ISSN 2250-3153

Improve QoS (Quality of Service) by destructing Security Issues

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Abstract- VoIP services are scarifying, due to delay in transmission of voice packets (Latency), echo, jitter, packet loss and other Quality of Service (QoS) problems arrived in VoIP phones. The weakness on QoS is managing by voice and data service together on a VoIP network. In this paper we dedicated to studying the response of data transmission quality, characteristic of voice quality, the service quality and diagnoses the problem arrived on VoIP protocol suite. For this reason, we suggest, to manage the weakness of VoIP related protocol by design of measurement management process model also use to analyzed quality item driven per voice service that can increase the quality and come up the result of MOS, delay, jitter, packet loss value, voice processing, filtering, try to achieve Qos requirements, removing the miscellaneous QoS security issues, Improving QoS, and obtained the idea about Software Queueing Mechanisms etc.

Index Terms- VoIP; Performance and Improvement of QoS; QoS Security Issues; Testing QoS and Measuring Process; QoS Software Queueing Mechanisms;

I. Introduction

This research paper guides a stepwise overview about QoS improvement through destruct the vulnerabilities. According to TIA (Telecommunications Industry Association), more than 75% private branches are IP-based. In telecommunications some places the bandwidth is no longer an issue there is a lot of space for data and voice to be dispatched together only two main things to worry about in VoIP calls, first is Latency (data packets are delivered too slowly, due to network congestion) and second is Jitter (variation in delay of packets, leading to packet loss) thus, enough bandwidth is not the only solution for safe, secure and fast telephony. Further, Internet telephony development has many of the secure methods of transmitting voice, data, fax and related services using IP based networks. Voice over IP (VoIP) is the transmission of voice and data over networks using the internet Protocol. IP networks have become very popular in the past few years, due to the exponential growth of the public Internet leading the way in to the IP world. Voice over IP (VoIP) is the set of protocol that allow voice traffic to be transported on IP networks. The simplest way for transmission of audio signals across the IP network is to digitize an analog signal first to produce data file and packetized at the sender at regular intervals, using an encoding algorithm and then sent over IP network using transport protocol such as User Datagram Protocol (UDP) or the enhanced Real-Time transport Protocol (RTP).When voice packets are received at the destination, they are reordered and converted back to analog for audibility. VoIP systems as a packet-based technology has several advantages are lower cost, highly flexible and provide better service.

II. RELATED WORK

In this paper we estimate the current work related to VoIP, the objective method and observed the performance of the system. The objective of this research is to minimize the error, introduced in channel due to the network deterioration and hence improve the quality of sound, analyze the voice quality by using channel coding scheme. There are two QoS measuring techniques first is subjective method and second is objective method. In subjective method we measure the Mean Opinion Score (MOS) and in objective methods we use matching algorithms to measure the difference between decoded outputs of source coding system and joint with source-channel coding system. Then we used E-Model to illustrate the attempts to develop QoS measures which provides better solution for monitoring QoS, There are different methods for estimating the packet delay used in E-Model to calculate delay from the header fields of the RTCP packets calculation by using the RTCP stream. This method is the good for calculating delay and proposed a new algorithm to measure the packet loss burst Ines in the ITU E-Model. E-Model has a new parameter called equivalent random packet loss used to replace the random probability of packet loss in order to calculate the R-Factor and MOS value and achieved improved measurement accuracy under burst packet loss conditions. Many QoS system solutions have been proposed for monitoring voice quality for VoIP, but not accepted to give full efficient solution for transmission factor (R). Our proposed work is targeted for the improvement of the VoIP quality measures in the future networks with new coding schemes that should be improved control of system quality, improved the measurement of customer satisfaction and provide the efficiency for network mechanisms example, system performance progress and billing. This paper illustrate various QoS security issues where we discuss problem about latency, jitter and packet loss, Discuss how to improve Quality of Services, Testing QoS and Performance measured, QoS design issue, Factors Affecting Voice Quality & Quality Item to be measured and at last we illustrate QoS software Queueing Mechanism which includes First-In First-Out (FIFO) (default),Priority Queuing (PQ), Custom Queuing (CQ), Weighted Fair Queuing (WFQ), Class-Based Weighted Fair Queuing (CBWFQ),Low-Latency Queuing (LLQ).

