocol)
H.323 Is an ITU VoIP protocol. It was created at about the same time as SIP, but was more widely adopted and deployed earlier.
SCCP (Skinny Call Control Protocol)
IAX (Inter-Asterisk eXchange protocol)
MGCP (Media Gateway Control Protocol)
RTP (Real-Time Transport Protocol)
LFI (link fragmentation and interleave)
ITU (International Telecommunication Union)
Public Switched Telephone Network (PSTN)
MOS (Mean Opinion Score)
Quality of Service (QoS)
Application-level gateway (ALG)
Network Address Translation (NAT)
Internet protocol address (IP)
Analog telephony adapter (ATA)
Dual-Tone Multi Frequency (DTMF)
Versatile Interface Processor (VIP)
Minimum Point of Entry (MPOE)
Digital Subscriber Line Access Multiplexer (DSLAM)

References:

http://Voip-Info.org
http://www.kccommunications.com/