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A Simulation Based Performance Evaluation of VoIP

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Abstract

All the Communication Technologies of the modern age are going through a change which is termed as “Convergence” and IP is playing the role of the backbone of this trend. VoIP is one of blessings of this IP based communication trend though it has still so many demerits in terms of QoS degradation including Codec Compression, Packet Loss, Discarded Packets, Bit Errors, Frame Erasures and various compression schemes. This paper delivers an experimental study which evaluates the performance of VoIP by monitoring the different QoS parameters. The parameters which are aimed to take into consideration in this experiment are Delay, Jitter, Packet Loss and MOS observed by the end user.

Keywords: QoS, Jitter, MOS, Latency, R-Factor

1. INTRODUCTION

There are about 1 billion fixed telephone lines and 2.5 billion cell phones in the world that use the traditional public switched telephone network (PSTN) systems. Soon, they will move to networks based on open protocols- known as Voice over Internet Protocols (VoIP). This migration is fueled by many factors, like the tremendous growth of the Internet and the World Wide Web, the rapid and low cost coverage capability through wired and wireless networks, the availability of a variety of fixed and mobile Internet accessing tools (e.g., desktop, laptop, pocket pc, dual-band cellular, etc), and the feasibility of integrating voice and data into a single infrastructure. Performance measurement of VoIP Quality of service is very vital now days. Since more and more VoIP systems are appearing, characterizing the traffic of a VoIP system and studying the protocols to increase the quality of service can help to better understand and improve these systems.

In this experimental study QoS has been estimated from the side of end user and performance metrics have been analyzed, i.e. average packet loss rate and average jitter, maintained stable values and acceptable QoS levels. In VoIP there are also unique sources of degradation including codec compression, packet loss, discarded packets, bit errors, frame erasures and various compression schemes. Voice quality measurement considers the extent of all of these factors, whether they occur in the network or outside it, and determines the overall impact of quality in the opinion of the customer, a measurement known as a Mean Opinion Score (MOS). This paper presents VoIP software and VoIP hardware phones only for the G.711 codec.

2. QUALITY OF SERVICE AND RELEVANT ISSUES

2.1 Quality of service

Communication on the IP network is inherently less reliable in contrast to the circuit-switched public telephone network, as it does not provide a network-based mechanism to ensure that data packets are not lost, or delivered in sequential order. It is a best-effort network without fundamental Quality of Service (QoS) guarantees. Therefore, VoIP implementations may face problems mitigating latency and jitter [2].

2.2 QoS Parameters

In ensuring the QoS for VoIP, researchers have come out with some QoS parameters as to be observed. In general, the focus of this paper is directed to the Network QoS, based on the QoS Framework as suggested by [3]. This simply means that the analysis that has been taken out is to evaluate the network environment in which VoIP communication is being conducted. There are several QoS parameters that have been identified to be implemented in this study. Table 1 provides a list of Network QoS parameters available as derived from [4].

Table 1: Network QoS Parameters

Category	*Parameters*
Timeliness	Delay Response time Jitter
Bandwidth	Systems-level data rate Application-level data rate Transaction time
Reliability	Mean time to failure (MTTF) Mean time to repair (MTTR) Mean time between failures (MTBF) Percentage of time available Packet loss rate Bit error rate

In the analysis, three QoS parameters have been selected, namely delay, jitter, and packet loss rate. The main justification for this selection is such that the study focuses on the performance of VoIP communication over different networking conditions and environment, and hence the time and reliability would be the major concerns for evaluation. The results of studies conducted by also shown that high delay and high delay variability (jitter) has been experienced by a large number of Internet paths that resulting in poor VoIP performance. It should be noted that these QoS parameters have also been adopted in a number of previous studies [2, 4-5]

2.3 VoIP Quality Metrics

In a well planned network, the Quality of Service (QoS) features in the network equipment intelligently distinguish and route traffic based on its priority. By helping to guarantee that voice traffic gets the bandwidth it needs, the network controls the factors that compromise voice quality. These factors are:

2.3.1 Latency

As a delay-sensitive application, voice cannot tolerate too much delay. Latency is the average time it takes for a packet to travel from its source to its destination. The maximum amount of latency that a voice call can tolerate one way is 150 milliseconds (100 milliseconds is preferred) 1. If there is too much traffic on the line, or if a voice packet gets stuck behind a bunch of data packets (such as an email attachment), the voice packet will be delayed to the point that the quality of the call is compromised [4].