III. MISCELLANEOUS QOS SECURITY ISSUES

SIP phone endpoints may freeze, and crash due to high rate of network traffic. SIP proxy servers may experience failure, with VoIP-specific signaling attack which is less than 1 MBPS. This

ISSN 2250-3153

research paper gives several security steps of an irreversible chain of effects on Quality of Service of VoIP systems.

A. Latency

Latency refers how much time it takes for a voice signal to go from source to destination. Ideally, it should be zero, but it has upper and lower toleration limits. In the United States, the acceptable limit is 150 ms for one-way traffic and for international calls; it is 400(ms). The VoIP service provider should allow a Latency budget, which may look like this:

Table1. Sample Latency budget

Delay Source (G.729)	**Budget(ms)**
Device Sample Capture	0.1
Encoding Delay(Algorithmic Delay + Processing Delay)	17.5
Packetizketization/DePacketization Delay	20
Move to Output Queue/Queue Delay	0.5
Access (up) Link Transmission Delay	10
Backbone Network Transmission dealy	Dnw
Access(down) Link Transmission Dealy	10
Input Queue to Application	0.5
Jitter Buffer	60
Decoder Processing Dealy	2
Device Playout Dealy	0.5
Total	121.1 + Dnw

B. Jitter

Jitter refers to non-uniform packet delays. It occurs in low bandwidth situations. Jitter may cause packets to arrive, and processed out of sequence. When jitter is high, the packets arrive at their destinations in that situation several automobiles coming to a unplanned stop at a traffic signal. There are basically three useful mechanisms to control jitter for Usage buffer:

1)

- *It is used at the endpoint of traffic, where the buffer has to release its buffer packets once every 150 ms.*
- *Routers with effective headers are regulated using routers with headers, or buffer instruments.*
- *Efficient use of bandwidth; a great deal of jitter is taken care of by effective use of bandwidth.*

B.

C. Packet Loss

In VoIP occurs extreme manner of packet loss while communication of IP network. Packet loss is a byproduct of excessive latency or jitter. The Combination of the packet loss problem is trust on RTP, which uses unreliable UDP for transfer. By that time a packet get reported for something, QoS always gets exceeded. Packet loss is an inescapable reality, and apart from increasing bandwidth, there are no other independent solutions for counteracting packet loss.

IV. QUALITY OF SERVICE DESIGN ISSUES

The design of Voice Quality is concern of the main issue through VoIP. VoIP is a synchronous and real time application. When it comes together with IP the call quality problems may arrive. For considering the VoIP design, we must examine the factors that negatively affect voice quality, and the fault tree analysis must take a subjective account into the quality of service which being addressed as a central issue.

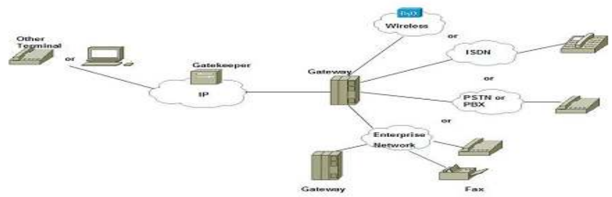

Fig 1: Design of VOIP Network

To sensed voice quality is a function of many factors: jitter, delay, Echo and packet loss. Echo, is occurred when hybrid circuits in a telephone network convert between a four-wire circuit, and a two-wire circuit. Echoing occurs when there is an audible leak between sending and receiving areas. Echo reduced by physically shortening the distance between the sending and receiving circuits. To building a reliable security substructure comes at the cost of declined satisfaction in Quality of Service (QoS) for VoIP communications. Here are several appropriate measures designed to improve this factor In order to quantitatively describe the performance of VOIP modules, it is important to consider over QoS issues, the following performance table is useful:

Table2. Voice Quality Measurement

Parameter	**SCORE**				
	1	2	3	4	5
Latency (ms)	< 50	50 – 75	75 – 100	100 – 200	>200
Jitter (ms)	> 5	5 – 10	10 - 50	50 – 100	>100
Packet/ Loss (%)	0	0 – 1	1 - 2	2 - 3	>3

VoIP deployment: Table 2 helps for Project Simulation. A Deployment for a new product has always been a great challenge so it's a responsibility of network administrators, to balance the productivity with costs and integrate with existing infrastructure. Other issues deserve priority of troubleshooting, teething and compatibility problems during initial stage, and performance tuning and maintenance later. So that in this paper, we discuss different parameters connected with a reliable p2p (peer to peer) VoIP deployment.

