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Measuring the Queuing Delay and Reducing the Delay using Priority Queue in VoIP packets

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Abstract: In order to compete effectively in present telecommunication market, Service providers and enterprises are turning towards Voice over Internet Protocol (IP) technology. Initially the goal of using Voice over Internet Protocol (VoIP) in enterprises was a better utilization of internal packet networks an avoiding long distance toll charges. Queuing Delay is an important measures of Quality of Service (QoS) particularly for voice traffic in network environment. The objective of this research paper is to measure the delay using synchronize clock between sender and receiver. This is achieved using Network Time Protocol (NTP).And also to reduce the delay using the Priority Queue algorithm.

Key Words: Quality of Service, Queuing Delay, VoIP, Priority Queue, Network Time Protocol

I. INTRODUCTION

The purpose of this research is to identify queuing delay on Voice over Internet Protocol (VoIP) and finding a suitable mechanism to reduce the delay and improve the quality of service (QoS) of VoIP. VoIP is a kind replacement for the basic telephones. It is need to receive the same quality of voice transmission to the receiver through the internet like other real time applications do. This VoIP service is allowing users to communicate with each other over the internet services with low cost. VoIP has extremely bandwidth and delay sensitivity. There are different algorithms that can be used in VoIP to increase their quality.

The Research finds out how the time will get affected to the VoIP queuing delay and how it is going to reduce the delay of queuing over the network. . In many ways it is best to use Priority Queuing (PQ) for the VoIP networks in order to reduce the timing delays. PQ objective is to classify network VoIP traffic into high, medium, normal, and low Priority Queues and servicing the high priority first, then the medium priority traffic followed by normal and low priority traffics.

The first section of the research paper discuss background and related works and second section discuss about Quality of Service which contains two sub areas related to measuring delay and priority queue. The approach is discussed in third section and finally conclusion and related works.

II. BACKGROUND AND RELATED WORK

According to the referred research papers research group found many related work gaining many comparisons between each components that related to the VoIP. In [01] it is showing how to measure the packet delay on VoIP without clock synchronization. Richard T. (2009)used a method called sync and sense periodic streaming (SSPS) over protocols [01]. Going through with [02] research paper that was published by Hack, A., (1986) it shows decomposition solution with using a system called dynamic locking in queuing network model. Using the following [03] research paper of Tanvi Sharma (2014),it implements how to control the queuing delay on VoIP by using Active Queuing Management and Control Delay algorithm.

Dr. Hussein A., Hawraa J., Dr. Adnan H., (2013) shows in their research paper [04] how use OPNET simulators to measure the Quality of Service (QoS) on VoIP in different queuing algorithms. Research paper [05] of Michal H., Stanislav K. (2012) is showing what the variations of delaying packets across the network are. It [05] shows there are three types of variations coursed to delay packets over the network. Mohammed M. (2014) is targeting on research paper [06] to conclude by using some mathematical approaches to measure the queuing delay on VoIP. He is using two mathematical operations called interpolation and integration to produce a uninterrupted function and it is showing different times in delay.

Rashed M.M.G., Kabir M. (2010) is analyzing three main queuing systems (FIFO, PQ, WFQ) on his research paper [07] to handle packets over the network. And it [07] says the Priority Queuing is the best queuing system that can be used. Research paper [08] published by Voznak M is showing how to avoid the jitter and increase the quality in VoIP over the low speed links. In research paper [09] published by Anton K., (2002) describes by focusing to increase the performance of VoIP by using different techniques.

Kathleen Nichols, Van Jacobson, (2012) is introducing on their research paper [10] some buffer bloat solutions that they have experienced on VoIP. Research paper [11] by Md. Zahirul Islam is implementing a hybrid queuing mechanism to reduce the queuing traffic delay of VoIP over the network. In [11] they using class-based weighted fair queuing (CBWFQ) for Hybrid queuing. In research paper [12] published by Sanjay Dahiya, Deepak Kumar, Chaman Verma, (2015) is showing a methodology to measure the queuing delay overall network and how to reduce the delay by using protocol called LEDBAT. In the final research paper [13] published by H. Bulut, S. Pallickara, G. Fox, is implementing a Network Timing Protocol (NTP) by using open source middleware system.

III. QUALITY OF SERVICE (QoS)

Quality of service (QoS) is a major issue in voice over Internet protocol (VOIP) implementations. The problem is how to prevent packet traffic for a voice or other media connection not being delayed or dropped from other low priority traffic. VOIP can be guaranteed of high-quality voice transmission only if priority is given to the voice packets for both the signaling and audio channel over the other kinds of network traffic. When VOIP is deployed for the user to receive an acceptable level of voice quality VOIP traffic must be assured certain bandwidth, latency and jitter requirements. QoS makes sure that the VOIP packets get the special treatment that they require. In general QoS delivers better network service by providing the following features such as:

- Supportive devoted bandwidth;
- Improving loss characteristics;
- Avoiding any management network congestion;
- Modeling network traffic;
- Setting traffic priorities through the network.