2.3.2 Jitter

In order for voice to be intelligible, consecutive voice packets must arrive at regular intervals. Jitter describes the degree of variability in packet arrivals, which can be caused by bursts of data traffic or just too much traffic on the line. Jitter is the delay variance from point-to-point. Voice packets can tolerate only about 75 milliseconds (40 milliseconds is preferred) of jitter delay [6].

2.3.3 Packet loss

Packet loss due to congestion is the losing of packets along the data path, which severely degrades the voice quality. Packet loss occurs frequently in data networks, but many applications are designed to provide reliable delivery using network protocols that request a retransmission of lost packets (e.g. TCP [7]). Dropped voice packets, on the other hand, are discarded, not retransmitted. Voice traffic can tolerate less than a 3 percent loss of packets before callers feel perceivable gaps in conversation. When these factors are properly controlled by QoS mechanisms, VoIP delivers better quality voice than they are accustomed to from dedicated voice networks, even over the lower speed connections. At the same time, data applications are also prioritized and assured of their share of network resources.

3. EXPERIMENTAL SETUP FOR MONITORING VOIP PARAMETERS

VoIP technology uses shared Internet bandwidth unlike other traditional communication technologies. Also, being a real-time application using the same bandwidth, VoIP quality is drastically affected by network parameters such as packet loss, delay and jitter as compared to other applications such as e-mail and Instant Messaging. This makes it important to have a monitoring system in place so as to keep one aware of the health of all vital VoIP quality parameters on a 24/7 basis[8].

For monitoring or analyzing VoIP QoS parameters and the sequenced schemes of a VoIP network here we use software. To generate Bulk calls or virtual calls for the monitoring Test Call Generator System (TCS) is established. Here we use the machine as a Local host.

The two interfaces we used are:

Device_1(C5C6C252-1C06-41D4-ACF9-952FC12847D4) for this IP:192.168.16.157

Device_2(56BBAF1B-9650-49D0-B968-AF7D942109F6) for this IP: 192.168.14.103

And 172.16.3.20

When the software settings are completed it opens a monitoring page which delivers us the every portion of a VoIP systems and its information. The Monitoring page is given below:

Fig.1 VoIP monitoring cell of Call Volume and Voice Quality

Fig.2 VoIP monitoring cell for Call Quality Trend, Call by status and Traffic monitoring

The "Monitoring" tool has the Call Volume graph showing the volume of calls that were successful or unsuccessful, of good, tolerable or poor quality, the peak and low usage periods, the Average Call duration and overall ASR (Answer Seizure Ratio) etc. The Voice Quality graph that shows snapshot of the QoS metrics with their Min, Max and Avg values and whether good, tolerable or poor as defined by the user.

4. Result and Analysis

4.1.1 Call Volume Analysis

The Call Volume Graph [Fig.3] provides the statistics of all the calls that were initiated in the chosen time frame (23hrs) as set in the calendar. The bar graph has the light areas representing the successful calls while the dark areas denote unsuccessful calls. The graph also provides the number of call attempts, successful and unsuccessful as a linked number. When these linked numbers are clicked one can view the information of the respective calls. The successful calls are divided into calls based on quality - Good, Tolerable and Poor quality calls. Calls with MOS (Mean Opinion Score) greater than 3.6 are classified as Good Quality Calls. Calls with MOS less than 3.1 are marked as Poor Quality Calls. Calls with MOS between 3.1 and 3.6 are marked as Tolerable Quality calls. Answer Seizure Ratio (ASR) - ratio of successfully connected calls to attempted calls is displayed Lists the time period at which there were a maximum number of calls - its peak usage; Similarly, time period at which the call volume was minimum - low usage Provides the average call duration time across all the calls during the selected time period .The Call Volume graph represents a summary of the calls that have happened over the time period as chosen in the calendar.