A. Bandwidth

We know that a voice is more sensitive than data so the bandwidth requirements must automatically go up. The configuration of VoIP device, use bandwidth-intensive codec, such as G.729-A, which can use 87.2 kbps of NEB (Nominal Ethernet Bandwidth) Also the existing traffic patterns on the network should be identified. If it is overcrowded with a lot of broadcast traffic then it will affect to response time. For the best results for p2p communication, it should not be merge with other signals, such as TV and Radio. At the time of VoIP deployment, it is mandatory to have a separate VLAN, which will keep the voice and data networks separate. About bandwidth the design criteria is to clarify the number of simultaneous voice calls that can be made over WAN links. For that the compression technique should be known for payload size of voice packets. For deployment of bandwidth several VoIP calculators are available on www.voip-calculator and www.ixiacom.com.

B. QoS

Toll-Quality calls require at least 16-20 kbps of bandwidth. There are also bandwidth management solutions.

C. Handling power problems

All p2p phones suffer from power outrages. So that good-built UPS is an effective solution. However, a good bit of power planning from source end is desired, as to how much backup power is needed.

D. Security issues

Apart from security mechanisms discussed in the paper, the network should be subject to routine maintenance, taking regular back-ups, installing antivirus software, etc. Also, implementing VoIP performance analyzers, like “Appmanger 6.0” from NetIQ for smooth functioning of VoIP is a good idea. At each LAN, network architecture should be such that at least 70% throughput is achieved within 10 msec.

V. MEASUREMENT BASED ADMISSION CONTROL

Measurement based admission control consists of two parts: first is measurement, which is used to estimate the current network load and second is admission control based on estimated network load. So the measurement-based admission control algorithm can be classified into two types, local load measurement-based and end-to-end load measurement-based. Where in local load-based measurement algorithms, each router along the flow path measures its local load status and conducts admission control where End-to-end measurement based admission control algorithms, rely on the end hosts or edge nodes to estimate the available network resources. The end hosts and edge nodes may actively send probe messages to get the network status. The core routers usually do not involve in the estimation. If we compare with the traditional parameter-based admission control algorithm then measurement-based admission control algorithm suffers a certain degree of QoS degradation and it can only provide the soft QoS guarantees. Several measurement-based admission control algorithms have been used in existing VoIP products: Cisco’s measurement-based call admission control algorithm and traffic matrix-based admission control algorithm (TAMC). Cisco’s measurement-based call admission control algorithm has*C.* two types: Advance Voice Busy out (AVBO) and PSTN Fallback. They both use Security Assurance Agent (SAA) to conduct measurement. SAA adopts Response Time Reporter (RTR) method to measure the network performance, such as delay, packet loss, delay jitter and ICPIF (Calculated Planning Impairment Factor, ITU-T G.113), etc., and feeds the measurement results back to source gateway for admission control. When the network performance is below the pre-determined threshold then admission control unit will be activated. The difference between AVBO and PSTN Fallback is that PSTN Feedback doesn’t configure IP addresses statically and uses cached addresses. Those two mechanisms can only be used in the network with the support of Cisco’s SAA. Traffic Matrix admission control (TMAC) provides QoS guarantees within a single domain or multi-domains. It uses Clean House architecture to provide intra- and inter- QoS guarantees to Voice over IP networks. TMAC uses four metrics: demand metrics, upper-bound Matrix, Node-level Traffic Matrix and POP-level Traffic matrix to describe the total bandwidth requirements, the resource threshold, and measured load of each entering pair or pop pair respectively. Therefore the admission control only checks whether the sum of the peak rates of the new flow and the measured traffic of the same path exceeds the corresponding upper-level item. In summary, measurement-based admission control can only provide soft (rather than deterministic) delay guarantees for voice traffic. It relies on the measurement period to reflect the dynamics of network status. In Short the measurement period, conduct more frequently and reflect the networks better, but consumes more network resources by this reason longer measurement period cannot reflect the network dynamics well. Also in some cases the research messages are not payload messages themselves, which may not be able to reflect how the network treats the real voice packets.