Queuing delay is a major issue in VoIP QoS. When more packets are sent out at a given interval more than the interface can handle, queuing delay occurs. This can be measured and reduced by use of priority queue.

a. *Measuring Delay*

One of the critical parameter in VoIP is network delay since it determines the call quality. The delay that a data packet suffers in the trip from source to destination. One-way delay of the VoIP packet can be measured, having a synchronized clock between the sender and receiver. GPS (Global Positioning System), NTP (Network Time Protocol) or CDMA (Code Division Multiple Access) are some ways the clock synchronization can be achieved. One way delay variation is defined as the difference in One-way delay experienced by two packets of the same length, where the definition of one way delay variation among any two packets a and b (a ≠ b) is the variance of their One-way delay. Figure 1 devotes how synchronization works.

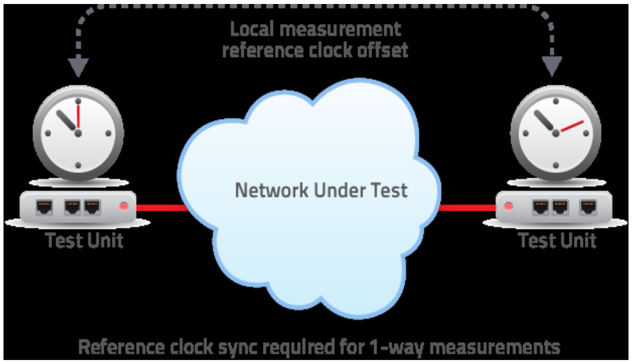

Figure 1: Clock Synchronization

Source: One-way Delay measurement technique, Accedin Networks, <Available: http://accedian.com/wp-content/uploads/2015/05/ One-WayDelayMeasurementTechniques-AccedianWhitePaper.pdf>

b. *Priority Queue*

Priority queuing allows to prioritize certain traffic flows (such as latency-sensitive traffic for example; voice and video) rather than other traffic. Priority queuing uses a priority queue on an interface, while all other traffic drives into the "best effort" queue, as the queues are not of infinite size, it may lead to overflow of packets. When a queue is full, any extra packets cannot get into the queue and are dropped; this is known as tail drop. To avoid having the queue fill up, can increase the queue buffer size. Also it can fine-tune the maximum number of packets allowed into the transmit queue. These options allows to control the robustness and latency of the priority queuing. Packets in the queue are always transmitted before packets in the best effort queue.

IV. OUR APPROACH

a. Network Time Protocol

The research group will be using Network Timing Protocol (NTP) to measure the one-way delay, since NTP messages received from time servers are filtered and selected using advanced algorithms.

First step is to get samples form NTP servers by sending NTP message and wait for response. It is also required to set a timeout for these NTP messages since these messages may lost in the network. Servers reply back with NtpInfo object which includes NTP parameter values. Once the samples from NTP server collected, checks for timestamps to validate the NTP message received by use of TP filtering algorithm. Then finally by use of selection algorithms the offsets are computed.

b. Priority Queue Configuration

Firstly it is necessary to enable standard priority queuing for traffic on physical interface, and then to create the priority queue on each interface as well. Each physical interface consists of two queues; one for priority traffic, and the other for all other traffic.

When configuring priority queue there are few steps to be followed;

Step 1: Create the priority queue.

hostname(config)#priority-queue interface_name

Entering the above command a priority queue can be created, where the *interface_name* argument states the physical interface name on which needs to be configured the priority queue.

Note: It is not necessary to create a priority queue for an interface with high bandwidth.

Step 2: Define size of the queue

The queues are not of infinite size, the default queue limit is 1024 packets. So it may lead to fill and overflow of packets. When a queue is occupied, any other extra packets cannot get into the queue and are dropped as mentioned above (tail drop). To avoid having the queue fill up, by use of the below *queue-limit* command the queue buffer size can be increased.

Hostname (config-priority-queue)# queue-limit number_of_packets

The range of values of the upper limit for the *queue-limit* command is determined dynamically at run time. This limit can be viewed by entering "*queue-limit?*" in the command line. The key determinants are the memory needed to maintain the queues and the memory existing on the device.

Step 3: Specify depth of queue

hostname(config-priority-queue)# tx-ring-limit number_of_packets

Using the above command depth of the queue can be specified, the default tx-ring-limit is 128 packets. This command sets the determined number of low-latency or normal priority packets allowed into the Ethernet transmit driver before the driver drives back to the queues on the interface allows to buffer packets until the congestion clears. This setting make sure that the hardware based transmit ring imposes a certain limited amount of extra latency for a high-priority packet. The range of values of the upper limit for the *tx-ring-limit* command is determined dynamically at run time. Entering "*tx-ring-limit?*" in the command line this can be viewed. The key determinants are the memory needed to maintain the queues and the memory existing on the device.

For example:

hostname(config)# priority-queue outside

hostname(config-priority-queue)# queue-limit 360

hostname(config-priority-queue)# tx-ring-limit 3

V. CONCLUSION AND FUTURE WORKS

The existing research suggest how the queuing delay can be measured using a Network Time Protocol in VoIP. A synchronize clock between sender and receiver is been used in this approach. And also the research suggests how the delay can be reduced using configuring priority queue algorithm.

The proposed research can be further applied to other ways of traffic not only for VoIP which will improve the quality of service. And also the research team would work to implement proposed mechanism.

VI. ACKNOWLEDGEMENT

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