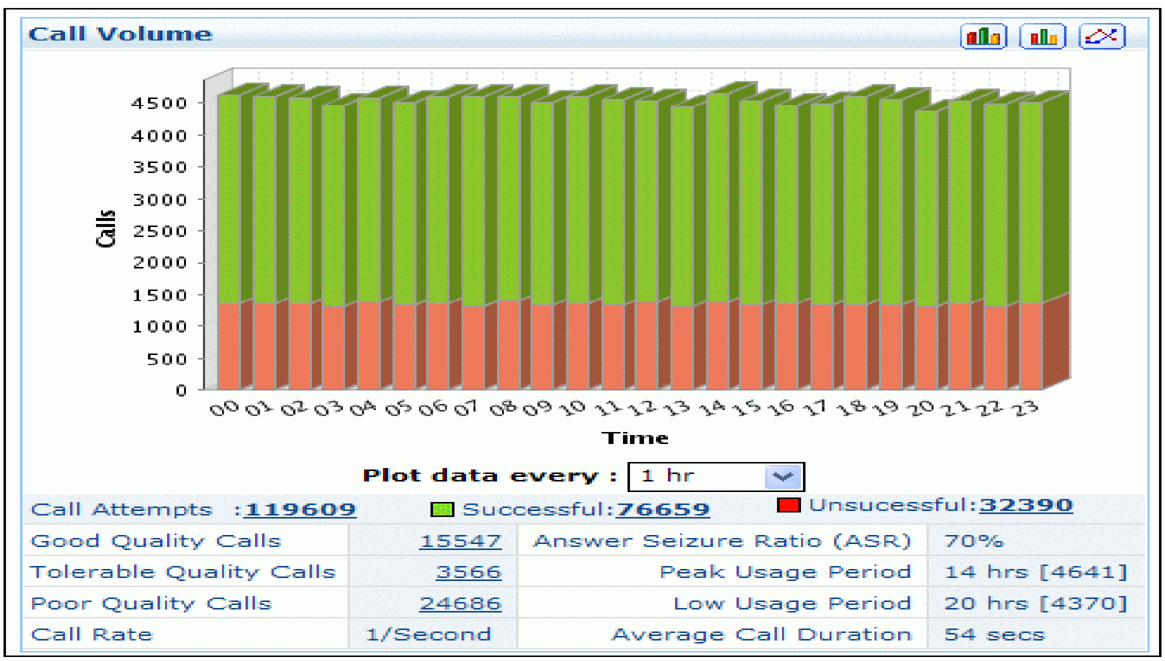

Fig.3 Call Volume Graph

4.1.2 Voice Quality Analysis

The Voice Quality graph [Fig.4] provides trends of vital quality parameters Delay, Jitter, Packet Loss, MOS and R- Factor over the defined time period as set in the calendar. A table of information shows the "Min", "Max" and "Avg" values for each quality parameter and also which of these values were "Good", "Tolerable" or "Poor" as defined by the user. Trend graphs are displayed for each parameter. From graph it can be concluded that delay and jitter parameters of the successful calls are good where as the MOS (Mean Opinion Score) is poor.