VI. QUALITY ITEM TO BE MEASURED

About traffic parameters which point to quality features like dial tone delay, post dialing delay and rate of transmission at VoIP service call occurred in connection quality. And other parameters which point to quality features like loss, delay, phase jitter, bit error ratio and noise at transmission phase occurred in transmission quality.

Table3.Quality features classification

Classification	Contents	Quality Item
Connection Quality	Quality related to Call setup, Call retain, Call clear	-Dial Tone Delay -Post Dialing Delay -Rate of Transmission
Transmission Quality	-Data Transmission Quality after call setup -Clarity of call	-Loss -Noise -Phase Jitter -Echo

Table4.Connection Quality Measurement Process

Dial tone delay	-Measuring unit for dial tone delay is set for 1/1000sec. -Measure dial tone delay time from switch connected to measuring device of transmitting side -Measure time from point of off-hook of measuring unit of transmitting side (subscribers' aspect) to hear the dial

ISSN 2250-3153

	tone
Post dialing delay	-Measuring unit for dial to noise ratio is dB -Measure noise ratio and signal arrived at receiving side by transmitting test call of standard frequency from transmitting side -Measuring unit is 1/100sec for post dialing delay -Measuring section is set from transmitting side's measuring unit to receiving sides. -Measure time from the point of ending of dialing of transmitting measuring unit to tone receiving at receiving side
Rate of transmission	Measuring unit is % for the rate of transmission -Measuring section is from measuring unit of transmitting side to measuring unit of receiving side -Rate is calculated by the ratio between call success and total call made from measuring unit of transmitting side to measuring unit of receiving side

II. Table5. Transmission Quality Measurement Process

Rate of transmission	-Measuring unit is % for the rate of transmission. -Measuring section is from measuring unit of transmitting side to measuring unit of receiving side Rate is calculated by the ratio between call success and total call made from measuring unit of transmitting side to measuring unit of receiving side.
Loss	-Measurement unit for loss is in dB. -Internal impedance of measuring unit of both sides is set for 600. -Standard signal is transmitted from receiving side (transmitting side) and receiving level is measured at transmitting side (receiving side). -Measured value is the difference of transmitting level [dB] and receiving level [dB].
Noise	-Measurement unit for noise is in dB. -Set impedance of both measuring unit to be 600. -Apply CCITTO.41 standard's Psophometric Weighting filter method for measuring noise.
Phase jitter	-Measurement unit for phase jitter is in degree. -Measure phase jitter at signal level with frequency of 4~300Hz (LF+ Standard Filter).
Echo	-Measurement unit for echo is in dB. -Measure voice echoed back to the speaker. -Measure echo delay and echo loss -Measurement unit for echo is 1/1000sec
Bit Error ratio	-Measurement unit for bit error ratio is in error bit -Measure error bit by transmitting test pattern with Connection between transmitting side and receiving side retained -Range from 2400bps to 14,400bps can be used to measure when using internal modem -Measuring time is set for 15 minutes and it can be changed.

VII. TESTING QOS & PERFORMANCE TESTS

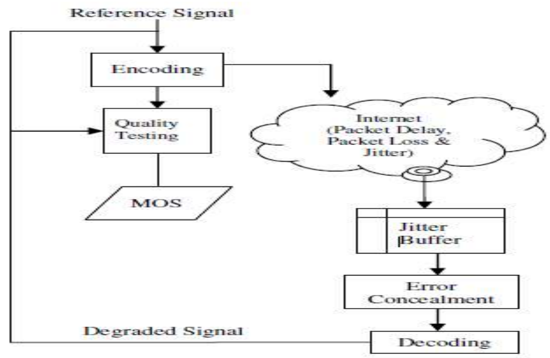

Fig 2: QoS Model Diagram

Assumed that two computers were utilized in different cities of Brazil (Ananindeua, Terminal A, and Belem, terminal B), both using technology ADSL (operator) to 256 Kbps and the ethereal to generate the grid of sending and receiving data and being measured at A and B with a pattern lecture utilization packets. We separately tests for each system and analyzed with the sniffer in each situation and saw the behavior and function of each system as meeting the common traffic between the internet users, observing the packing characteristic and the way how the protocols work during the conversation, besides measuring the audio that could qualify the understanding of the sent texts (talked). Many loss packets analysis can be calculated based on specific network information as follows

