



VOIP PERFORMANCE MONITORING SYSTEM

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Abstract- Voice over IP is transporting speech signals in an acceptable way from sender to destination over an IP network. The definition of 'acceptable' depends on the particular situation we are dealing with. When speech signals are being transported as part of a real-time communication between two persons, it will mean that the real-time aspects of this conversation must be respected. The overall delay between sending and receiving should be low to avoid irritably long gaps of silence. If, however, speech signals are being transmitted as part of a one-way process - e.g. an on-line radio show or a lecture - the delay constraints are less strict since the interactive aspect is no longer present. This acceptable form of voice transmission and receipt is possible when the parameters, which contribute to the voice degradation, are measured and optimized. This is the scope of the Quality of Service (QoS) for Voice transmission using IP. Quality of Service, a term that specifies a guaranteed throughput level that is the size of data transferred from one place to another or processed in a specified time frame.

Keywords- Packet loss, Bandwidth, Latency, Jitter, CODEC-G.711, RTP

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Introduction

Today market, technology and regulatory factors are converging to make the integration of voice over data networks economically attractive. With the development of high-speed microprocessors and transmission medium, the ability for real time voice or image transmission is made possible. Since the last decade, the industry had been researching heavily in the use of the network and optimizing its bandwidth for real time data transmission. IP telephone involves the integration of the voice data into the network. With the consolidation of data in digital format, integration of different communication systems and improvement of the system could be made easier. As Internet is evolving faster and faster, people are trying to fully utilize their bandwidth, by having both voice and data services on the same network. Carrying Voice over Internet Protocol (VoIP) is becoming more popular as it reduces operational cost as well as optimizing the available bandwidth.

Voice over IP is transporting speech signals in an acceptable way from sender to destination over an IP network. The definition of 'acceptable' depends on the particular situation we are dealing with. When speech signals are being transported as part of a real-time communication between two persons, it will mean that the real

-time aspects of this conversation must be respected. The overall delay between sending and receiving should be low to avoid irritably long gaps of silence. If, however, speech signals are being transmitted as part of a one-way process - e.g. an on-line radio show or a lecture - the delay constraints are less strict since the interactive aspect is no longer present.

Interface Functions

This section gives the finer details of the software developed and is divided into the following sub-sections:

Performance Perspective: This deals with the overall performance issues of the software. It gives the external entities, which interact with the system for the input and outputs.

Product Functions: This gives the major functions, which the software performs, and the steps taken by the User Entities for the initialization and stimulation of the software.

User Classes and Characteristics: This gives the major classes of people who can use the software and derive useful data from it for further analysis.

Operating Environment: This gives the environment characteristics under which the software works in terms of the platform and software used.

Overview of Data requirements: This section includes the data that is input or output from the user and the system. It gives the important data, which circulates in the system.

General Constraints, Assumptions, Dependencies, Guidelines: This section describes non-functional requirements, those factors that impose constraints on the implementation of the software product.

Monitoring Parameters

Jitter Jitter controls the regularity in which voice packets arrive. Typical voice sources generate voice packets at a constant rate. The matching voice decompression algorithm also expects incoming voice packets to arrive at a constant rate. However, the packet-by packet delay inflicted by the network may be different for each packet. The result: packets that are sent in equal spacing from the transmitting client arrive with irregular spacing at the receiving client. Since the receiving decompression algorithm requires fixed spacing between the packets, the solution is to implement a jitter buffer at the receiving side. The jitter buffer deliberately delays incoming packets in order to present them to the decompression algorithm at fixed spacing.

Packetloss Packet loss is a normal phenomenon on packet networks. Loss can be caused by many different reasons: overloaded links, excessive collisions on a LAN, physical media errors and others. Transport layers such as TCP [1] account for loss and allow packet recovery under reasonable loss conditions. Audio CODECs also take into account the possibility of packet loss; especially since RTP data is transferred over the unreliable UDP layer. However, packet loss starts to be a real problem when the percentage of the lost packets exceeds a certain threshold (roughly 5% of the packets), or when packet losses are grouped together in large packet bursts. In those situations, even the best CODECs [5] will be unable to hide the packet loss from the user, resulting in degraded voice quality. Thus, it is important to know both the percentage of lost packets, as well as whether these losses are grouped into packet bursts.

Bandwidth A very important factor affecting voice quality is the total network load. When the network load is high, and especially for networks with statistical access such as Ethernet, jitter and frame loss typically increase. For example, when using Ethernet, higher load leads to more collisions. Even if the collided frames are eventually sent over the network, they were not sent when intended to, resulting in excess jitter. Beyond a certain level of collisions, significant frame loss occurs. Hence Bandwidth measurement becomes important. With the count of the number of Packets arriving per second with the buffer size gives the total bandwidth used.

