Queuing Impact on VoIP Quality of Service (QoS) in Telecom Networks

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***Abstract* - Today's Internet only provides Best Effort Service. Traffic is processed as quickly as possible, but there is no guarantee of timelines or actual delivery. With the rapid transformation of the Internet into a commercial infrastructure, demands for service quality have rapidly developed. People are now very much dependent on various network services like VOIP, Videoconferencing and File Transfer. Voice over Internet Protocol is a service from the Internet services that allows users to communicate with each other. Quality of Service is very sensitive to delay so that Voice over Internet Protocol needs it. The objective of this research is to assess the effect of different queuing algorithms within the router on Voice over Internet Protocol Quality of Service. Simulation tool OPNET Modeler is used to implement the proposed network (Voice over Internet Protocol Network). The proposed network consists of two logical subnets one for clients and one for servers located at two different countries around the world, the core network connecting the two subnets is composed of 10 routers connected in half mesh topology in order to simulate the communications between two locations as a long distance and analyze Voice over Internet Protocol Quality of Service through measuring the major factors that affect the Quality of Service for Voice over Internet Protocol according to international telecommunication union standards such as: delay, jitter, end to end delay and packet loss. Different types of Traffic Management systems are used in those services. Queuing is one of the very vital mechanisms in traffic management system. Each router in the network must implement some queuing discipline that governs how packets are buffered while waiting to be transmitted. This paper gives a comparative analysis of six queuing systems FIFO, Priority queueing, Weight Fair Queuing, MWRR, DWRR and MDRR.**

Keywords - Queuing Impac, VoIP, Quality of Service (QoS, Telecom Networks

I. INTRODUCTION

VoIP stands for voice over internet protocol. (VOIP) is a technology for communicating using "internet protocol" instead of traditional analog systems. Some VOIP services need only a regular phone connection, while others allow you to make telephone calls using an internet connection instead. Some VOIP services may allow you only to call other people using the same service, but others may allow you to call any telephone number - including local, long distance, wireless and international numbers [1].

VoIP is a technique that encodes voice to the low rates and route the relatively low bandwidth signals as packetized "data", over dedicated transmission facilities or the internet using the internet protocol, The long run benefits of VoIP include support for multimedia and multileveled applications, something which today's telephone system can't compete with, An integrated voice/data network allows more standardization and reduces total equipment needs, there can be a real savings in long distance Telephone costs which is extremely important to most companies, particularly those with international markets.

In this paper study the effect of different queuing algorithms within the router on Voice over Internet Protocol Quality of Service. In Sec. II we present a brief overview of VoIP Quality of Service parameters. In Sec. III the Queuing is presented. In Sec. IV Network Design and Configuration and The Average results of different queuing disciplines on VOIP Quality are presented.

The conclusion and future work in addition to suggestions for further refinements are discussed in Sec. V.

II. QUALITY OF SERVICE

QoS can be defined as the ability of the network to support good services in order to accept good customers. In other words, QoS measures to the degree of user satisfactions and network performance. Applications like FTP, HTTP, video conferencing and e-mail are not sensitive to delay of transmitted information assess QoS in important problems, while other applications like voice and video are more sensitive to loss, delay and jitter of the information. Therefore, QoS of VoIP is an import interest to ensure that voice packets are not delayed or lost while be transmitted over the network. VoIP QoS is measured according ITU recommendations based on different parameters like (delay, jitter, and packet loss), these parameters can be changed and controlled within the acceptable range to improved VoIP QoS [8].

End to End Delay. [8] In general voice is a delay-sensitive application while most data applications are not, End-to-end delay is defined as the time it taken to deliver a packet from the sender to the receiver. Objective is end-to-end delay < 150ms when end-to-end delay reaches about 150 milliseconds, participants in a telephone conversation begin to notice its effects. For real-time voice conversation (VoIP), the end-to-end delays between 150 and 400 msec is acceptable but are not perfect. The receiver of a VoIP call will define certain threshold (e.g. 400 msec); any packets that are delayed more than that threshold will be discarded. End-to-end delay above 400 milliseconds can make normal conversations impossible. Delay affects the quality of a conversation without affecting the actual sound of the voice signal – delay does not introduce noise or distortion into the voice channel.

Jitter (Delay Variability). [6] Jitter is defined as a variation in the delay of received packets. At the sending side, packets are sent in a continuous stream with the packets spaced evenly apart. Due to network congestion, improper queuing, or configuration errors, this steady stream can become lumpy, or the delay between each packet can vary instead of remaining constant. This diagram illustrates how a steady stream of packets is handled.