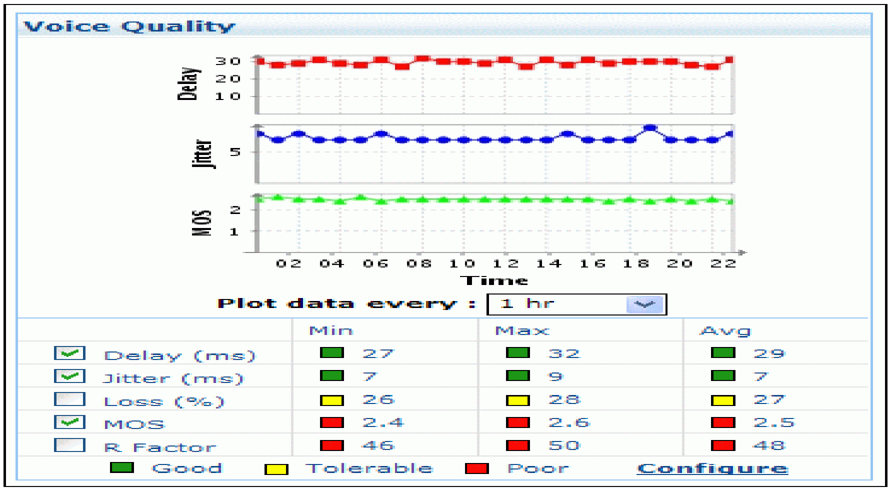

Fig.4 Voice Quality Graph

4.1.3 Call Quality Trend

Call Quality Trend shows the quality parameter - Good, Tolerable and Poor quality calls on an hourly basis. Each individual Quality Trend plot on the graph can be clicked to provide details of the total number of calls for the selected time period.

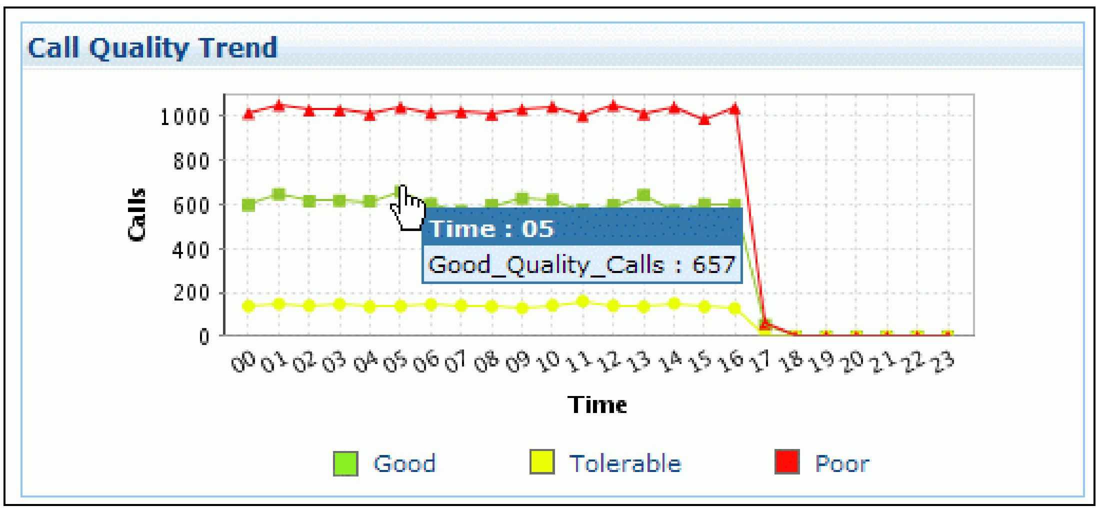

Fig.5 Call Quality Trend

The Call Quality Trend graph provides us the standard of the successful calls over the period of 23 hours .Over the period around 33% calls shows good quality, approximately 11% shows tolerable and 55.55% shows poor quality.

4.1.4 Calls by Status

Fig.5 shows the percentage and number of Successful, Unanswered, Error and Unmonitored calls on a day to day basis as set in the calendar. The successful calls are shown in green and include Good and Poor quality calls. The Unsuccessful calls are divided into unanswered calls and Error calls. Calls that were not answered by the user (user busy, user not available etc.,) fall into the Unanswered Call category. Other calls that failed due to errors in server or client are marked as Error Calls Unmonitored calls are those that monitor has abruptly stopped monitoring - this could be due to an abrupt stop and restart of the monitor server or because there were no voice packets that were received for a continuous duration (30 seconds) of time.

