VOIP Network Readiness

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Introduction.

VOIP is not a new technology. It exists in various formats for close to twenty years. However, its economic value and acceptance as a mainstream technology increased greatly during the last 4-5 years due to:

- Ability to achieve proven cost saving
- Unified infrastructure for voice and data
- New converged applications and services opportunities

In the USA, IP based phone installation is expected to reach parity with the classical analog and digital phones by the middle of that decade [*Reference A1*].
We at Connegy see this trend in our daily sales report where we identify a rapid shift in the mix of product sold – from TDM centric solutions into the UNITe-IP family of products.

However, the most problematic aspects in deploying VoIP-based solutions are the proper assessment of the underlying data infrastructure that will carry the voice packets from source to destination.

This article, and the TVSE site-survey form attached to it, are meant to provide sales and presales engineers with tools to help you assess, plan and build VOIP installations that can support high quality voice over the LAN and WAN.

Data Networks issues.

PlanNet, a market research group, "*Implementing IP-PBX systems: Lessons from the Trenches" [Reference 2]* estimates that 30-40 % of data networks need upgrading before adding VOIP. A 2003 assessment by GARTNER group specifies that 85 % of the existing networks are not ready for IP telephony.

How can that be? How come you have a perfectly working network; you get your e-mail in time, save/backup your data with no complaints, can access the Internet so smoothly and still cannot have VOIP right away?

Consider the following questions for a data network:

- Does it matter if the network is down for up to 20 minutes during a whole year?
- Does it matter if you receive e-mail with a two seconds delay?
- Will you cry if a big file transfer took 8 minutes and sometimes 10 minutes?
- Are you even aware that the same packet is re-transmitted over and over again until it reached its destination correctly?
- Would anybody care if some packets arrive out of order to the destination port but put in order by the application utilizing the packet?

For most users and most existing data applications the answer is NO!
However, in the classical telephony networks a 99.999 % availability, which is common in today's PSTN world, translates to less than 6 minutes downtime in a year! Reference A3 puts that number to 3 to 4 nines (99.98 %) for public telephony in USA.

When it comes to VOIP communications, none of the five items above are acceptable situations. For good quality voice a delay of more than 150 milliseconds in one direction is not acceptable data loss of more than 3% is catastrophic and retransmission or re-ordering of packets immediately affects the clarity of the conversation.

We can consider VOIP as a litmus paper for network performance and quality. You can see VOIP as something that makes network deficiencies visible to untrained eyes.

The following VOIP parameters affect the voice quality directly:

- Delay
- Jitter
- Packet loss
- Packet disorder
- CODECs used
- Bandwidth required versus bandwidth available

Delay
This is the network delay in one direction.

Delay from source to destination is not always the same as the delay from source to destination. Connegy VOIP installation requires delay less than 150 milliseconds.

Jitter
This is a measure of the change in delay in the network. Connegy VOIP requires this parameter to be less than 80 milliseconds.

Packet loss
Some packets originating at the source might sometimes not arrive to their destination. Connegy VOIP installations require less than 1 % random packet loss.

Bandwidth
The CODEC used for VOIP telephony imposes a minimal bandwidth requirement. Under various packet technologies headers are also added which increases the required bandwidth. Please refer to Reference A7 for a detailed Bandwidth analysis.

The following issues affect the VOIP quality parameters:

- QOS/COS policy
- Network topology and configuration
- WAN protocols used

The basic quality parameters for Connegy VOIP installations are supposed to be measured for all IP-Phone to IP-phone and IP-Phone to main site.

Issues affecting Voice quality: A theoretical background

Some parameters affect the VOIP quality directly. This section deals with those issues. Other network deficiencies indirectly affect those parameters. For example a heavy CPU load in a router might cause network delays, which in turn affect the voice quality.

Delay

Delay is the time it takes for one packet to get from source to destination. ITU-T G.114 (The ITU-T recommendation for end-to-end one-way delay) puts the limit to 300 milliseconds round trip time: 150 milliseconds one-way delay.
The contributing factors to delay are:

- *Packetization delay*
  This is the time it takes to convert speech from its original analog format to packetized digital data. Among other things it is dependent on the CODEC used and the size of the voice sample.

- *Propagation delay*
  This is the time it takes information to travel from source to destination. Inside a LAN this time is usually insignificant. When crossing the Internet or a satellite link that becomes important. In some networks VOIP might get from one point to another through a geostationary satellite. If we assume that signal is traveling approximately at 200,000 Kilometers/seconds than each 1000 kilometer adds 1/200 seconds = 5 milliseconds. If we assume a Geostationary Communication satellite at 35,000 kilometers high the delay becomes at least 350 milliseconds.