A. SUBJECTIVE LISTENING TESTS

The proposed work judges the worth or quality measures for audio in CODEC G.711, G.729, AMR and GSM-FR. The challenge for the audio CODECs is the output quality. This work will use the subjective method for evaluating audio quality where the classical objective measures are considered insufficient. Subjective listening tests are considered most accurate and most reliable method for assessing the quality of CODEC and widely supported by the ITU. The combined efforts of ITU and the three telecommunications companies that specialize in the evaluation of audio quality: Opticom, Psytechnics, Swissqual have been working With direction to and developing an “objectively subjective” solution for live network monitoring which is based on the customer experience to provide the industry with customer information about the performance of their real-time systems. The live audio tests will be conducted using people as subjects to measure the pre-recorded audio samples. An automated system will use the pre-recorded samples and play them out to the live subjects, then rate the perceived quality of the speech on a scale 1 to 5. The rating scale will be based on the MOS scoring system where score of 1 is being lowest and 5 is being the highest score. The audio samples will contain recorded sentences with female and male voices in 8 different languages, each in duration of 10 seconds. To assure reliability of the measured data, the live testing will include a wide number of human subjects, as well as a diverse group with both genders and a wide range of age groups. The proposed number of subjects will be in the total of 241, female and male each testing 10 samples, and the age spanning from 18 to 50+. This will cover all possible combinations of 2401 scenarios to test and the new VoIP-eM equation proposed. We choose several different languages, since research has shown a variation between the languages in scoring the voice quality, and following the ITU Recommendations P.800 the testing should be conducted in the subject's native language. Conducting the listening tests from the public domain and targeting a large number of human subjects certainly may attract people from non-English speaking background. The live measurements perform with different background conditions and using different CODEC types. The automated system will play out VoIP voice-recording samples using a relative testing approach, where a reference signal is played first and followed by the degraded file. The subjects will be asked to rate the perceived quality using the MOS, ACR, and CCR scaling system. The Absolute Category Rating (ACR) scale is recommended by the ITU P.800/830 as the most suitable. The MOS scoring will be used in evaluating the smaller impairments, such as echoes. When it measures the subjective quality of a given CODEC, then it may be allow considering a more detailed scale when conducting subjective listening tests. This gives an example of a world satellite communication system for a 16-bit rate CODEC where a more precise scale has been used. The CCR (Comparison Category Rating) is described in ITU-T Recommendations P.800/P.830 and is indicating to be used for evaluation of large deterioration. The testing will follows the standard ITU recommendations. Method of conducting a subjective test experiment is described in detail in the ITU-T Recommendation P.830. The results from live tests will be collected and analyzed, and the new coding algorithm can be*A.* derived based on the measured impairments.

B. Speech coding/decoding schemes

In this paper we chose four types of speech coding algorithms: CODEC G.711, G.729, AMR and GSM-FR for their suitability in heterogeneous networks. ITU CODEC G.711 uses Pulse Code Modulation (PCM) coding format and it operates at a higher rate of 64 kb/s. The Packet-Loss Concealment (PLC) can be added to the G.711 to decrease the loss in the transmission. ITU CODEC G.729 uses a conjugate-structure algebraic code worked up linear-prediction (CS-ACELP) coding format and operates at 8 kb/s. The error concealment is already included in G.729 to reduce the impact of the lost frames. AMR (Adaptive Multi-Rate) is an ETSI speech-coding scheme which is mainly used in wireless VoIP networks and is compulsory for UMTS, but is also suitable for wired VoIP networks. AMR codec uses Algebraic Code Excited Linear Prediction (ACELP) coding format and it operates at a multi-rate between 4.7 and 12.2 kb/s, and it includes packet loss concealment. GSM-FR CODEC uses RPE-LTP coding format and it operates at 13 kb/s. GSM-FR is designed for the use in UMTS network. The mention signals as recommended in the ITU-T Recommendations P.800/P.830 will be used in the experimental testing.