Packet Loss [7] A packet loss is defined as the number of packets that are lost during transmission process inside the network within a defined time period when a device (router, switch, and link) is overloaded and cannot accept any incoming data at a given moment. Moreover, packet loss may occur due to: network congestion, lower layers errors, network element failures or due to the end application errors.

Throughput [8] The amount of data that can actually be transmitted over the communication channel is called throughput. It is used to estimate the efficiency of net-work. The ratio between the quantity of information and the sum of user data, control data and retransmitted data if error is concluded as throughput of a network. Throughput refers to how much data can be transferred from source to destination in a given amount of time. It depends upon the bandwidth also. Throughput is a measure of data rate (bits per second) generated by the application.

MOS or Mean Opinion Score gives VoIP testing a number value as an indication of the perceived quality of received voice after being transmitted and compressed using codecs. This measurement is the result of underlying network attributes that act upon data flow and is useful in predicting call quality and is a good VoIP test tool in determining issues that can affect your VoIP quality and your conversations. Testing the quality of VoIP has become easier and services have greatly improved over the last few years due to both the providers becoming more reliable and the ISPs offering better connections. This advancement in the quality of services has helped increase the number of VoIP subscribers, but occasionally issues affecting voice quality do arise and being able to test your VoIP and identify these instances can be helpful in addressing them. Having a metric to measure changes or degradation in the quality of the voice/VoIP connection after testing can help identify problems. VoIP calls often are in the 3.5 to 4.2 MOS range.

III. QUEUING AND CONGESTION

QoS can be defined as the ability of the network to support good services in order to accept good customers. In other words, QoS measures to the degree of user satisfactions and network performance. Applications like FTP, HTTP, video conferencing and e-mail are not sensitive to delay of transmitted information assess QoS in important problems, while other applications like voice and video are more sensitive to loss, delay and jitter of the information. Therefore, QoS of VoIP is an import interest to ensure that voice packets are not delayed or lost while be transmitted over the network. VoIP QoS is measured according ITU recommendations based on different parameters like (delay, jitter, and packet loss), these parameters can be changed and controlled within the acceptable

A. *Queuing Algorithms*

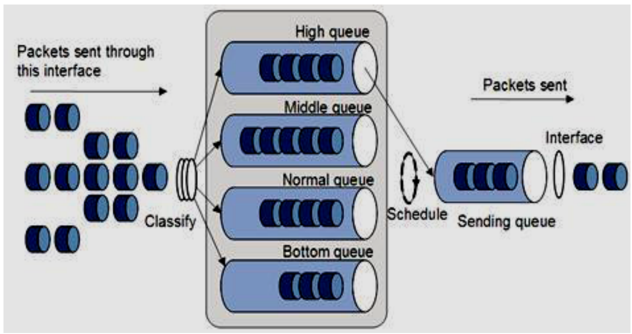

Fig. 1

The biggest problems in a network are related to the allocation of network resources, as buffers and link bandwidth, to different users. A limited amount of re-sources has to be shared among many different competing traffic flows in an efficient way in order to maximize the performance and the use of the network resources. The behavior of routers in terms of packet handling can be controlled to achieve different kind of services. Various queuing Algorithms can be used to control which packets get transmitted and which packets get dropped.

1) *First In First Out (FIFO) queuing:* FIFO queuing is also referred to as First Come First Serve (FCFS) queuing This expression describes the principle of a queue or first-come first serve behavior: what comes in first is handled first, what comes in next waits until the first is finished etc. Thus it is analogous to the behavior of persons “standing in a line” or “Queue” where the persons leave the queue in the order they arrive. First In First out (FIFO) is the most basic queuing discipline. In FIFO queuing all packets are treated equally by placing them into a single queue, then servicing them in the same order they were placed in the queue. An exception here happened if a packet arrives and the queue is full, then the router ignores that packet at any conditions.

DOI 10.5013/IJSSST.a.20.05.14 ISSN: 1473-804x online, 1473-8031 print

Although a single FIFO queue seems to provide no QoS features at all, it actually does affect drop, delay, and jitter. Because there is only one queue, the router need not classify traffic to place it into different queues and router need not worry about how to decide from which queue it should take the next packet—there is only one choice. Due to this single queue uses FIFO logic, the router need not reorder the packets inside the queue. With a longer queue, however, the average delay increases, because packets may be enqueued behind a larger number of other packets[3].