- *Queuing Delay*
  When packets are sent through a switch or router it must compete with other packets for right of way. If the router/switch is congested or the CPU is busy than some packets must wait. Queuing delay is the interval between the moment that the first bit of a packet appears at the port of entry until that first bit appears at the port of exit. For that reason it is important that switches/routers in the path of VOIP packets handle the packet priority correctly.

Jitter

Jitter is a measure for the variation of packet delay VOIP packets' path from source to destination is decided by the network entities independently from one packet to another. When packets cross network entities, switches, router etc. from source to destination the delay is generally not the same through different network entities. In a specific network entity, for example router, the queue length and CPU load changes with time. As a result packets crossing the same switch at different times might suffer different delays.

Excessive jitter is disturbing to human ear. By buffering the packets and delaying their delivery this problem can be resolved. The buffer collects the packets that arrive at different delays at the entry point and deliver them with a longer but stable delay. This special buffer is called Jitter Buffer. If jitter buffer size is increased we must make sure that the delay is inside acceptable limits.

Packet loss

Some data crossing a network gets lost and does not arrive to the destination for the following reasons:

- Erroneous packets are dropped.
  In VOIP no retransmission is permitted and erroneous packets are dropped and considered as packet loss.
- Errors can occur at the cable plant level.
  A common cause of such error is caused when low quality cables are utilized, or the cable length is longer than the standard permits. The maximum distance permitted in a 100 MBPS CAT-5 cable is 100 meters. A large deviation from that distance also contributes to packet errors.
- Sometimes packets are deliberately dropped by switches/routers along the path. There are various reasons for that including:
  - A full buffer at the entrance to the switch/router discards the newly arrived packet.
  - Packets with higher priority than the VOIP packets hog the switch/router and causes lower priority packets to be dropped.
  - The CIR (Committed Interface Rate) in a Frame Relay network is lower than the bandwidth required for the voice traffic together with other data traffic of the network. As a result packets that cross this limit are marked to be dropped first if the Frame Relay network gets crowded.

Some CODECs can compensate for packet loss if the loss is randomly distributed and not excessive.

Packet disorder

In VOIP the voice packets utilize the RTP over UDP. Each packet travels independently from one another and can take different paths with possible different delays. As a result a packet with a longer delay can arrive later than its predecessor. As long as the number of such packets and the time distance between them is low buffers can overcome this problem.

If an out of order packet cannot be recovered in the buffer before it is emptied (generally before that) than it is considered as a lost packet by the application.

Bandwidth

This is the amount of bits/second (or KBPS if you like) required for VOIP packets. A detailed BW analysis can be found in Reference A7.Figure 5 in that document puts the BW required in various situations as follows:

- G.711- Between 82.4 KBS and 127 KBPS
- G.729- Between 20.3 KBPS and 38.2 KBPS

QOS/COS and QOS Policy

Every application has its Quality of Service requirements. VOIP also has its own requirements that are radically different and more stringent than regular IP applications. In previous sections we covered the delay, jitter, packet loss and packet disorder parameters and their significance in telephony.

A successful VOIP installation expects end-to-end compliance to QOS parameter requirements. On the one hand VOIP equipment has to make its claim for a preferential treatment inside the network and on the other hand the network, LAN and WAN, must correctly interpret those requests and provide the QOS.

In a nutshell there are three important aspects of QOS

- Marking packets or frames for treatment in line with the required QOS
- Admission control and policy enforcement
- Correctly interpreting the marked frame/packet treatment in the network Frame is meant as OSI model layer 2 data units (MAC layer) and by packet OSI model layer 3 (mostly IP) info unit.

A spin off of the VLAN standard IEEE 802.1Q, the IEEE 802.1P is a good example of a standard for marking frames. In this standard three bits in the frame provide a marking of priority from 0 to 7.
Differentiate Services (*Diffserv*) is another marking method at layer 3(IP level) packets. It is defined in detail in *RFC2475* standard. Diffserv can mark IP packets with priority between 0-63.

A similar but simpler method that operates at layer 3 utilizes the TOS (Type of Service) bits of IP version 4 and its equivalent, Class of Service in IP version 6.
TOS utilizes 3 bits in the IP frame. Connegy VOIP products utilize TOS and by default assign the value of 5. This value can be configured but it is strongly advised not to do that without consulting the IT manager of the network.
Not all network equipment supports all TOS levels. A higher value might cause such network entities/routers to ignore the TOS completely! As a matter of fact we advise checking if all the network entities support TOS and if yes to ask the IT manager to activate it.