C. E-Model

E-Model is used for significant testing of QoS. The E-Model, defined in ITU-T G.107 has computational tools that measure the transmission parameters which affect the conversation quality. The output of the E-Model is a transmission-rating factor R that combines with all the deterioration parameters into a total value. The transmission factor R equation consists of the base quality value Ro, the impairment (harm) factors Is, Id and Ie and of an prospect or advantage factor A. The formula for the transmission factor R is as follows: R = Ro – Is – Id – Ie +A where, Ro represents the signal to noise ratio at the 0dB point Is, is the parameter for simultaneous impairments like excessive loudness ratings, uncomfortable side tone levels and simple quantization distortions Id contains of impairments due to echoes and delays Ie represents the impairments from transmission equipment A is a factor that adjusts the quality value the transmission factor R is shown at a scale from 0 to 100 but is typically ranging between 50 and 90. The conversion from R to MOS is shown in Table 6. In subjective testing, the standard formula to calculate MOS based on the number of participants (represented by "n") that rated the quality is as follows: MOS = [(n*5) + (n*4) + (n*3) + (n*2) + (n*1)] / 100 (2)

Table6: R-Factor and MOS Value

R Factor	User Satisfaction	MOS
90 – 100	Best	4.34 – 4.50
80 – 90	High	4.03 – 4.34
70 – 80	Medium	3.60 – 4.03
60 – 70	Low	3.10 – 3.60
50 – 60	Poor	2.58 – 3.10

The principle of E-Model is based on the premise that transmission impairments can be transformed into psychological factors and that psychological factors on the psychological scale are linearly additive.

D. Suitability to VoIP

E-Model is good for an accounting for the impairments currently the E-model provides only limited and inflexible values for the Ie parameters, so the writers proposed a new formula for calculating Ie that depends on packet loss. Moreover they extended the current formula for calculating the R-factor by adding a new parameter Ij, the jitter impairment factor. This work will aim to develop a novel QoS method that is designed for the new VoIP systems, and propose experimental measures to achieve this.

Table 7: VoIP Impairments Analysis

Impairments	Coverage	VoIP Applicability	Importance
Impairments due to packet loss	Ie *random not for all CODEC*	Applicable	High
Impairments due to packet delay	Id *not for all CODEC*	Applicable	High
talker echo	Id	not applicable	Low
Loudness rating	Is	Not applicable	Low
Quantization distortion	Is	Not applicable	Low
side tone rating	Is	Not applicable	Low

ISSN 2250-3153

errors due to coding/decoding		Not applicable	Low
Quality Adjustment	A	Not applicable	NIL

VIII. SOFTWARE QUEUEING MECHANISM TO SIMPLIFY THE QOS SERVICES

All voice call traffic has been placed into QoS classes, based on QoS requirements, here we need to provide bandwidth guarantees and priority servicing through an intelligent output queueing mechanism In order to provide a preferred level of service for high-priority traffic, some form of software queuing must be used. Software queuing techniques can include:

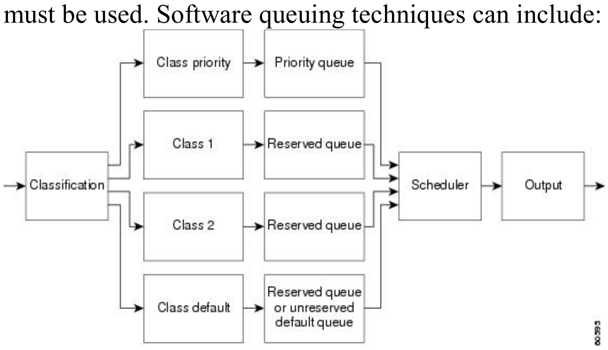

Fig 3: Queueing Mechanism

A. *First-In First-Out (FIFO) (default)*

B. *Priority Queuing (PQ)*

C. *Custom Queuing (CQ)*

D. *Weighted Fair Queuing (WFQ)*

E. *Class-Based Weighted Fair Queuing (CBWFQ)*

F. *Low-Latency Queuing (LLQ)*

Software queuing normally employs multiple queues, and each is assigned a specific *priority*. Traffic can then be assigned to these queues, using access lists or based on classification. Traffic from a higher-priority queue is serviced before the traffic from a lower-priority queue.