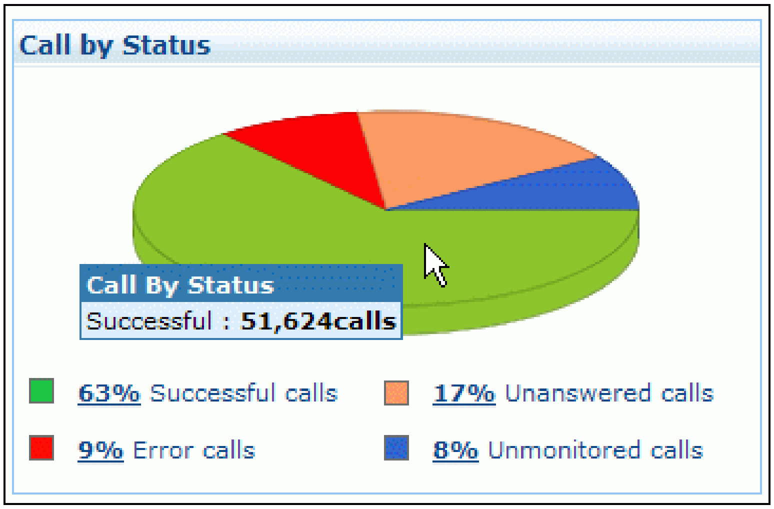

Fig. 6: Calls By Status

4.4.5 Call Report

The Call Report [Fig.7] delivers the view of Average Call Duration (ACD) graph and a pie chart gives Quality Split and Unsuccessful Calls Split. The Average Call Duration (ACD) Trend shows the duration of successful calls (in seconds) for every hour. The Quality Split identifies the percentage and number of Good, Tolerable and Poor call quality. The min and max threshold MOS value configured under Voice Quality indicates the Good, Tolerable and Poor Quality calls. The Unsuccessful Call Split lists the top 10 classification for an unsuccessful call. The pie chart indicates the classification of all unsuccessful calls along with the reason for all failures and number of calls.

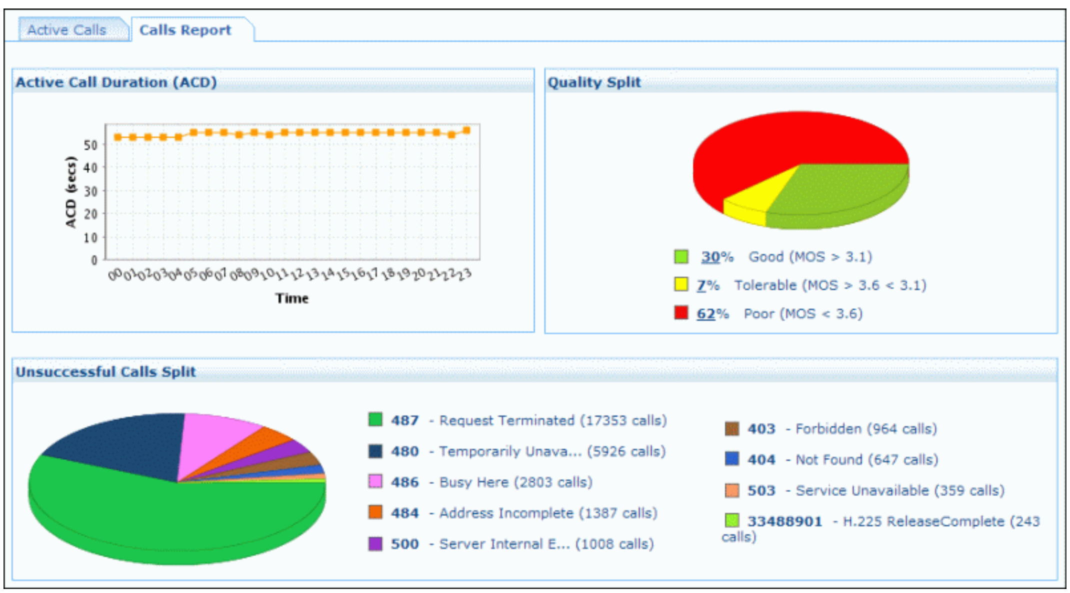

Fig.7: Call Report

CONCLUSION

In this research work, a simulation has been performed for 23 hours to monitor the VoIP QoS parameters and a large number of VoIP calls have been monitored with a monitoring tool. In terms of call volume it has been found that almost 64% calls are successful. The quality of the successful calls is good in terms of Jitter and Delay whereas it is poor in terms of MOS (Mean Opinion Score). The call status analysis shows that among the unsuccessful calls, 17% are unanswered, 9% are error calls and 8% are unmonitored. The call report of Fig.8 also provides a very precise analysis of the unsuccessful calls. From all the analyzed data it is very clear that still there is a huge space left for the researchers to improve the QoS of VoIP which can be a good ground for the further research in this field.

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