Another QOS approach in layer 3 would be a per flow approach. . A flow is characterized with a set of 5 numbers: Source/Destination IPs, Source/Destination port numbers and the protocol number (UDP, TCP). Integrated Services (*IntServ*). IntServ is a per flow based QOS method.

It is also possible to reserve resources for QOS in advance of the voice traffic.
RSVP is one such protocol. RSVP tries to reserve resources from source to destination before RTP packets are forwarded into the path. RSVP does not scale well and is not much helpful when crossing the Internet.

CODECS (Coders/Decoders)

CODECs are devices/chips that convert analog signal to digital form and vice versa. There are many standards. Each standard has its own bandwidth requirements and provide different levels of QOS. While higher bandwidth preserves the authenticity of the source better than compressed voice the characteristics and quality of the network carrying the voice might give results that might necessitate fine-tuning.
Connegy supports the G.711 and G.729 CODECs.

Network Topology and Configuration

Hubs/Switches

It is strongly recommended that the VOIP network utilize end-to end Switches and not Hubs. Hubs create collisions and can severely affect voice quality.
In addition the utilization of a HUB can go down as low as 30 %. That is a 10 MBPS HUB can end up providing only 3 MBPS throughput for the VOIP channels which cross it.

VLAN (Virtual LAN)

VLAN is a method to limit the broadcast domain of a LAN at layer 2. Ports in switches that are configured to a specific VLAN receive broadcast frames if those frames are tagged to the same VLAN as the port.
Using different IP subnets for VOIP equipment and other non-VOIP equipment does not protect from the other from layer 2 broadcasts, but VLAN does!
The VLAN standard, IEEE 802.1Q tags each Layer 2 frame (not the IP) frame with a VLAN ID. Switch ports can be configured to accept only packets from a specific VLAN ID, from a group of VLANs or all. Not all available switches support VLAN and have some minor differences in the implementation.
Whenever possible put all the VOIP equipment in a separate VLAN. This configuration protects the IP phones and other VOIP equipment from unwanted broadcasts from PCs and other network equipment coming from other VLANs.

Port configuration

The port speed and mode of operation (Full duplex/half duplex) can affect the performance and quality of Voice. The Connegy VOIP products support Auto Negotiation in the Ethernet. The best performance can be received for FDX 100 MBPS. If the switch port does not fully support Auto negotiation, a rarity nowadays, then the voice quality and the performance fall. If the VOIP equipment expects 100 MBPS FDX while the switch port stays at 100 HDX than big trouble! In that case the situation will also be seen in data transfer. Such situations cannot be diagnosed with the help of software based Network Analyzer but with the help of specialized Hardware based Network analyzer that can cut through the switch port and the VOIP equipment.

Spanning Tree (IEEE 802.1D)

It is a layer 2 (Link layer) protocol to provide loop-free redundant switching environment. Its operation is automatic and the network stabilizes after a known period of time. A change in the link status, down to up or up to down cause it to activate. Until a stable loop free topology is reached the ports do not receive/transmit payload data.

There is another per-VLAN Spanning tree which can have multiple active roots from any source to any destination; one per VLAN.
A newer modified and faster converging spanning tree protocol based on the same principle, a single path from any source to any destination at any given time is called RSTP (IEEE 802.1W).
802.1W can interoperate with the older 802.1D but than the Fast Spanning Tree capable ports also use 802.1D.
In a network with Spanning Tree activated the ports that have VOIP equipment connected, IP phones or TVSE/Vocoders we advise disabling Spanning Tree or configuring those ports to Fast Spanning Tree.
Do not disable Spanning Tree/Fast Spanning tree for the whole switch!

VPN (Virtual Private Network)

It is a method to transfer data between secured locations over unsecured networks; Internet, Frame relay, Leased lines, etc.
It provides security similar to that of a Leased Line with less cost.
VPN creates tunnels between LANs, and/or PCs/IP phones, across unsecured networks.
When utilizing VPN for VOIP we must take into account the delay introduced by those devices.

Network reliability and Resiliency schemes

Network reliability can be characterized as

- Availability- % of Uptime
- Mean Time to Repair and Mean Time To Repair (MTBF/MTTR)
- Packet loss
- Errors

The user can have control on the quality of the LAN but have limited influence on the Internet Service Provider (ISP). This influence must be asserted with Service Level Agreement with ISPs (Internet Service Providers)
One of the factors affecting network reliability is of course the cable plant. A CAT5 cable plant or better is advised for 100 MBPS Ethernet.
Spanning Tree, Link Trunking, and link redundancy is a short list for resiliency methods at Link Layer.
At the networking layer router redundancy schemes such VRRP between routers is helpful in increasing the network availability.
Each routing hop adds some delay as well. When checking the network configuration we advice to do the VOIP routing at the edge and not at the backbone routers which might have a heavy load which can be seen as additional delay.

