VoIP Network Testing:
A Capstone Project Spring 2005

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05/13/05

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# Table of Contents

VoIP overview........................................................................................................... 3  
Purpose...................................................................................................................... 4  
Testing ..................................................................................................................... 5  
Lab Equipment........................................................................................................ 5  
Jitter ......................................................................................................................... 10  
MOS .......................................................................................................................... 11  
QoS............................................................................................................................. 12  
Security .................................................................................................................... 18  
Project Summary..................................................................................................... 20  
Special Thanks ....................................................................................................... 21  
Reference ............................................................................................................... 22
VoIP Overview

VoIP, or Voice over Internet Protocol, is a method for taking analog audio signals like the kind you hear when you talk on the phone, and turning them into digital data that can be transmitted over the Internet. The interesting thing about VoIP is that there is not just one way to place a call. There are variations of VoIP devices that enable a conversion of digital signal to an analog signal. The first is an ATA (Analog Telephone Adaptor) which is attached to a traditional analog phone, and installation is complete. The second option is an IP phone, this device typically has more features, and a built in ATA to convert analog to digital signals. Third option is PC to PC, or PC to phone; this requires VoIP specific software to be installed on a PC to dial outside the house.

With VoIP, you can make a call from anywhere you have broadband connectivity. This enables travelers to take their phones or ATA’s with them on trips and have access to the home phone. Remote access to voicemail and dialing features online.

Existing phone systems are driven by a very reliable but somewhat inefficient method for connecting calls called circuit switching. Circuit switching has been used for more than 100 years. When a call is made between parties, the connection is maintained for the duration of the call, unlike VoIP which is only sending packets when there is conversation. The circuit switching was the foundation of the Public Switched Telephone Network (PSTN).
Telephone conversations over today’s traditional phone network are somewhat more efficient and they cost less. Your voice is digitized, and your voice along with thousands of others can be combined onto a single fiber optic cable for much of the journey. If you look at a typical phone conversation, much of this transmitted data is wasted. VoIP technology uses the Internet’s packet-switching capabilities to provide phone service.

VoIP has many features that are beneficial that will go above and beyond current Telephony services. Seamless call routing, tying phone numbers to customer records that can be displayed to and operator as some specific benefits. The key to the success of this particular product was the use of dedicated end-to-end transmission facilities. The VoIP compression technology was effective, but the application relied on avoiding the unpredictable public-access Internet for inter-city transmission.

**Purpose**

As a group our concentration in this project was to generate traffic in order to stress or break our VoIP network that we setup in a professional manner. Working as a group knowledge was gained in various areas of understanding and managing the VoIP network. The question, then, is whether VoIP can produce the quality required for acceptable speech communications in a business environment. And in spite of the problems experienced in many applications to date,
the answer is still a yes - it just takes some care in addressing the various elements that can contribute to poor performance. We challenged ourselves with the task to find the limit of these elements that would cause poor performance.

Using G.711 as a benchmark for testing and comparing various speeds of connection to others as well as amounts of traffic across the VoIP network was the concentration of the project. An MOS (Mean Opinion Score) calculation was used to rate the quality of voice carried between phones.

**Testing**

Testing on the VoIP network was done by using two different information transfer protocols such as TCP and UDP based traffic generator programs. As well as file transfer protocol servers to upload and download files to a specified server. While the traffic was being generated we made various phone calls between IP Phones that were provided, and recorded quality of the call using an MOS calculator.

**Lab Equipment**

Equipment for the VoIP lab was provided by Nex-Tech as well as some assistance on explanation of equipment the capabilities. The equipment used includes:
1- 7912 series, Cisco IP Phone

The Cisco IP Phone 7912G is a basic IP Phone addressing the voice communication needs of a cubicle worker who conducts low to medium telephone traffic. A pixel display and dynamic soft keys allows easy access to a core set of business features. A maximum of two calls and one directory number is supported, in addition to inline power and an integrated 10/100 Ethernet switch for connecting a PC.
1- 7960 series, Cisco IP Phone

The Cisco IP Phone 7960G is designed to meet the communication needs of a professional worker in an enclosed office environment - an employee who experiences a high amount of phone traffic in the course of a business day. It has access to multiple telephone lines (or combination of lines and/or direct access to telephony features). High quality, hands-free speakerphone capability and built-in headset connectivity are included. A large pixel-base display provides supplemental information, access to applications, and makes it easy to use telephone features.
1- 7905 series, Cisco IP Phone