2) Priority Queuing (PQ)[10]: For applications that are sensitive to delay and that are not able to handle packet loss, In most cases when the average delay and average jitter increases Priority Queuing assigns multiple queues to a network interface with each queue being given a priority level. A queue with higher priority is processed earlier than a queue with lower priority. Priority Queuing has four preconfigured queues, high medium, normal and low priority queue. If packets arrive in the high queue then priority queuing drops everything its doing in order to transmit those packets, and the packets in other queue is again empty. This mechanism is good for important traffic, but can lead to queue starvation. The only problem with these packets is that has lower-priority in queue.

3) Weighted Fair Queuing (WFQ): Weighted fair queuing (WFQ) is scheduling mechanism in which the queue servicing is based on bits instead of packets. In WFQ, each queue or flow is allocated a weight that is a proportion of the interface rate or the shaping rate. WFQ is aware of packet sizes and can thus support variable-sized packets. The benefit is that sessions with big packets do not get more scheduling time than sessions with smaller packets, because effectively the focus of WFQ is bits and not packets. So there is no unfairness in the scheduling for sessions with smaller packet sizes. With WFQ, each queue is scheduled based on a computation performed on the bits of each packet at the head of the queue. Because the traffic computation is done based on stream of bits and not of packets, and because what the router receives and transmits are indeed packets, WFQ implicitly increases complexity [2].

4) Modified weighted round robin (MWRR): Weighted round robin (WRR) assign a weight to each connection then the connections served according to their weights. The main problem of WRR is that when the traffic has a variable packet size, WRR provides incorrect percentage of bandwidth allocation. WRR accomplishes this by al-lowing several packets to be removed from a queue each time that queue receives a scheduling turn.

5) Deficit Weighted Round Robin (DWRR): Deficit weighted round robin (DWRR) is the same as DRR but adds a weight variable for each queue and the Q (Quantum) value depends on the weight value. With WRR for each scheduling turn, the number of packets that are granted service is based on a weight that reflects the bandwidth allocation for the queue. bandwidth allocation can be unfair when the average packet sizes are different between the queues and their flows. This behavior can result in service degradation for queues with smaller av-erage packet sizes. Deficit Round Robin (DRR), or Deficit Weighted Round Robin (DWRR), is a modified weighted round-robin scheduling discipline that addresses the limitations of WRR. Deficit algorithms are able to handle packets of variable size without knowing the mean size. A maximum packet size number is subtracted from the packet length, and packets that exceed that number are held back until the next scheduling turn[2].

6) Modified Deficit Round Robin (MDRR): Another mod-ification on DRR named modified deficit round robin (MDRR) works in the same way as DRR but a priority parameter is added for each queue to contribute to queue selection, it is a queue priority. In MDRR, all queues are serviced in a round-robin fashion with the exception of the low-latency queue. LLQ is based on Class-Based WFQ algorithm, which can be used for multimedia data transmission. This way the advantages of PQ and WFQ are combined and the disadvantages of these algorithms are reduced [4].

IV. NETWORK DESIGN AND CONFIGURATION

The simulation tool adopted in this research is Optimized Network Engineering Tools (OPNET) version 14.5. OPNET is an object-orientated simulation tool for making network modeling and QoS analysis of simulation of network communication, network devices and protocols. OPNET Modeler has a vast number of models for network elements, and it has many different real-life network configuration capabilities. These make real-life network environment simulations in OPNET very close to reality and provide full phases of a study.

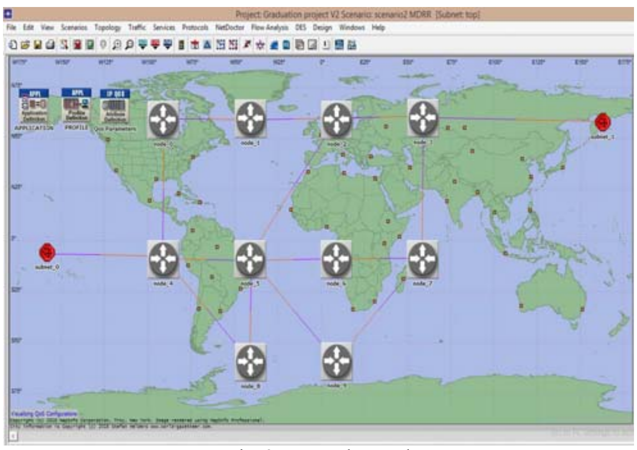

Fig. 2. Network Topology.

DOI 10.5013/IJSSST.a.20.05.14 ISSN: 1473-804x online, 1473-8031 print

OPNET also includes features such as comprehensive library of net-work protocols and models, user friendly GUI (Graphical User Interface), Web report is feature allows you to organize and distribute the results of your simulations in form graphical results and statistics. OPNET doesn't have any programming knowledge so that it's easy to use and to deal with for any person. Automatic simulation generation OPNET models can be compiled into executable code.