Network Investigation tools

During network assessment phase for VOIP readiness it is advised that some QOS parameters, such as delay, jitter, packet loss etc. be measured for the planned VOIP installation.

Appendix B gives a short list of tools. A network Investigation tool that will be able to measure RTT, jitter and packet loss is being developed in Connegy. Some of the tools listed in the appendix are freeware and some are quite expensive devices.

Preparing a successful VOIP Installation

We can talk for many hours on how to prepare and install a VOIP system. I will try to make it short. Anyone interested in more in depth coverage can refer to the many presentations listed as references.
Once the main VOIP components are up and running adding new IP phones, changing extension properties are almost plug and play.

One of the first perquisites for a successful VOIP project is to get people from different specialization around a table. IT people are necessary as are telephony people from the end-user and system Integrator. The ideal situation requires an Integrator who has an overall responsibility for the whole project. The Integrator can be the dealer that sells the equipment, an outside contractor, the vendor or the end user itself.

VOIP project phases- a simplistic view

1-System requirements definition
- Definition of the existing telephony system, analog, digital and IP phones if any.
- Additional IP phones, total number and location
- Expected quality and bandwidth requirement
- Adding VOIP and/or replacing some digital/analog systems
- Stages of the project and future growth

2- VOIP readiness and planning
- Existing network infrastructure and capabilities
- Technology used in the network
- Access technology for remote sites
- ISP SLA (service Level Agreement)
- QOS parameter tests

3- Implementation

Refer to following two papers presented at VOICE2004 for an interesting and good approach to the VOIP Pre and Post Installation issues *[Reference A8]***.**

Appendix A

A1- 2004 USA PBX Market Review:
IP Telephony Drives PBX Market Resurgence from TEQConsult Group White Paper

A2-“ Implementing IP-PBX Systems: Lessons from the trenches” parts 1to 4 from PlanNet Consulting. VOICECON2004 Conference Proceedings.

A3- Assessment of VOIP service Availability in Current Internet
By Wenyu Jiang and Henning Schulzrinne - University of Columbia

A4- QoS Measurement and Management for Internet Real-time Multimedia Services
Wenyu Jiang – University of Columbia

A5- **http://www.www1.avaya.com/enterprise/whitepapers/ip_networking.pdf**
A VOIP requirements document.

A6- VOIP
Anshul Kundaje, Geetali Bhatia, Mangesh Dalvi, Meghna Haridas and Sumeru Nandi
University of Bombay
The first half of this paper is a good simple introduction to telephony and VOIP.

A7-TVSE Tip of the month June 2004 TELRAD-Connegy
VOIP bandwidth requirements.

A8- RTP standard RFC1889 from IETF

A9- Voice Quality in converging telephony and IP networks
http://www.iec.org/online/tutorials/voice_qual/

A10- Managing QoS in a Multi provider Network
By Jean Walrand- University of California-BERKLEY

A11- Internet Quality of Service: An overview
Weibin Zhao, David Olshefski and Henning Schulzrinne- University of Columbia

Appendix B

Network and VOIP Investigation tools

B1- VQProbe from TELRAD-Connegy.
This is a PC based software tool in a Master-Slave configuration to test Round trip time, jitter and packet loss under different situations, such as using different CODECS. Estimated GA is before the end of 2004.

B2-ETHEREAL Network protocol Analyzer.
This is a freeware protocol analyzer that can be used to analyze networks, trouble shoot. There are PC Windows, LINUX/SOLARIS and MAC based installation. It is very powerful and helpful.
It can also analyze capture files from different analyzers.
http://www.Ethereal.com/

B3- Prismlite
From RADCOM. A network traffic analyzer and traffic generator for LAN/WAN; TCP/IP, ATM and VOIP traffic.
http://www.radcom.com/

B4- Qcheck IP network performance tool
A freeware tool. The console can run on Windows 2000, XP or NT. End point can also run on LINUX. It can check delay for UDP/TCP traffic for different packet sizes.

B5- NetIQ Chariot
A tool for testing network and VOIP performance. Can generate traffic and test network Performance in different network parameters.
http://www.netiq.com/

B6- NetAlly from Viola Networks
A tool for testing VOIP readiness and troubleshooting. It can measure delay jitter, loss etc
The WEB site also contains a MOS (Mean Opinion Score) calculator that can give a measure of the perceived voice quality by changing the jitter, loss and delay parameters.
http://www.violanetworks.com/

B7- Empirix Hammer
VOIP and Network test tool. Can create and analyze VOIP traffic and VOIP quality.
http://www.empirix.com/