The Cisco IP Phone 7905G is a basic IP Phone addressing the voice communication needs of a cubicle worker who conducts low to medium telephone traffic. A pixel display and dynamic soft keys allow easy access to a core set of business features. A maximum of two calls and one directory number is supported, in addition to inline power for receiving power over Ethernet.
1-800 series, Cisco Router

This router provides secure Internet and corporate network connectivity to small remote offices and telecommuter. The Cisco 800 Series of Secure Routers provide a wide range of rich integrated security services, advanced Quality of Service (QoS) features for high quality voice, video and data applications, and easy deployment and remote management features with Cisco IOS software.

(www.cisco.com)
1- Zyxel Prestige 600 series modem

This modem enables a user to connect with a PPPoE username and password provided by the ISP.

**Jitter**

Jitter, when referring to VoIP, is the delay between packets that are arriving caused by network congestion, timing drift, or route changes. Jitter will contribute to poor performance and lost packets. Jitter can be thought of as shaky pulses of power. Causes of jitter may include electromagnetic interference and cross-talk with other signals. Abuses such as spam, viruses, worms and denial-of-service attacks can overload networks and routers, causing normal traffic to grind to a halt. The allowable jitter depends greatly on the application.

Recommended amounts of jitter on a VoIP network according ITU are 1 percent maximum packet loss, and 200 milliseconds of one-way latency as well as maximum jitter of 30 milliseconds. Results of some testing in our project indicated packet loss of 74.14%%, maximum
jitter of 800+ ms, and 400+ ms of average one-way latency. Voice and video applications are less tolerant of loss, delay, and delay variation (jitter) than data, but their requirements are more straightforward. Data applications vary widely in their QoS requirements, so they should be profiled before determining appropriate classification and scheduling treatment.

Packet jitter that is experienced is caused by changes in the inter-arrival gap between packets at the endpoint. Traveling packets should arrive appropriately and in order for a seamless conversion into analog voice. If the gap changes there may be a decrease in quality of the call. If the gaps are substantially large the packet jitter buffer is unable to wait for the packet that was lost, and the phone will drop that packet.

**MOS**

Mean Opinion Score (MOS) scale, which is the ITU Recommendation for defining voice quality scores. The quality of a voice sample by a wide range of listeners on a scale of 1-5, 1 being bad and 5 being excellent is the basis of the scale. This scale monitors live customer calls, producing a quality score based on ITU-T P.862 standard for objective speech quality assessment. MOS scales are available to license today for integration into network management equipment, VoIP devices, and network infrastructure.
Mean Opinion Scores (MOS) for Various Voice Quality Tests

<table>
<thead>
<tr>
<th>Score</th>
<th>Opinion Scale: Conversation Test</th>
<th>Difficulty Scale</th>
<th>Opinion Scale: Listening Test</th>
<th>Listening: Effort Scale</th>
<th>Loudness: Preference Scales</th>
</tr>
</thead>
<tbody>
<tr>
<td>5</td>
<td>Excellent</td>
<td>----</td>
<td>Excellent</td>
<td>Complete relaxation possible, no effort required</td>
<td>Much louder than preferred</td>
</tr>
<tr>
<td>4</td>
<td>Good</td>
<td>----</td>
<td>Good</td>
<td>Attention necessary; no appreciable effort required</td>
<td>Louder than preferred</td>
</tr>
<tr>
<td>3</td>
<td>Fair</td>
<td>----</td>
<td>Fair</td>
<td>Moderate effort required</td>
<td>Preferred</td>
</tr>
<tr>
<td>2</td>
<td>Poor</td>
<td>----</td>
<td>Poor</td>
<td>Considerable effort required</td>
<td>Quieter than preferred</td>
</tr>
<tr>
<td>1</td>
<td>Bad</td>
<td>yes</td>
<td>Bad</td>
<td>No meaning understood with any reasonable effort</td>
<td>Much quieter than preferred</td>
</tr>
<tr>
<td>0</td>
<td>----</td>
<td>no</td>
<td>----</td>
<td>----</td>
<td>----</td>
</tr>
<tr>
<td>Result</td>
<td>MOS(C)</td>
<td>%D</td>
<td>MOS</td>
<td>MOS(LE)</td>
<td>MOS(LP)</td>
</tr>
</tbody>
</table>