A) The bellow configurations applied in the OPNET Modeler and simulated to get results:

1. The ten routers are connected (Half mesh topology) with PPP_DS1link.
2. The Work stations and the servers are connected with routers with 100Base_T links.
3. In the field of FTP application "Medium Load" has been selected to ToS is assigned.
4. In the field of Video Application "High Resolution Video" has been selected for Video Conferencing, to ToS is assigned.
5. In field of Voice application Interactive Voice (6) is assigned.
6. In the field of HTTP application "Searching" has been selected, (2) to ToS is assigned.

B) Simulation Results

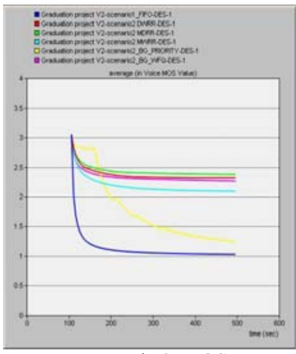

Fig.3. MOS

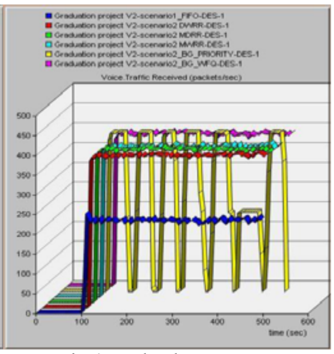

Fig.4. Packet loss

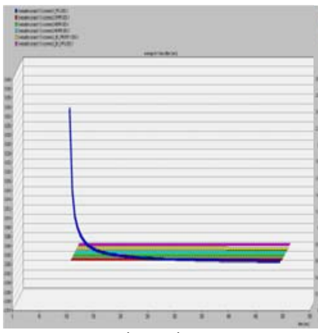

Fig.5. Jitter

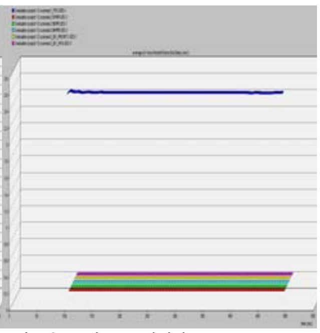

Fig.6. End to end delay

TABLE I. AVERAGE RESULTS OF DIFFERENT QUEUING DICIPLINES ON VOIP QUALITY

Queuing	*End to End Delay*	*Jitter*	*Packet Loss atio*	*MOS*
FIFO	2.61	0.0003	42.8%	1.02
PQ	0.201	0	36%	1.25
WFQ	0.202	0	4.1%	2.26
MWRR	0.197	0	5.2%	2.1
DWRR	0.192	0	3.8%	2.32
MDRR	0.192	0	3.4%	2.38

V. SUMMARY AND CONCLUSION

We presented a short overview of congestion management algorithms used by routers. The simulation environment was provided by the OPNET application. We used a generalized, extendable and factual network topology. Congestion can occur at any point in network but particularly at points of speed mismatches and traffic aggregation. Six queuing algorithms were assessed to manage congestion: FIFO, PQ, WFQ, MWRR, DWRR, and MDRR. The Three most efficient algorithms (MDRR, WFQ, and DWRR) were compared in respect of the delay of voice packets, jitter, packet loss and throughput. The reason of choosing the voice packets is that they are very sensitive to delay and jitter. The voice packet is not very long but it is highly sensitive to delay and jitter due to the nature of service. The Priority of voice in QoS configuration is therefore the highest. If its priority is brought down, the end user will receive low grade and broken voice. It is obvious FIFO has the highest value for jitter, end to end delay, and has the lowest MOS value then the worst performance happened when FIFO algorithm is used. But according to the simulation it is already proven that a modernized format of fair queuing WFQ (Weighted Fair Queue) can perform better because it supports variable sized packets. So it can be said with confidence that user traffic stream like voice, video & data can be easily transferred with its efficient level performance by using Weighted Fair Queue algorithm in routers where the voice, video and data streams are routed to go to their desired destination. But WFQ one disadvantage is that it increases complexity because it is complicated to be implemented. After enhancement by using Round Robbin Algorithms, MDRR delivered much better performance for voice packets than WFQ due to the addition of a priority parameter for each queue to contribute to queue selection to form a queue priority.

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DOI 10.5013/IJSSST.a.20.05.14 ISSN: 1473-804x online, 1473-8031 print