Latency

In contrast to broadcast-type media transmission (e.g., RealAudio), a two-way Phone conversation [5] is quite sensitive to delay, most callers notice round-trip delays when they exceed 250mSec, so the one-way latency budget would typically be 150 mSec. 150 mSec is also specified, as the maximum desired one-way latency to achieve high-quality voice. Beyond that round-trip latency, callers start feeling uneasy holding a two-way conversation and usually end up talking over each other. At 500 mSec round-trip delays and beyond, phone calls are impractical.

Main Modules

Server Mib Module

Identification: SERVERMIB is a module placed on the server machine. **Type:** It is an independent software or process running on the server machine.

Purpose: Its main purpose is to act as an information base for the users details like:

- Username
- Password
- IP Address

Port Number

Functions: This module performs functions such as:

- Add New User
- Authenticate existing Users
- Sending list of Online Users
- Retrieving IP Address and port number of Requested User

Logging out users by deleting entry from table

Interfaces: There are two interfaces:

User Interface

It shows the status of the server as:

SERVER STATUS: currently active or not.

CURRENT STATUS: It displays messages such as Waiting for client.

Connected to client

Closing connection

STOP SERVER: will completely stop the server.

RESET SERVER: will remove the list of online users from the table but keep the server activated. It will stop the current transaction with the client.

PURGE LIST: will remove the list of online users displayed in the GUI without stopping the current transaction.

Client Interface

The TCP Socket acts like an interface between the CLIENT and SERVERMIB.

Client Modules

Identification: CLIENT module.

Type: It is a program running on each client machine.

Purpose: Its purpose is to mainly create an environment for the proper functioning of the two sub-modules:

USERSERVER

USERCLIENT

Functions: Its main functions are:

Creates the TCP socket for the USERSERVER module.

It calls the USERSERVER and USERCLIENT module.

It gives the portno of the TCP socket created, to the USERCLIENT module

Conclusion

I have designed our software with the approach in mind that the existing code can further be enhanced to provide additional features like:

- The Session Initiation Protocol (SIP) can be implemented in the CALLSETUP and CALLSETUPCOMPLETE modules.
- The different compression techniques can be included for efficient bandwidth utilization.

- For more secure voice communication encryption techniques can be implemented.
- Interfacing a Web-Camera and reading data from the video card can do video conferencing.
- Code can be implemented to optimize the performance parameters
- based on the measurements carried out by the software

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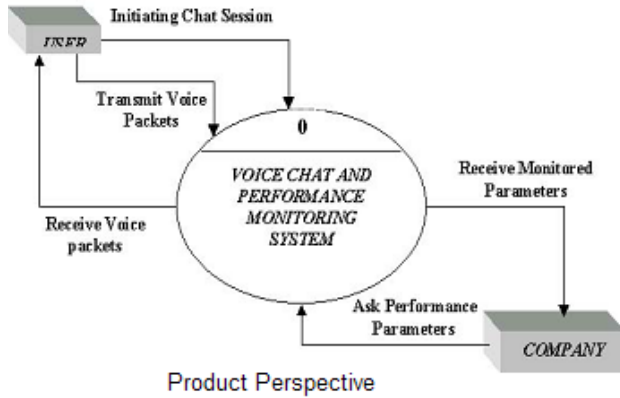


Fig. 1– Product Perspective

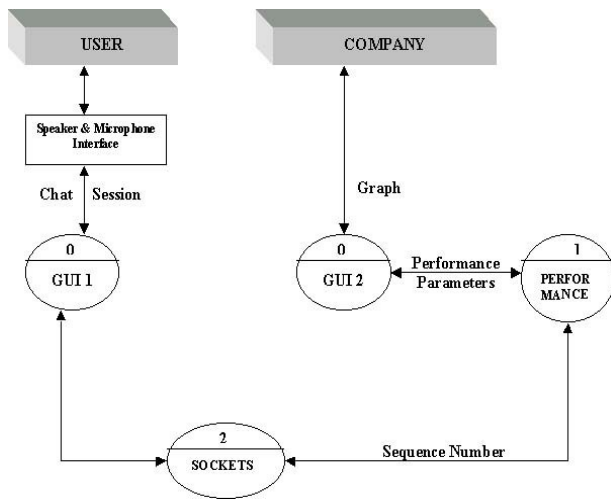


Fig. 2- Flow of Data

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