**QoS**

Our VoIP labs used two methods of QoS which included both Low Latency Queuing (LLQ) and Link Fragmentation and Interleaving (LFI). The Low Latency Queuing provides strict priority for Class-based Weighted Fair Queuing, which our case the priority was placed on
the voice transmissions. The Link Fragmentation and Interleaving is used to breakdown or fragment the voice streams into smaller delay-sensitive frames which allows them to take priority over larger transmission types such as file transfers (FTP, TFTP, etc.). The combination of these two allows for less jitter and delay in the voice transmission and in result produces far superior voice quality for VoIP calls.

Diffserv, Differentiated Services, is a method for adding QoS to IP network and is an IETF standard. Diffserv operates solely at the OSI layer 3, the network layer, using the first seven bits of the IP Type of Service (TOS) field as the Diffserv byte. The Diffserv byte is used for traffic classification and marking with Diffserv. The Cisco 831 router that was used in the VoIP lab tests assigned the traffic to a Priority Queue (PQ) in its CBWFQ mechanism. The router was configured to assign or mark the traffic with one of three different classification Diffserv bytes:

1. VoIP control traffic was tagged as the Diffserv Per-Hop Behavior (PHB) of Class-Selector 5 (CS5), which in binary format is 101000.

2. VoIP payload traffic is tagged as the Diffserv PHB of Expedited Forwarding (EF), which in binary format is 101110.

3. All other traffic, may it be basic web traffic or file transfers is tagged with the Diffserv PHB of Default (0), which in binary format is 0000000.
Each of the classes listed above has its own characteristics. Class-Selector 5 is basically used because of its backwards compatibility to older network hardware/software. Expedited Forwarding (EF) is designed to create a transmission with little to possibly even no delay. EF packets are guaranteed a configurable rate or amount of bandwidth in order to ensure that those packets are transmitted in a time-sensitive manner. The Default Diffserv PHB is nothing more than a “best-effort” type of transmission. If there is enough available bandwidth it will not experience many problems, but if the network is loaded down it may experience a large packet-loss percentage.

There are many other Diffserv PHB classes that are available but to stay relative to our VoIP testing labs, here is the list of the classes from highest priority to lowest priority:

1. Expedited Forwarding – Voice payload traffic.
2. Class-selector 5 – Voice control traffic.
3. Default – All other traffic.

The long and short of it, VoIP payload traffic (EF PHB) takes priority over all other traffic, which is then followed by the VoIP control traffic (CS5 PHB), and then all other traffic (Default PHB) is sent on a best-effort basis, with no guarantee of service.

One of the greatest ways to view the difference between VoIP calls using QoS and those without is comparing their Mean Opinion Scores. The industry standard for an acceptable call is
somewhere around an MOS score of 3.5. All of our non-QoS monitors have lower than acceptable scores, while the entire group of QoS monitors received MOS scores that far exceed the acceptable baseline. Our group used the MOS calculator provide by Volptroubleshooter.com, which is relatively accurate to providing quality of voice scores.


Here are the results from our MOS comparisons:

![MOS Score Comparison](image)

*Keep in mind that the acceptable score bar is based on the 3.5 MOS score standard for acceptable voice quality.

On the next few pages the screenshots from the MOS Calculator will be provided.
256Kbps without QoS (MOS score of 1.9)

- Select Codec Type: G.711 PLC
- Select Frame Size: 20ms
- Packet Loss Rate (%): 50
- MOS Score: 1.9
- R Factor: 17
- Bandwidth (kilobits per second): 80800

256Kbps with QoS (MOS Score of 4.4)

- Select Codec Type: G.711 PLC
- Select Frame Size: 20ms
- Packet Loss Rate (%): 0
- MOS Score: 4.4
- R Factor: 93
- Bandwidth (kilobits per second): 80800

512Kbps without QoS (MOS Score of 3.3)

- Select Codec Type: G.711 PLC
- Select Frame Size: 20ms
- Packet Loss Rate (%): 15
- MOS Score: 3.3
- R Factor: 64
- Bandwidth (kilobits per second): 80800
512Kbps with QoS (MOS Score of 4.3)

1.544Mbps without QoS (MOS score of 3.3)

1.544Mbps with QoS (MOS score of 4.3)
This is a captured file on a 256k non-QoS line that we tested. This can be played to hear the difference of the call quality on the same speed with and without Quality of Service implemented.