Table 8 Software Queueing Mechanism

Software Queueing Mechanism	Description	Benefits	Limitations
FIFO	Packets arrive and leave the queue in exactly the same order.	Simple configuration and fast operation.	No priority servicing or bandwidth guarantees are possible.
WFQ	A hashing algorithm places flows into separate queues where weights are used to determine how many packets are serviced at a time. You define weights by setting IP Precedence and DSCP values.	Simple configuration. Default on links less than 2 Mbps.	No priority servicing or bandwidth guarantees are possible.
Custom Queueing (CQ)	Traffic is classified into multiple queues with configurable queue limits. The queue limits are calculated based on average packet size, maximum transmission unit (MTU), and the percentage of bandwidth to be allocated. Queue limits (in number of bytes) are dequeued for each queue, therefore providing the allocated bandwidth statistically.	Has been available for a few years and allows approximate bandwidth allocation for different queues.	No priority servicing is possible. Bandwidth guarantees are approximate, and there are a limited number of queues. Configuration is relatively difficult.
Priority Queueing (PQ)	Traffic is classified into high, medium, normal, and low priority queues. The high priority traffic is serviced first, then medium priority traffic, followed by normal and low priority traffic.	Has been available for a few years and provides priority servicing.	Higher priority traffic can starve the lower priority queues of bandwidth. No bandwidth guarantees are possible.
Class-Based WFQ (CBWFQ)	MQC is used to classify traffic. Classified traffic is placed into reserved bandwidth queues or a default unreserved queue. A scheduler services the queues based on weights so that the bandwidth guarantees are honored.	Similar to LLQ except that there is no priority queue. Simple configuration and ability to provide bandwidth guarantees.	No priority servicing is possible.
Priority Queue WFQ (PQ-WFQ, also called IP RTP Priority)	A single interface command is used to provide priority servicing to all UDP packets destined to even port numbers within a specified range.	Simple, one command configuration. Provides priority servicing to RTP packets.	All other traffic is treated with WFQ. RTCP traffic is not prioritized. No guaranteed bandwidth capability.
LLQ (Previously called PQ-CBWFQ)	MQC is used to classify traffic. Classified traffic is placed into a priority queue, reserved bandwidth queues, or a default unreserved queue. A scheduler services the queues based on weights so that the priority traffic is sent first (up to a certain policed limit during congestion) and the bandwidth guarantees are met.	Simple configuration. Ability to provide priority to multiple classes of traffic and give upper bounds on priority bandwidth utilization. You can also configure bandwidth guaranteed classes and a default class.	No mechanism for providing multiple levels of priority yet—all priority traffic is sent through the same priority queue. Separate priority classes can have separate upper priority bandwidth bounds during congestion, but sharing of priority queue between applications may introduce jitter.

IX. CONCLUSION AND FUTURE WORKS

In this paper we proposed VoIP Quality of Service as a absorbing development path ahead, and develop a naturalistic Design framework for this technology area, and verify our design parameters for QoS successfully. QoS is a important issue for VoIP industry and it competes with other technologies. The Key issues include the linearity of the existing models, the relevance of all impairment factors and the validity of the model in the face of background, noise and language types, and the treatment of errors in bursty traffic conditions. The key research develops a new method for QoS determination and dynamic control of the network that is applicable to VoIP industry. This followed on our established experimental means to conduct live subjective testing of VoIP quality. The adequate measurement of QoS is important to the VoIP industry for various reasons, mainly for the cost reduction and high user satisfaction. Inadequate quality monitoring can beginning to undesired problems. For example too high QoS rating is costly, and sudden changes in the network may put down QoS to the level that bothers clients. The network reporting mechanisms will be used for detecting immediate problems, and forecasting trends. In this paper we calculated delay, bytes tax, packets per second, packet loss, utilizing Ethereal to capture the packet to be analyzed, explained all the established communication between the terminal where is observed each packet being sent from one terminal to the other. After the analysis of the utilized protocols behavior for each system, the results were that the Skype is the best software for voice over IP conversation, because it has more voice quality with less phonetic loss, since the transmitted information, with the bigger broadband among the analyzed software, at 69Kbps, through the TCP protocol, allows the quality VoIP messages

ISSN 2250-3153

delivering. Our future work will consist of integrating the new algorithmic approach into a dynamic monitoring system for VoIP QoS, The new VoIP system will be integrated into the LINUX based CORBA middleware for the industry solution, work on QoS Protection Mechanism of VoIP Systems, New routing methods over better links for an active network control based on our new quality estimator, new priority and path selection mechanism, Analysis the requirements of how VoIP will provide QoS guarantees also improve the quality to VoIP message delivering, Fragmentation and interleaving actually VoIP transmissions are extremely delay-sensitive.

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