As a result, when things go wrong, users may hear bursts of static or other noise. Also, when the performance of the VoIP system starts to deteriorate, speech intelligibility generally drops to a point where functional communication is impossible.

This is a captured file on a 256k QoS call that we tested. This can be played to compare with the 256k non-Quality of Service call that was made. QoS implementation plays a major role in sound and voice quality of this call.

**Security**

Companies that are planning to migrate to the VoIP infrastructure are generally concerned with three areas aside from the obvious monetary concerns:

1. Interoperability
2. Voice quality
3. Latency
One of the major things that is often overlooked is the security risks that come with implementing voice and phone services over existing IP infrastructure. Security isn’t something that we tested throughout our project, but we do feel that it is important for us to at least touch on this subject due to its relevancy to the decision of choosing VoIP over standard telephone service.

Some measures that should be considered and taken include encryption of the voice data, building redundancy in your voice network, and locking down your VoIP server (Which in the geographical location will more than likely be controlled by your ISP, or VoIP provider).

Eves-dropping is still considered a problem with VoIP networks as well as its traditional telephone predecessor. When VoIP information is sent across the network it is sent as individual packets, which can be picked up by any packet-sniffing program and then compiled or restored into an audio file that is identical to your voice call transmission. Encryption is a relatively effective way to protect against such reconnaissance actions. Also, if you have an insecure VoIP server it could be easy for a hacker to obtain vital information about calls, customer information, etc. There are several ways to prevent attacks on your VoIP server, some of which include firewalls, access list, authentication, and other pre-established security methods.

The last security risk that we are going to touch on is the failure of your voice network, which can be caused by a handful of different reasons. One common cause of this is problems
with the physical network itself. If it very important to have a well documented network summary, so it is easy to troubleshoot any physical problems. Another way to prevent problems occurring is to restrict access to the room or rooms in which the network resides. Secondly, the network could be shutdown due to flooding. Flooding can be an induced by someone who has hacked in and used a traffic generator to flood the network. Using security measures like firewalls, user filters, authentication, and other security conventions again best protects against these threats.

**Project Summary**

Users and manufacturers need to seek ways to minimize the impact of congestion on VoIP to the greatest extent possible. Remember that VoIP generally works well if everything is perfect - as if life were that easy! It is quite obvious that each company, or customer is going to require different services and capabilities. It is because of this, that it is really hard, nearing impossible to make one distinct recommendation on what a company should do in regards to VoIP. The plus side being that there are many options a consumer can take, and most likely there will be a close fit to the desired services and performance for which the consumer is searching.

There is a common misconception that adding more bandwidth will solve all of network congestion problems. This theory is clearly proven inaccurate with our testing. Our tests’ results
show that adding bandwidth actually does improve the voice quality, but it still doesn’t improve enough to experience voice transmissions that are at an acceptable level. Basically, added bandwidth is a “band-aid fix” for a problem that requires much greater attention.

The major recommendation that we would like to suggest is that some type of QoS is implemented. If there is any substantial traffic across the network, the voice transmission is generally considerably jeopardized. It is absolutely crucial for the smaller voice packets to have somewhat of priority over the larger transmissions, such as FTP, TFTP, etc.

It is important to keep the service and performance in which you require in mind. If you are going to predominantly use the network for web and email, it is very acceptable to use a lower broadband connection. If you are going to be sending many large file transfers, and other bandwidth intensive applications, it will be very important that you have a link speed that will accommodate for those services.

**Special Thanks:**

We would like to thank Nex-Tech for allowing us to use their equipment and for assisting us greatly throughout our project. We would also like to thank all of the faculty and staff who have provided input and directed us through the project. And finally, we would like to thank www.voip-troubleshooter.com for providing the MOS calculator that we used to figure the MOS scores for the individual tests.
Reference

1. 7940 series -

2. 7912 series -

3. 7905 series -

4. 7960 series -

5. Cisco 800-

6. MOS Score
   http://www.voiptroubleshooter.com/diagnosis/emodel.html